Manager

Manager Manager 8.1

15-601011 Issue 25l - (11 February 2011)

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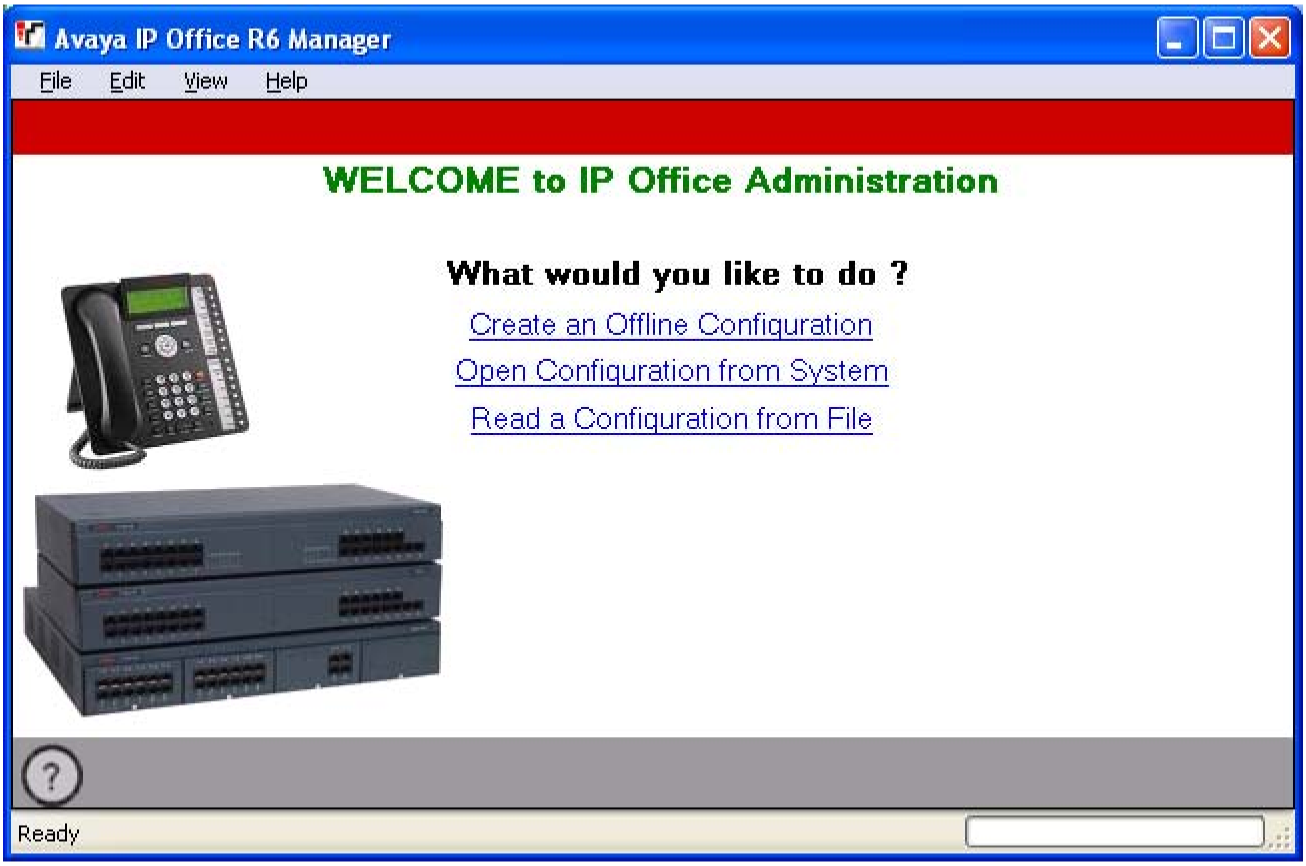
Chapter 1.
Manager

1. Manager

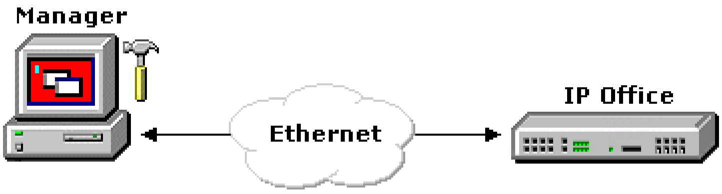
Manager is an application for viewing and editing an Manager system's configuration. It is a tool for system installers and maintainers.

This documentation covers the use of the Avaya IP Office R6.1 Manager to configure an Manager system. For a summary of the new features in Release 6.1 refer to the Appendix 939.

IP Office systems can run in several modes. Unless specifically mentioned, details within this documentation are for systems running in IP Office standard mode.



Manager runs on a Windows PC and connects to the system via Ethernet LAN or WAN connections.



- WARNING - Password Changes Required
  New systems use default security settings. These settings must be changed to make the system secure. As a minimum, you must change the default passwords of the Security Administrator, the default Service Users and the Unsecured Interfaces System Password. Failure to do so will render the system potentially unsecure. See the Security Mode 68 section for details.

- IMPORTANT - Manager is an Off-Line Editor
  Manager is an off-line editor. It receives a copy of the system's current configuration settings. Changes are made to that copy and it is then sent back to the system for those changes to become active. This means that changes to the active configuration in the system that occur between Manager receiving and sending back the copy may be overwritten. For example this may affect changes made by users through their phone or voicemail mailbox after the copy of the configuration is received by Manager.

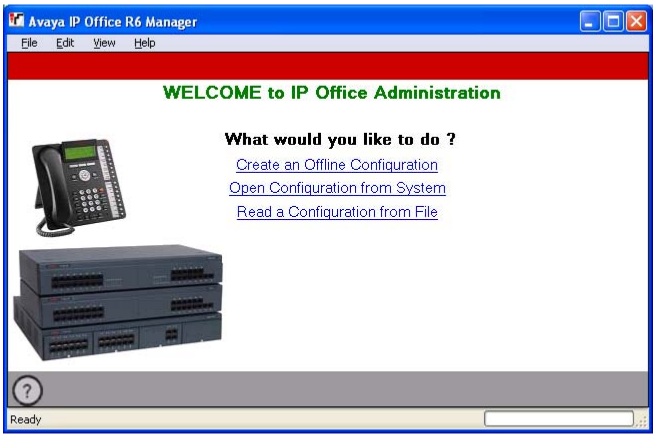
Manager is part of the IP Office Admin suite of programs. It is important to note that the software level of Manager application is 2 higher than the software level of the system core software with which it is released. For example Manager 8.1 was released with Manager 6.1 core software. The Manager application can be used to manage configurations from systems running IP Office 2.1 core software upwards. See Backward Compatibility 23.

Manager Modes

The menus and options displayed by Manager vary depending on the actions you are performing. Manager can run in the following modes.

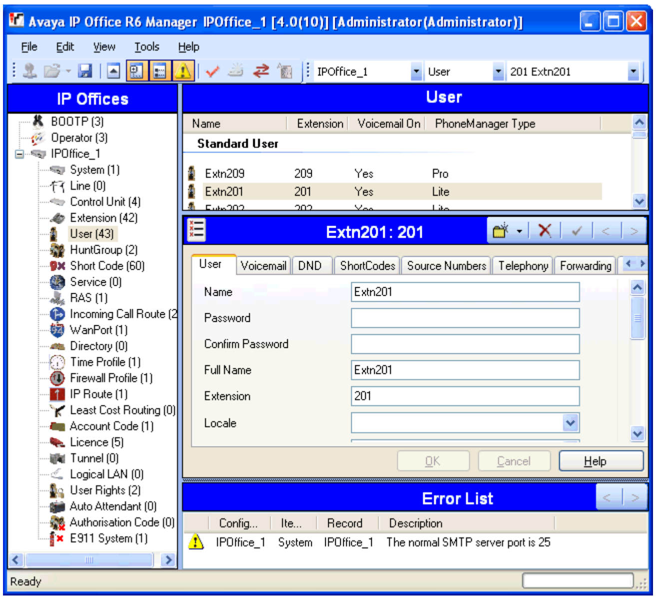
Simplified View

This is Managers default mode when no IP Office configuration has been opened.



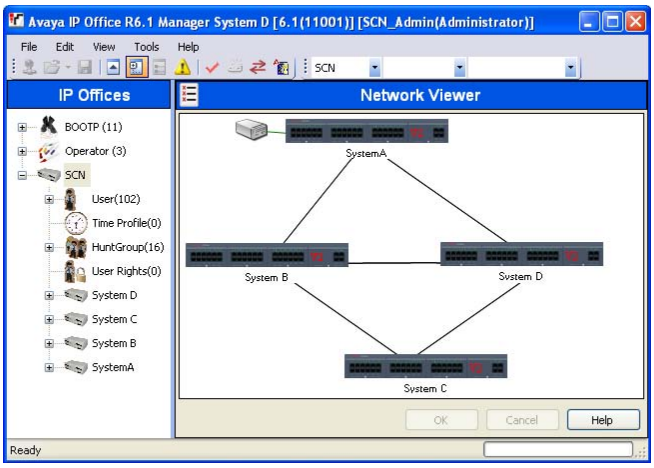
[Advanced Configuration Mode](32)

When the configuration from a system running in IP Office Standard Version mode is opened in Manager, it displays options for this mode.



[SCN Management Mode](817)

Manager 8.1+ supports loading the combined configurations from systems in a Small Community Network. For more details see [SCN Management](817).

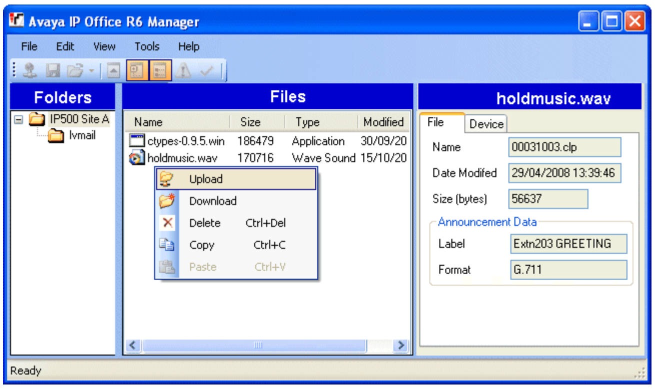


Simplified Administration Mode

When the configuration from a system running in IP Office Essential Edition - PARTNER® Version or IP Office Essential Edition - Norstar Version mode is opened in Manager, Manager switches to this alternate management mode.

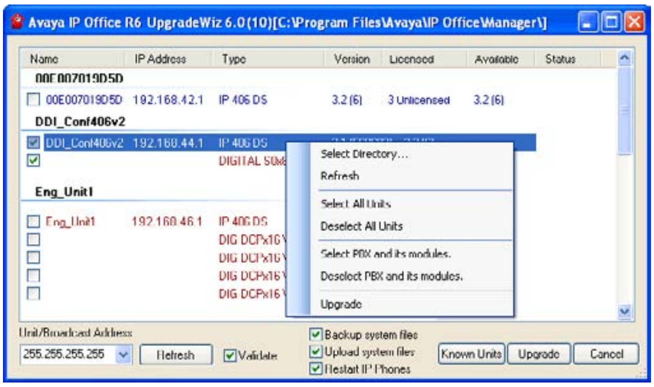
Embedded File Management 119

For systems with a memory card installed, Manager can be used to view and manage the files stored on the card. This is accessed through the File | Advanced | Embedded File Management.. 119. .



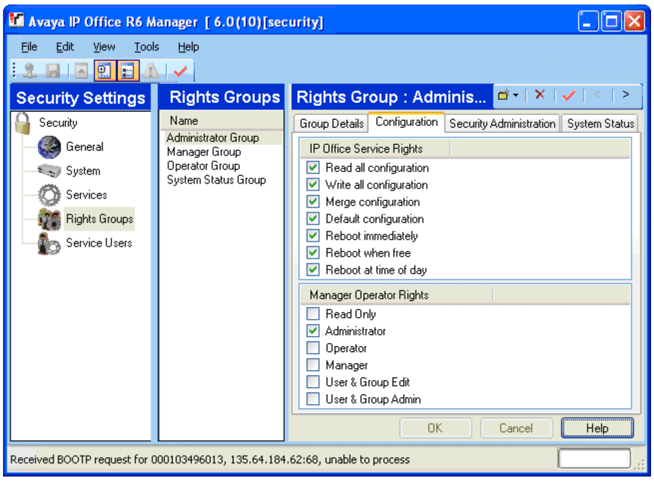
Upgrade Wizard 115

The Upgrade Wizard is a component of Manager used to upgrade the firmware run by the control unit and expansion modules within a system.



Security Configuration Mode 68

Manager can be used to edit the security settings of systems running in IP Office Standard Version mode.



1.1 Installing Manager

Manager is a component of the IP Office Admin suite of applications. This suite is supplied on the IP Office Release 6.1 DVD. Alternatively, the Admin Suite can be downloaded from Avaya's support website http://support.avaya.com.

In addition to Manager, the IP Office Admin suite includes options to install the following applications:

- System Monitor
  This is a tool for system installers and maintainers. Interpreting the information output by System Monitor requires detailed data and telecoms knowledge.
- System Status Application
  This is a Java application that can be used to monitor the status of the system such as extension, trunks and other resources. It displays current alarms and most recent historical alarms.
- Call Status
  This application is for pre-4.0 systems. For IP Office 4.0 and higher use the System Status Application. Manager 6.2+ is backwards compatible and can be used to manage systems running software from IP Office 2.1 upwards.

PC Requirements

Minimum PC Requirements	
RAM	256MB
Hard Disk Free Space	1GB*
Processor:	
- Pentium	PIII 800MHz
- Celeron	Celeron 3 800Mhz
- AMD	Athlon Opteron, Athlon64/XP
Additional Apps:	
- .NET2	Installed with Manager if not already present.

Operating System Support	
Server OS:	
2003 Server	✓
2008 Server	✓
Client OS:	
XP Professional	✓
Vista Business/Enterprise	✓
Vista Ultimate	✓
Windows 7	✓

*Includes disk space required for .NET2 component.

- Language Support
  The Manager application can run in English, French, German, Brazilian Portuguese, Dutch, Italian and Latin Spanish. By default this determined by the best match to the PC's regional setting. The online help is only provided in English, French and German. To change the language used see Changing the Manager Language 23.

Ports

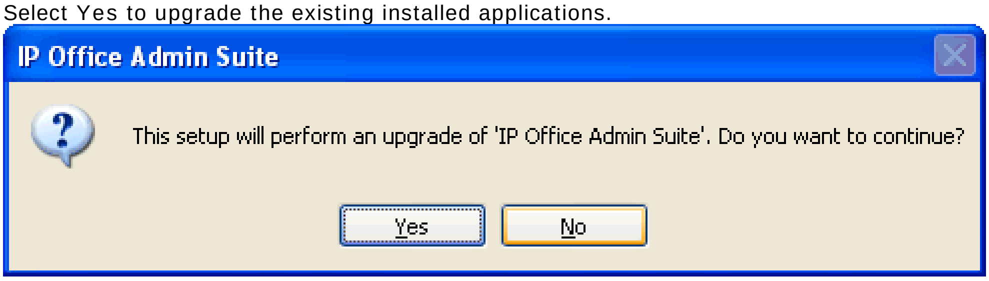
Component	Location - %ProgramFiles%\Avaya\IP Office\	Ports
Manager	Manager\manager.exe	TCP Port 50802. TCP Port 50804. TCP Port 50812. UDP Port 50798.
IP Office Upgrade Wizard	Manager\upgradewiz.exe	UDP Port 50798.

Installing Manager

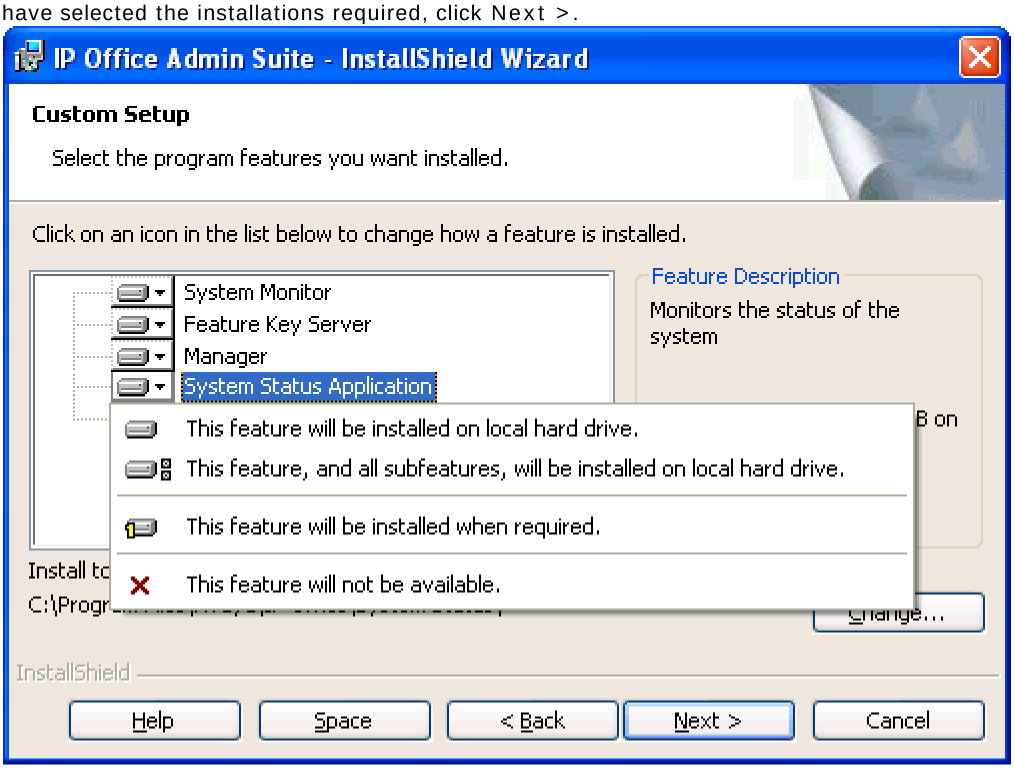
If applications in the IP Office Admin Suite other than Manager are required, we strongly recommend that you refer to the Manager Installation Manual.

- **Note**: This installation process will install Windows .NET2 if not already present. The installation of .NET2 may require some systems to restart and the installation process to then be restarted.

1. If a pre-4.0 version of the IP Office Admin Suite is installed it must be removed. This is done using the **Add or Remove Programs** option in the Windows Control Panel and selecting **IP Office Admin Suite** and then **Remove**.
2. If installing from the IP Office Admin DVD, insert the DVD and when the page is displayed click on the link for the IP Office Admin suite. This will open a file windows showing the installation files for the suite. Locate and double-click on the **setup.exe** file.
3. Select the language you want to use for the installation process. This does not affect the language used by Manager when it is run. Click **Next >**.
4. If the following appears it indicates that a previous installation of the IP Office Admin suite has been detected. Select Yes to upgrade the existing installed applications.



5. If required select the destination to which the applications should be installed. We recommend that you accept the default destination. Click **Next >**.
6. The next screen is used to select which applications in the suite should be installed. Clicking on each will display a description of the application. Click on the ▼ next to each application to change the installation selection. When you have selected the installations required, click **Next >**.



7. The applications selected are now ready to be installed. Click **Next >**.
8. Following installation, you will be prompted whether you want to run the IP Office Admin Suite. Selecting *Yes* runs Manager.
9. On some versions of Windows, you may be required to restart the PC. Allow this to happen if required.

Changing the Installed Applications

The **Add or Remove Programs** option can be used to change the selection of IP Office Admin suite applications that are installed. Locate **IP Office Admin Suite** in the list of programs and select **Change**.

1.2 Starting Manager

No operator name or password is required to start Manager. A name and password is only required when performing an action that requires communication with a system.

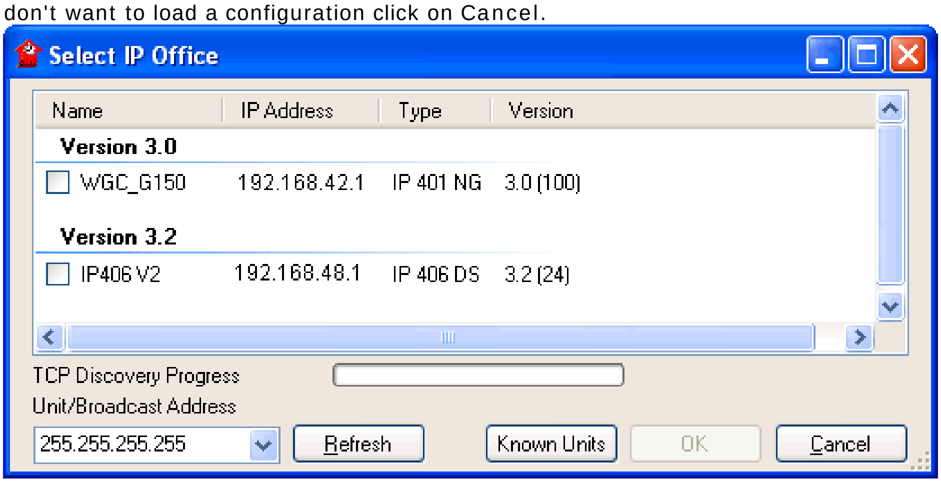
When started, by default Manager will attempt discover any systems on the network. If it finds any it will display a list from which you can select

1. Select Start and then Programs or All Programs depending on the version of Windows. Select the IP Office program group.

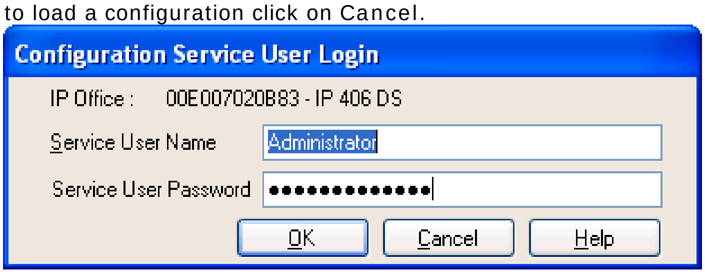
2. Select Manager. If a Windows Security Alert appears select Unblock to allow Manager to run.

3. By default Manager will scan the network for any systems. What appears next depends on whether it finds any systems.

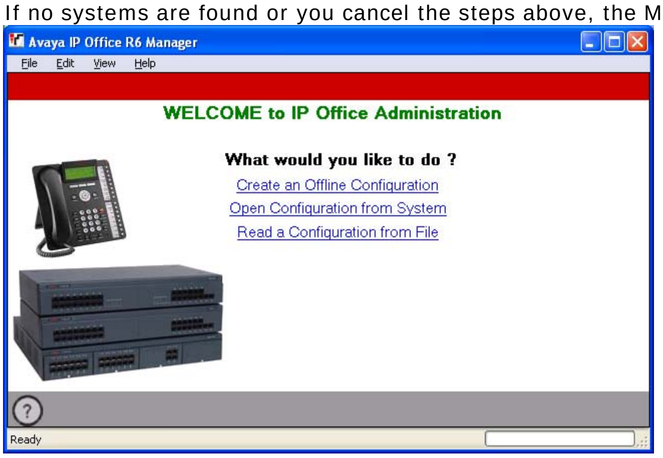
   - If it finds multiple systems, it will display a list of those systems from which you can select the one whose configuration you want to edit. If you want to open a configuration go to Opening a Configuration 19. If you don't want to load a configuration click on Cancel.



   - If it finds a single system, it will attempt to open the configuration of that system by displaying the user name and password request. If you want to open a configuration go to Opening a Configuration 19. If you don't want to load a configuration click on Cancel.



   - If no systems are found or you cancel the steps above, the Manager simplified view is displayed.



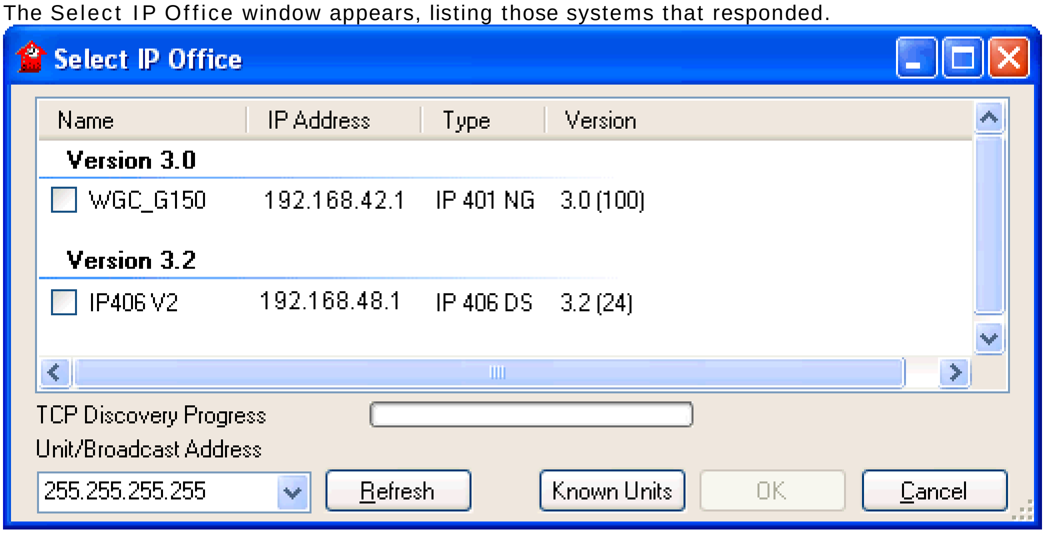
4. Use the view to select the action you want to do.

   - Create an Offline Configuration 60
   - Open a Configuration from a System 19
   - Read a Configuration from a File

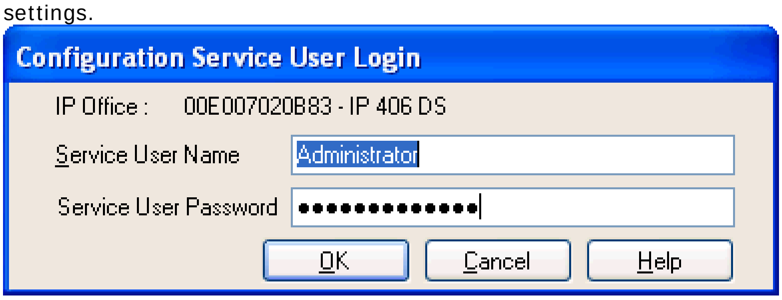
1.3 Opening a Configuration

The initial IP address ranges in which Manager searches for systems are set through the Manager preferences (File | Preferences | Discovery 108). By default it scans the local network of the Manager PC.

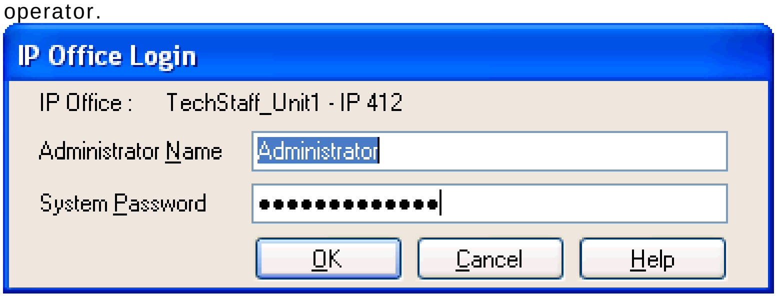
1. Start Manager. If Manager is already started and a configuration is open in it, that configuration must be closed first.
   - If Manager is set to Auto Connect on start up 106, it will scan for systems automatically and either display the list of systems discovered or automatically start login to the only system discovered.
   - Otherwise, click on  or select File | Open Configuration.
2. The Select IP Office window appears, listing those systems that responded.



   - If Manager has been set with SCN Discovery 108 enabled, systems in a Small Community Network may be grouped and can be loaded into Manager's SCN management 817 mode.
   - If the system required was not found, the Unit/Broadcast Address used for the search can be changed. Either enter an address or use the drop-down to select a previously used address. Then click Refresh to perform a new search.
   - The address ranges used by Manager for searching can be configured through the File | Preferences | Discover 108 tab.
   - A list of known systems can be stored and used. See Known System Discovery 59.
   - Manager can be configured to search using DNS names. See the Use DNS 108 option.
   - Systems found but not supported by the version of Manager being used will be listed as *Not Supported*.
   - If the IP500v2 system detected is running software other than from its primary folder, a  warning icon will be shown next to it. The configuration can still be opened but only as a read-only file.
3. When you have located the system required, check the box next to the system and click OK.
4. The system name and password request is displayed. Enter the required details and click OK.
   - IP Office 3.2 and higher systems:
     The name and password used must match a Service User account configured within the system's security settings.



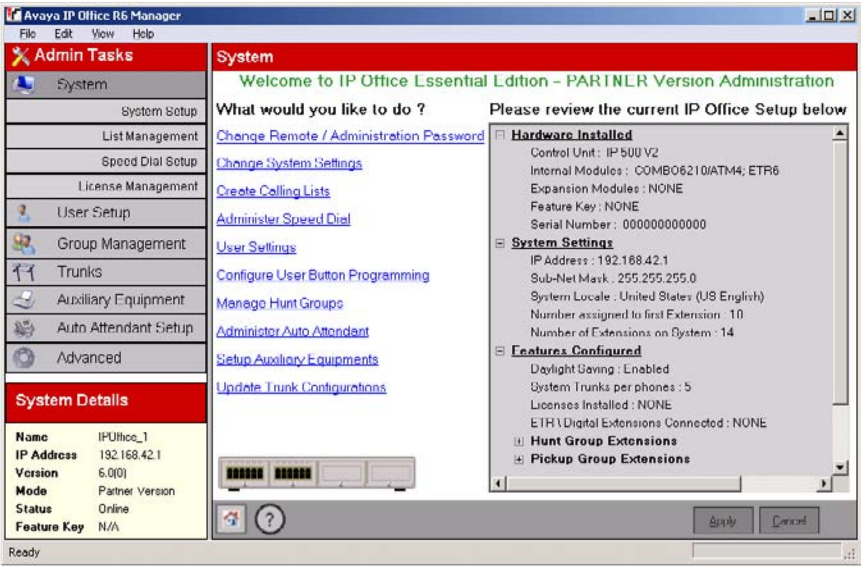
- **Pre-3.2 IP Office Systems:**
  The name must match a Manager operator and the password must match the system's system password. If the name does not match a Manager operator, the config will still be loaded by using the *Guest (read-only)* operator.



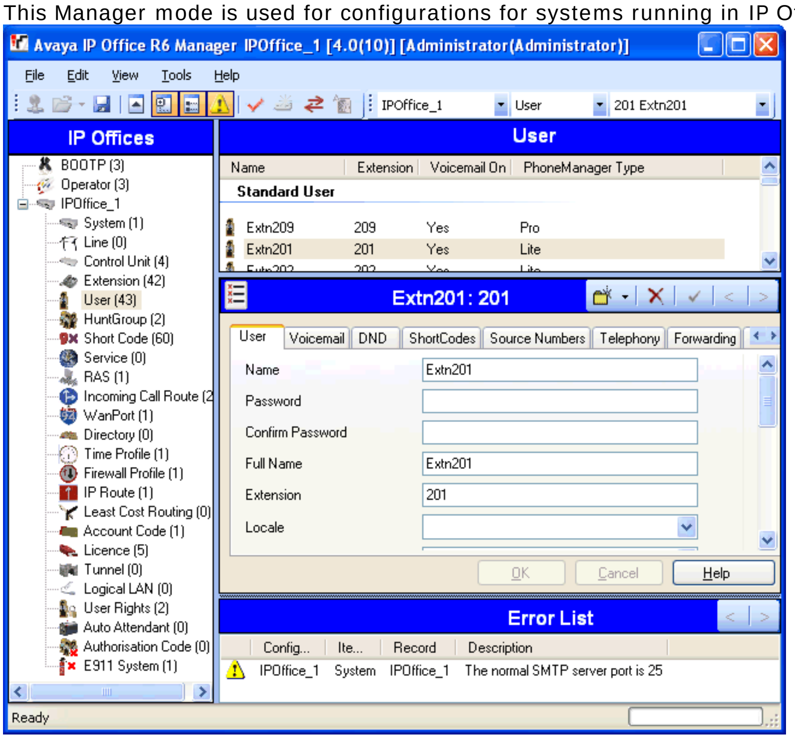
5. Additional messages will inform you about the success or failure of opening the configuration from the system.

6. Following a successful log in, the configuration is opened in Manager. The menus and options displayed will depend on the type of system configuration loaded.

- **PARTNER Version/Norstar Version Configuration**
  This Manager mode is used for configurations from systems running in PARTNER Version or Norstar Version modes.



- **IP Office Configuration**
  This Manager mode is used for configurations for systems running in IP Office Standard Version mode.



Login Messages

While attempting to login to a system, various additional messages may be displayed.

Configuration Not Loaded Messages

- **Access Denied**
  This is displayed as the cause if the service user name/password were incorrect, or the service user has insufficient rights to read the configuration. The Retry option can be used to log in again but multiple rejections in a 10 minute period may trigger events, such as locking the user account, set by the Password Reject Limit and Password Reject Action options in the systems security settings.

- **Failed to communicate with system**
  This is displayed as the cause if the network link fails, or the secure communication mode is incorrect (for example Manager is set to unsecured, but the system is set to secure only).

- **Account Locked**
  The account of the Service User name and password being used is locked. This can be caused by a number of actions, for example too many incorrect password attempts, passing a fixed expiry date, etc. The account lock may be temporary (10 minutes) or permanent until manually unlocked. An account can be enabled again through the system's security settings.

Additional Messages

- **Your service user account will expire in X days**
  This message indicates that an Account Expiry date has been set on the system service user account and that date is approaching. Someone with access to the system's security settings will be required unlock the account and set a new expiry date.

- **Your password will expire in X days. Do you wish to change it now?**
  This message indicates that password ageing has been configured in the system's security settings. If your password expires, someone with access to the system's security settings will be required to unlock the account.

- **Change password**
  Through the system's security settings, a service user account can be required to change their password when logging in. The menu provides fields for entering the old password and new password.

- **Contact Information Check - This configuration is under special control**
  This message will appear if a Manager user with administrator rights has entered their contact information into the configuration. For example to indicate that they do not want the configuration altered while a possible problem is being diagnosed. The options available are:

- **Cancel**
  Select this option to close the configuration without making any changes.

- **Set configuration alteration flag**
  Select this option if the configuration is being opened because some urgent maintenance action. When the configuration is next opened, the fact that it has been altered will be indicated on the System | System 143 tab.

- **Delete Contact Information**
  Select this option to take the system out of special control.

- **Leave contact information and flags unchanged** ***(Administrators only)***
  This option is only available to service users logging in with administrator rights.

1.4 Changing the Manager Language

The Manager application can run in *US English*, *UK English*, *French*, *German*, *Brazilian Portuguese*, *Dutch*, *Italian* and *Mexican Spanish*. By default it tries to use the best match to the PC's regional location settings, otherwise it will use UK English.

The process below can be used to run Manager in one of its supported languages. However it does not change the language used for help file content.

1. Create a Windows shortcut to the Manager application .exe file. By default this file is located in *C:\Program Files\Avaya\IP Office\Manager\Manager.exe*.
2. Right-click on the shortcut and select Properties.
3. The Target field can be used to specify the locale setting that Manager should use.
   - For example, for Italian the Target should have *-locale:it-IT* added to the end. For example: *"C:\Program Files\Avaya\IP Office\Manager\Manager.exe" -locale:it-IT*
4. The available locales for Manager are:

Manager Language	Shortcut Locale Setting
Brazilian Portuguese	-locale:pt-Br
Dutch	-locale:nl-NL
French	-locale:fr-FR
German	-locale:de-DE
Italian	-locale:it-IT
Mexican Spanish	-locale:es-MX
US English	-locale:en-US

5. Click OK.
6. Manager should now run in the selected language when launched using the updated shortcut.

1.5 Backward Compatibility

Manager is part of the IP Office Admin Suite of programs. It is important to note that the software level of Manager application is 2 higher than the software level of the system core software with which it is released. For example Manager 6.2 was released with Manager 4.2 core software. The Manager application can be used to manage configurations from systems running IP Office 2.1 core software.

When an IP Office 2.1 or higher configuration is loaded, Manager adjusts the settings and fields that it shows to match the core software level of the system. If you attempt to load a pre-2.1 IP Office configuration, Manager will display an error message and does not load the configuration.

- To receive a pre-3.2 IP Office configuration requires entry of an operator name and the system password.
- To receive a 3.2 or higher IP Office configuration requires entry of a service user name and password stored by that system.

For IP Office 4.2+, Manager is able display systems with software levels it does not support in the Select IP Office discovery menu, however those systems are indicated as not supported.

- Backwards compatibility is only supported for General Availability releases of <IP Office software. It is not supported for private builds.

1.6 What's New in Release 6.1

IP Office 6.1 is supported on IP406v2, IP412, IP500 and IP500v2 systems.

- Phone Support 934
- Small Community Network Management 934
- one-X Portal for IP Office 934
- IP Office Customer Call Reporter 935
- Embedded Voicemail 935
- IP Office Essential Edition - PARTNER® Version 935
- IP Office Essential Edition - Norstar Version 937
- Voicemail Pro 937
- Music on Hold 937

Phones

The following additional phones and phone features are supported by IP Office 6.1:

- 1000 Series
  The 1010 and 1040 SIP devices are HD video softphone devices. The main units provided connections for a variety of video and audio inputs and outputs.
- 1100/1200 Series
  The following phones from the Avaya range of SIP phones are supported: 1120E, 1140E, 1220 and 1230.
- Automatic Call Log Expiry
  For call log entries written into a user's centralized call log, a expiry time can be set after which the call log entry is automatically deleted from the user's call log.
- Mobile Twinning Handover (IP Office Release 6.1)
  When on a call on the primary extension, pressing the Twinning button will make an unassisted transfer to the twinning destination. This feature can be used even if the user's Mobile Twinning setting was not enabled.
  - During the transfer process the button will wink.
  - Pressing the twinning button again will halt the transfer attempt and reconnect the call at the primary extension.
  - The transfer may return if it cannot connect to the twinning destination or is unanswered within the user's configured Transfer Return Time (if the user has no Transfer Return Time configured, a enforced time of 15 seconds is used).
- Show Last Call Duration
  1400, 1600 and 9600 phones will briefly display the duration of a call after it is ended. This setting is a user accessible option through the phone menu Features | Call Settings | Show Last Call Duration (except on 1403 and 1603 phones where it is on by default).

Small Community Network Management

Manager 8.1 is able to load and display the configuration of all systems in a Small Community Network. The configurations can be edited and saved. When working in SCN management mode, Manager can also display a graphic view of the network and allow the adding of additional systems and SCN lines between systems.

one-X Portal for IP Office

IP Office 6.1 supports one-X Portal for IP Office 6.1. one-X Portal for IP Office 6.1 provides the following new features:

- Adjustable Layout
  The gadgets provided on the one-X Portal for IP Office's main page can now be moved, resized and minimized by the user. The users positioning of the gadgets is retained between one-X Portal for IP Office sessions.
- Multi Directory Search
  It is now possible to display results for a search across all directories (System, Personal and External).
- Call Assistant
  The one-X Portal for IP Office Call Assistant is now supported for Windows based users of one-X Portal for IP Office. The Call Assistant provides pop up messages about calls even when not logged into one-X Portal for IP Office, screen popping to Outlook and hot key dialing of numbers.

IP Office Customer Call Reporter

- Microsoft SQL 2008 Support
  IP Office Customer Call Reporter is now supported using Microsoft SQL 2008.
- Visual Redesign
  Some elements of the browser display have been redesigned.
  - Many of the tabs used by Supervisors to move between the available pages have been replaced by icons.
  - The color used for warnings has been changed from yellow to orange.
- New Supervisor Pages
  Supervisors are able to access two new pages of call information in addition to the views that they share with their agents.
  - Dashboard Display
    The dashboard display is the default page displayed to a supervisor when they login. They can customize it to display a combination of up to three graphs and data display elements.
  - Customer Map
    Supervisors can display and configure a map that will plot calls based on caller ID numbers. The map can be used to show a combination of historical and realtime calls and can be overlayed onto Google or Yahoo maps.
- Force Agent State
  Supervisors can now force force a change to an agent's status. For example log an agent out or enable/disable an agents queue membership. The IP Office Customer Call Reporter administrator configures which supervisors have this function. This feature requires the IP Office Customer Call Reporter server to have access to a one-X Portal for IP Office server.
- New Statistics
  The following new statistics are available for use in views.
  - Agent Productivity Factor
  - Talk Average
  - Talk Inbound
  - Talk Inbound Average
  - Talk Internal
  - Talk Outbound
  - Talk Outbound Average
  - Talk Total
- Report Templates
  The follow changes have been made for historical reporting.
  - The Call Summary Report now includes Average Answer Time when reporting on agents (previously this value was blank when reporting on an agent or agents).
  - A new template called the Agent Report Card template has been added. It provides historical reporting on the Talk Time statistics.
- Message Color
  When scheduling a wallboard message, supervisors can now select the color for the message.
- IP Office Customer Call Reporter Help
  IP Office Customer Call Reporter help is now also available in French and Latin Spanish.
- Wallboard Controls
  The wallboard Background and Content Settings now include options to adjust the animation effect applied to changing statistic values and the aspect ratio used for the display of the wallboard elements.

Embedded Voicemail

- Skip Your Mailbox Greeting
  Caller's can skip your mailbox greeting by pressing 1. Instead they immediately hear the tone for the start of recording.

IP Office Essential Edition - PARTNER® Version

- Phantom Users
  Previously user settings were only created and configurable for the physical extensions present in the system (and excluding ports 7 and 8 on ETR6 cards). User settings are now created for all possible user extensions regardless of whether a matching physical extension port is present or not.
  - Calls to a phantom extension number go directly to that user's mailbox. This applies for normal dialing, DDI call routing, line coverage, transfers and routing from an auto-attendant.
- Auto Attendant Enhancements
  The following changes have been made to auto attendant support for IP Office Essential Edition - PARTNER® Version 6.1.
  - 9 Auto Attendants
    Up to 9 auto attendants are now supported. Where a line is associated with an auto attendant the auto attendant required can be selected.

- Transfer to Auto Attendant Action
  This additional menu action allows calls to be routed from one auto attendant to another. When this occurs, only the menu prompt of the new auto attendant is played.

- Language Selection
  Each auto attendant can be configured with a language selection. The selected language controls the prompts used by the auto attendant actions where applicable.

- Time Profile Dependant Menu Actions
  In addition to controlling with initial greeting is played to callers, each auto attendants time profiles now also control which set of menu actions are available to the callers with the appropriate time dependant menu actions greeting.

- Selectable Night Service Mode
  Previously when the system was put into night service, the auto attendant switched to its out of hours greeting. Each auto attendant now has a Follow Night Service setting. If selected, the previously behavior still applies. If not selected, when the system is put into night service, the auto attendant continues to follow its own time profile settings.

- Picking Up Auto Attendant Calls
  Bridging into a call being handled by an auto attendant drops the auto attendant from the call.

- Emergency Greeting
  Each auto attendant can have a recorded emergency greeting and a setting for whether that greeting is active or not. When active, the emergency greeting is played before any of the other auto attendant greetings. When an emergency greeting is active, a warning is displayed on extensions 10 and 11.

  - Transfer to Greeting Action
    This additional menu action allows the caller to play and record the emergency greeting. It also allows them to select whether the greeting is active or not. If a system password has been set, it is used to restrict access to this option.

- Line Enhancements

  - Unique Line Ringing
    Each incoming line, channel or DID can be assigned a ringing pattern. That pattern is then used for incoming calls on that line unless overridden by the user's setting.

  - Assigning Lines to Auto Attendants
    Previously assignment of a line to an auto attendant could only be done through the Manager. This option can now be selected through the administrator menus on extensions 10 and 11.

- User Setting Enhancements
  The following changes have been made to user support for IP Office Essential Edition - PARTNER® Version 6.1.

  - Immediate Voicemail
    The user setting for VMS Cover Ring can now be set to *0* for immediate voicemail.

  - Transfer Return Extension
    For users who have transfer return enabled, a different extension destination for the return calls can now be specified.

  - Override Line Ringing
    For selected users the new Unique Line Ringing can be overridden.

- Language Control
  Previously only individual users could be configured with a language selection. For IP Office Essential Edition - PARTNER® Version 6.1 the default language selection can now be done at the system level. In addition a language setting can be applied to each auto attendant.

- Embedded Voicemail Enhancements
  The following changes have been made to the embedded voicemail provided to IP Office Essential Edition - PARTNER® Version 6.1 users.

  - Caller Post Message Options
    After leaving a mailbox message, callers can now press # rather than hanging up immediately. The caller will hear a prompt informing them whether the message has been saved or whether the messages was too short (less than 3 seconds) and so was not saved.

  - Skip Your Mailbox Greeting
    Caller's can skip your mailbox greeting by pressing 1. Instead they immediately hear the tone for the start of recording.

  - Mark Messages as New
    You can now change an old or saved message's status back to new while it is being played or just after it has played by dialing **06*. The message waiting light is relit. However if voicemail email is being used no new message email is sent.

IP Office Essential Edition - Norstar Version

To replace Avaya Norstar systems in the middle eastern market, IP500v2 systems can be configured to run in IP Office Essential Edition - Norstar Version mode. This is done using a IP Office Essential Edition - Norstar Version System SD card. Operation is similar to IP Office Essential Edition - PARTNER® Version except with no support for ETR phones.

- The locales supported for IP Office Essential Edition - Norstar Version are: Bahrain, Egypt,, Kuwait, Morocco, Oman, Pakistan, Qatar and the United Arab Emirates.
- Automatic impedance matching is also available for the locales above.
- For embedded voicemail, Arabic language prompts are supported.

Voicemail Pro

The following features have been added to Voicemail Pro 6.1.

- Additional Generic Action String Manipulation Options
  The String Manipulation command has two additional options. They are:
  - Copy
    This action can be used to copy the value of one variable to another variable. The command can copy the whole value or can, treating the value as a string, copy a section to or from a specified matching character.
  - Length
    This action can be used to return the length of variable. It can return the full length or the length from or to a specified matching character.
- Post Call Completion Call Flows
  Call flows can be configured to continue running even after the caller has disconnected. If the current action which the call had reached has a *Timeout* or *Next* result, the connection from that result is followed immediately until the call flow either reaches a Disconnect action or an unconnected result.
- Automatic Call Recording for Internal Calls
  The user and hunt group options for call recording can now be set to work for internal calls. Previously they only applied to external calls.

Music on Hold

The following changes have been made for hold music support:

- Analog Extension Source
  For alternate music on hold sources 2 to 4, the audio input to an analog extension port can be specified as the source.
- Short Code Music on Hold Selection
  For calls routed by short codes, including outgoing external calls, the hold music source for the call can be specified using the h character in the short code telephone number.

IP Office Application Server

The IP Office Application Server is a Linux based server running server applications, currently Voicemail Pro and one-X Portal for IP Office. It does not require Linux knowledge to install. The sever and the applications are managed remotely via web browser and the existing Voicemail Pro client.

Miscellaneous Features

- Restrict Analog Extension Ringer Voltage
  If this system setting is selected, the available message waiting lamp indication settings for analog extensions are restricted to *None* and *Line Reversal*.
- Time Support
  A number of changes have been made to the way that the systems time and date can be set.
  - The existing RFC868 Time method using the Voicemail Pro server or Manager as the source is still supported. Use of other RFC868 server sources is not supported.
  - Using SNTP requests to a NTP server is now supported. When using SNTP, additional fields for the local time offset and for daylight savings time (DST) settings are also available.
  - If the time is being set manually through a system phone, the system can be configured with a manually adjustable offset and or with automatic daylight savings time settings.
  - The time settings, including the time server being used and whether DST is in use or not, can be viewed through system phones.

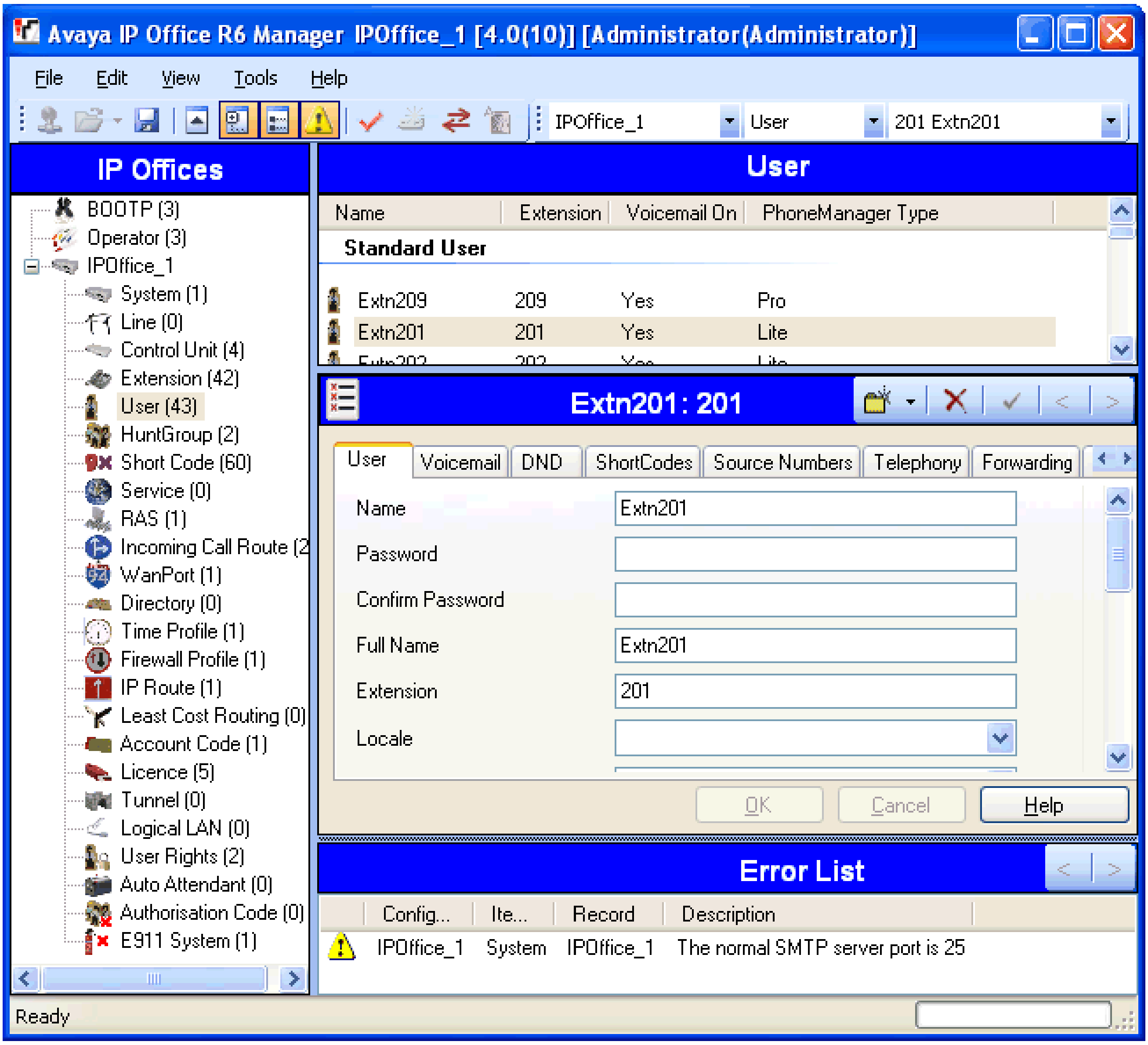
- **Restart IP Phones on System Upgrade**
  The Manager Upgrade Wizard now includes an option to restart all connected IP phones following a system upgrade. This will cause those phones to check and, if necessary, upgrade their phone firmware following the system upgrade.

- **Multiple SIP Proxies**
  The use of multiple SIP proxy servers in SIP line settings is now supported. The multiple proxies can be specified by IP address or obtained through DNS response (RFC3263). When specified by IP address, weighting can be applied to each address.

Chapter 2. Configuration Mode

2. Configuration Mode

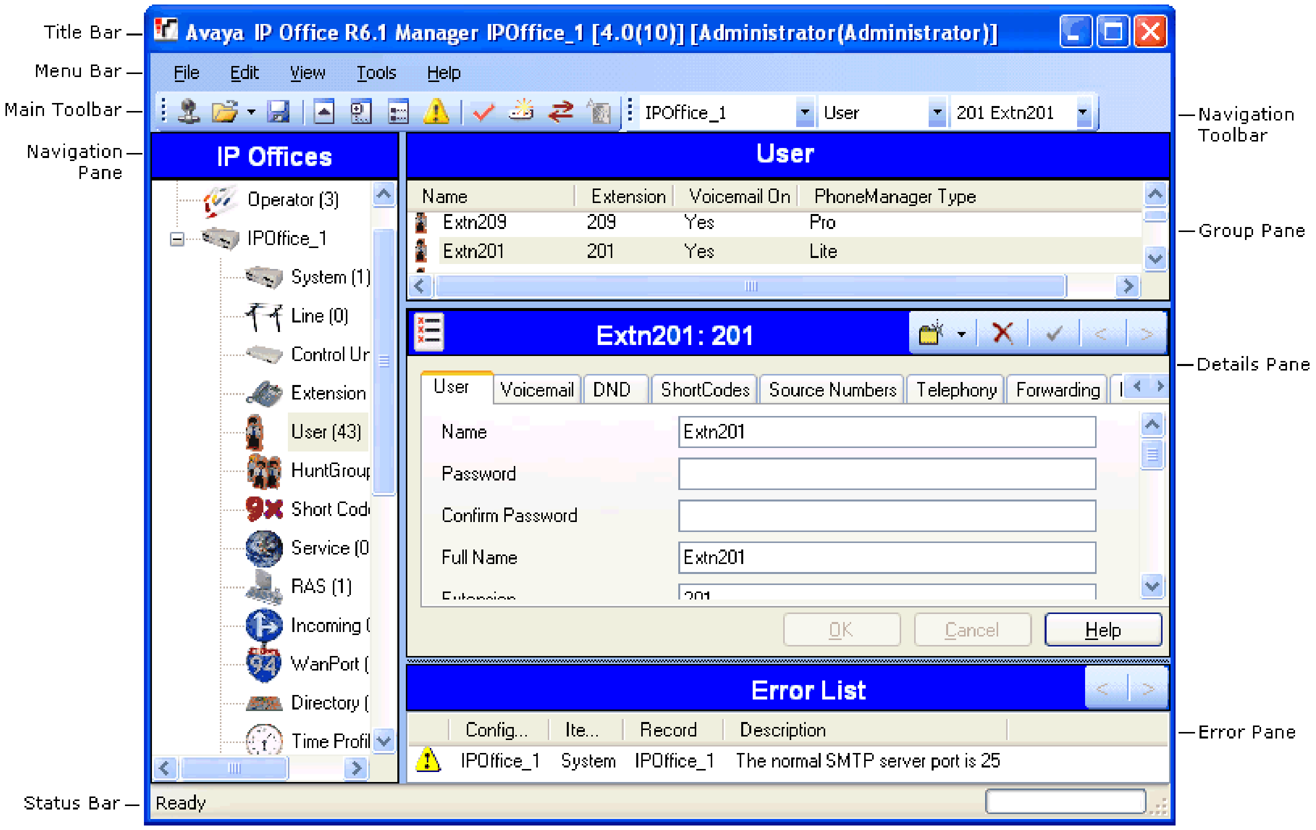
This section of the documentation covers the operation of Manager when being used to edit the configuration of a system running in Manager mode.



This section is divided as follows.

2.1 The Configuration Mode Interface

When Manager is in configuration mode, the screen elements shown are available. Some of these elements can be customized, moved and hidden.



Manager Configuration Mode Screen Elements

- Title Bar 36
  In addition to the application name, when configuration settings are loaded from a system, the title bar displays the user name used to load the settings and the operator view applied.

- Menu Bar 37
  The options available with the drop down menus provided here change according to whether Manager has a set of configuration or security settings loaded or not.

- Main Toolbar 38
  This toolbar provides icon shortcuts to the most frequently required configuration setting actions.

- Navigation Toolbar 38
  This toolbar provides a set of drop downs which can be used to navigate to particular entries in the configuration settings. The selected options in the navigation pane, the group pane and the details pane are synchronized with the navigation toolbar and vice versa. This toolbar is particularly useful if you want to work with the group pane and or navigation pane hidden in order to maximize the display space for the details pane.

- Navigation Pane 40
  This pane shows icons for the different types of entry that the configuration can contain. Each type is followed by the number of entries of that type already in the configuration. Selecting an icon displays the matching entries in the group pane and navigation toolbar.

- Group Pane 41
  This pane lists all the entries that match the type selected in the navigation pane or navigation toolbar. The list can be sorted by clicking on column heading. Selecting an entry in this pane displays its details in the details pane.

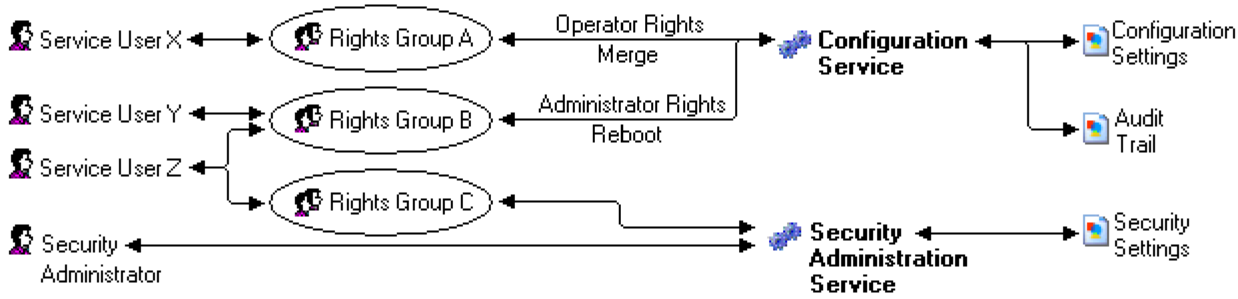
- Details Pane 43
  This pane shows the configuration settings for a particular entry within the configuration. The entry is selected using the navigation toolbar or using the navigation pane and group pane.

- Error Pane 45
  This pane shows errors and warnings about the configuration settings. Selecting an item here loads the corresponding entry into the details pane.

- Status Bar 48
  This bar display messages about communications between Manager and systems. It also displays the security level of the communications by the use of a padlock icon.

2.2 Security Settings

The following applies for IP Office 3.2 and higher. Access to system settings is controlled by Service Users and Rights Groups. stored in the control unit's security settings. These are stored separately from the system's configuration settings. All actions involving communications between Manager and the system require a Service User name and password. That Service User must be a member of a Rights Group with permissions to perform the required action.



In the example illustrated above:

- Service User X can read and write the configuration. However they can only edit Operator settings and can only make changes that can be merged.
- Service User Y can read and write the configuration, edit all settings and make changes that require reboots.
- Service User Z can read and write the configuration, edit all settings and make changes that require reboots. They can also access the security settings.
- The Security Administrator can only access the security settings.

Security Administrators

By default the security administrator is the only user who can access the system's security settings using Manager's security mode.

Service Users

Each Service User has a name, a password and is a member of one or more Rights Groups.

Rights Groups

The Rights Groups to which a Service User belongs determine what actions they can perform. Actions available to Rights Groups include configuration actions, security actions and system status actions:

Configuration	Security	System Status
• Read the configuration. • Write the configuration. • Merge the configuration. • Default the configuration. • Reboot immediately. • Reboot when free. • Reboot at time of day.	• Read all security settings. • Write all security settings. • Reset all security settings. (Manager 4.1+) • Write own password. (Manager 4.1+)	• System Status Access. • Read All Configuration.

Where a Service User has been configured as a member of more than one Rights Group, they combine the functions available to the separate Rights Groups.

Operator Rights

Each Rights Group has a Manager Operator Rights setting. This setting controls what types of configuration entries Manager will allow members of the Rights Group to view and what actions they can perform with those types of entries.

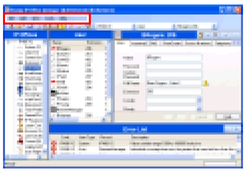
Operator	View / Edit / New / Delete	Configuration Entry Types
Administrator	All	View, edit create and delete all configuration entries.
Manager	View	View all except WAN Port.
	Edit	Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, Directory, Time Profile, Firewall Profile, IP Route, Least Cost Route, Account Code, ARS, E911 System.
	New	
	Delete	As edit except Short Code.
Operator	View	View all except WAN Port.
	Edit	Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, Time Profile, Firewall Profile, IP Route, Least Cost Route, Account Code, License, ARS, E911 System.
	New	None.
	Delete	Delete Incoming Call Route and Directory.
User & Group Edit	View	User and Hunt Group entries only.
	Edit	
	New	None
	Delete	
User & Group Admin	All	User and Hunt Group entries only.
Dir & Account Admin	All	Directory and Account Code entries only.
Time & Attendant Admin	All	Time Profile and Auto Attendant entries only.
ICR & User Rights Admin	All	Incoming Call Route and User Rights entries only.
Read Only	View	View all configuration entries.
	Edit	None.
	New	
	Delete	

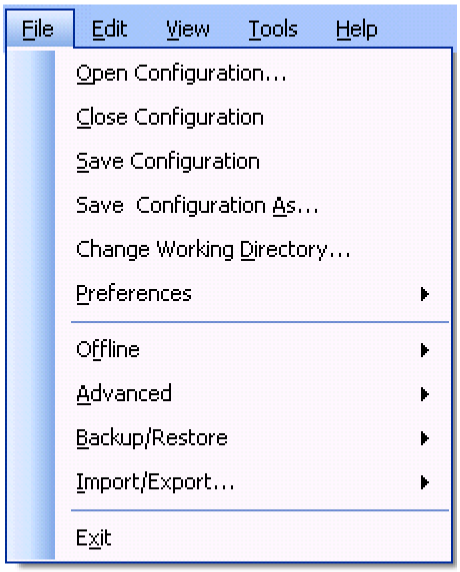
2.3 Title Bar

The Manager title bar shows several bits of information.

- The Manager application version.
- The system name of the system from which the currently loaded configuration was received.
- The software level of the system's control unit.
- For IP Office 3.2+ systems, the service user name used to receive the configuration and that user's associated operator rights. For pre-3.2 systems this is replaced with just the operator name.

2.4 The Menu Bar





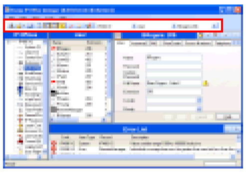
Details of all the options that may be available within the Menu Bar drop downs are contained in the section Menu Bar Commands 102.

The commands are context sensitive. Commands are grayed out when not useable.

For some commands, an ▶ symbol indicates that there are sub-commands from which a selection can be made.

File	Open Configuration... 104		
	Close Configuration 104		
	Save Configuration 104		
	Save Configuration As... 104		
	Change Working Directory... 105		
	Preferences 106		
	Offline	Create New Config 112	
		Open File... 112	
		Send Config... 112	
		Receive Config... 112	
	Advanced	Erase Configuration (Default) 113	
		Reboot 113	
		System Shutdown... 114	
		Upgrade.. 115	
		Change Mode... 116	
		Audit Trail... 118	
		Security Settings... 119	
		Erase Security Settings... 119	
		Embedded File Management 119	
		Format IP Office SD Card 121	
		Recreate IP Office SD Card 122	
		Memory Card Command 125	Shutdown... 125
			Start Up... 125
		Voicemail Pro 124	
		System Status 124	
		LVM Greeting Utility 124	
	Backup/Restore	Backup Binaries and Configurations 125	
		Restore Binaries and Configurations 125	
	Import/Export	Import 127	
		Export 126	
	Exit 127		
View	Toolbars 128		
	Navigation Pane 128		
	Group Pane 128		
	Details Pane 128		
	Error Pane 128		
	Simplified View 128		
	TFTP Log 129		
Tools	Extension Renumber 130		
	Line Renumber 130		
	Connect To 130		
	SCN Service User Management 130		
	Busy on Held Validation 131		
	MSN 132 Configuration 132		
	Print Button Labels 133		

2.5 Toolbars



The following toolbars are described here:

- Main Toolbar.
- Navigation Toolbar.
- Details Toolbar.

The Main Toolbar

This toolbar is also available when Manager is in security mode. However many of the controls will not function in security mode.

- Open Configuration from a System 55
  Advertises to the address currently shown in the Manager's title bar for any available systems. A list of responding systems is then displayed. When a system is selected from this list, a valid user name and password must be entered. Equivalent to File | Open Configuration.

- Open Configuration File 55
  Open a configuration file stored on a PC. The button can be clicked to display a browse window. Alternatively the adjacent ▾ arrow can be used to drop-down a list of the last 4 previously opened configuration files. Equivalent to File | Offline | Open File.

- Save Configuration File 64
  The action of this icon depends on whether the currently loaded configuration settings were received from a system or opened from a file stored on PC. If the former applies, the menu sending the configuration back to the system is displayed. In the latter case, the file changes are saved to the original file. Equivalent to File | Save Configuration.

- Collapse All Groups 40
  Causes all ⊟ symbols in the navigation pane to be collapsed to ⊞ symbols.

- Show / Hide the Navigation Pane 40

- Show / Hide the Group Pane 41

- Show / Hide the Error Pane 45

- Validate Configuration 45
  Runs a validation on all the currently loaded configuration settings. The results appear in the error pane. By default the configuration is automatically validated when loaded and changes are validated when made, however the validation preferences can be changed through File | Preferences | Validation 112.

- Create New Configuration 60
  Runs a series of dialogs that create a new configuration from scratch.

- Connect To 130
  For a standalone system, start the process of adding it to a Small Community Network.

- Voicemail Pro Client 124
  Launch the Voicemail Pro client if also installed on the Manager PC.

The Navigation Toolbar

This toolbar provides drop down lists which can be used to navigate to particular entries in the configuration settings. The selected options in the navigation pane, group pane and the details pane are synchronized with the navigation toolbar and vice versa. This toolbar is particularly useful if you want to work with the group pane and or navigation pane hidden in order to maximize the display space for the details pane.

This toolbar is not available when Manager is in security mode 68.

Details Toolbar

This toolbar is shown in the top-right of the details pane.

- Create New Record
  The arrow is used to select the entry type to be created. For example; when adding an extension clicking may allow selection of a VoIP Extension or IP DECT Extension.
- Delete Entry
- Validate Entry
- < > Show Previous/Next Entry

Moving to the Previous or Next Entry

1. Click < or > at the top-right to move to the previous or next entry.

Altering the Toolbars

Showing or Hiding Toolbars

The different toolbars can be hidden if not required.

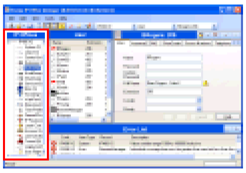
1. Select **View** and then **Toolbars**. Those toolbars currently shown are indicated by a tick mark.
2. To show or hide a toolbar, click on its name.

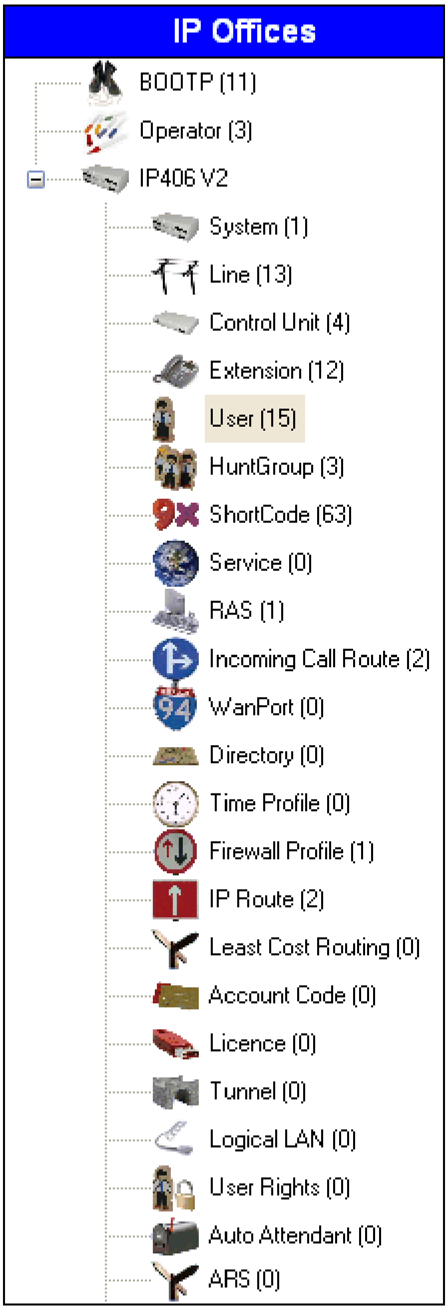
Moving Toolbars

The position of the Manager toolbars can be moved. Note that when moving a toolbar, the other toolbars and panes may adjust their size or position to ensure that all the toolbar icons remain visible.

1. Place the cursor over the end of the toolbar.
2. When the cursor changes to a four-way arrow, click and hold the cursor.
3. Move the toolbar to the required position and release the cursor.

2.6 Using the Navigation Pane





This pane shows icons for the different types of entry that the configuration can contain. Each type is followed by the number of entries of that type already in the configuration.

Selecting an icon displays the matching entries in the group pane and navigation toolbar.

Where ⊞ or ⊟ icons appear in the pane, they allow the structure to be expanded or collapsed. When the group pane is hidden, ⊞ and ⊟ icons are shown for each entry type and allow the entry type to be expanded to display all the existing entries of that type.

The icon in the main toolbar can also be used to collapse all the expanded entry types shown in the navigation pane.

The BOOTP and Operator icons are special. They represent settings stored on the Manager PC rather than configuration settings received from a system.

When Manager is used in security mode, this pane is also used by Manager in security mode 68 to display entries for security settings.

Navigation Pane Actions

Moving the Border Between the Panes

The border between the visible panes can be adjusted. Note that this is a proportional rather than exact position. If the whole window size is altered, the border position may also move.

1. Place the cursor over the border between two panes.
2. When the cursor changes to a double headed arrow with a bar through it, click and hold the cursor.
3. Drag the border to the required position and release the cursor.

Showing or Hiding Panes

The navigation pane can be shown or hidden. To do this use either of the following methods.

1. From the main toolbar, use the icon or select View. Those panes currently shown are indicated by a tick mark.
2. To show or hide the navigation pane, click on its name.

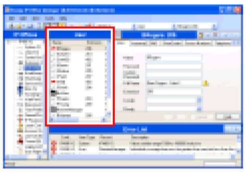
Changing the Size of Configuration Icons

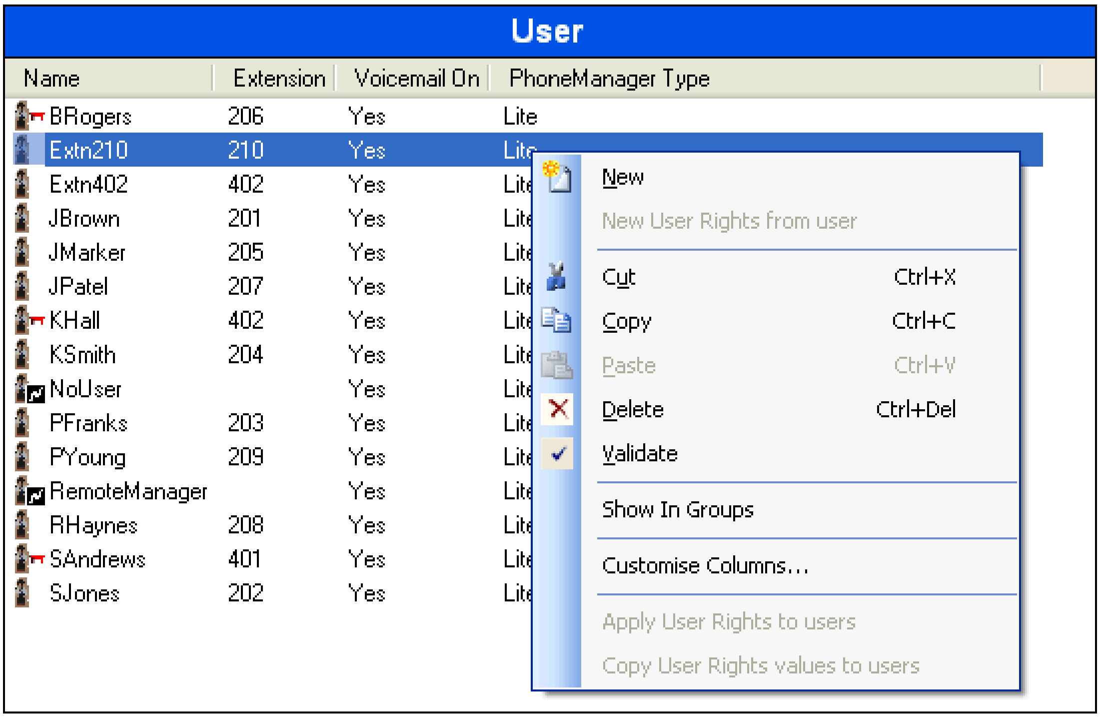
The size of the icons used on the navigation pane and details pane can be adjusted.

1. Select File and then Preferences.
2. Select the Visual Preferences tab.
3. Select the required icon size from Small, Medium or Large.
4. Click OK.

2.7 Using the Group Pane

This pane lists all the entries that match the type selected in the navigation pane or navigation toolbar. The list can be sorted by clicking on a column heading. Selecting an entry in this pane displays its details in the details pane.





The icons used in the pane may vary according to the state of the entry. For example, some of the users shown in this example have been configured for hot desking. This pane is also used by Manager in security mode 68 to display entries for security settings.

Group Pane Actions

Sorting the List
The entries shown in the group pane can be sorted using any of the columns displayed.

1. To sort the list using the details in a particular column, click on the column header.
2. Clicking on the same column header again reverses the sort order.

Customizing the Columns Displayed
For each entry type, which details are shown in the group pane can be customized. Also the order of the column can be adjusted. For details of the options available for each type of entry see Appendix D: Miscellaneous.

1. Right-click on the pane and select **Customize Columns**.
2. To add a column, selects its name in the left-hand Available Columns list and click >> to move it to the right-hand **Selected Columns** list.
3. To remove a column, select its name in the right-hand **Selected Columns** list and click << to move it to the left-hand **Available Columns** list.
4. To change the order of the **Selected Columns**, click on a column name and use the ^ and V controls.
5. Click OK.

Changing the Column Widths

1. In the column headers, place the cursor over the border between two columns.
2. When the cursor changes to a double headed arrow with a bar through it, click and hold the cursor.
3. Drag the border to the required position and release the cursor.

Adding a New Entry
The group pane can be used to add a new entry of the type currently displayed.

1. Right-click on the pane and select **New**.
   - A ▶ arrow symbol next to **New** indicates that you can select a particular type of new entry to create. Click the arrow and select an option from the list.
2. Use the details pane to configure the new entry.
3. Click OK in the details pane.

Deleting an Entry
1. Select the entry to be deleted by clicking on it.
2. Right-click on the pane and select Delete.

Validating an Entry
1. Select the entry to be validated by clicking on it.
2. Right-click on the pane and select Validate.

Show in Groups
This command groups the items shown in the group pane. The grouping method will vary depending on the entry type being listed. For example, short codes are grouped based on short code feature type such as all forwarding short codes together.

1. Right-click on the pane and select **Show In Groups**.

Moving the Border Between the Panes
The border between the visible panes can be adjusted. Note that this is a proportional rather than exact position. If the whole window size is altered, the border position may also move.

1. Place the cursor over the border between two panes.
2. When the cursor changes to a double headed arrow with a bar through it, click and hold the cursor.
3. Drag the border to the required position and release the cursor.

Showing or Hiding Panes
The group pane can be shown or hidden. To do this use either of the following methods.

1. From the main toolbar, use the  icon.

or

1. Select **View**. Those panes currently shown are indicated by a tick mark.
2. To show or hide the group pane, click on its name.

Changing the Size of Configuration Icons
The size of the icons used on the navigation pane and details pane can be adjusted.

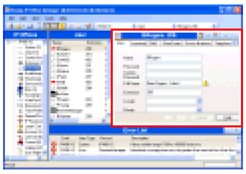
1. Select File and then **Preferences**.
2. Select the **Visual Preferences** tab.
3. Select the required icon size from **Small**, **Medium** or **Large**.
4. Click OK.

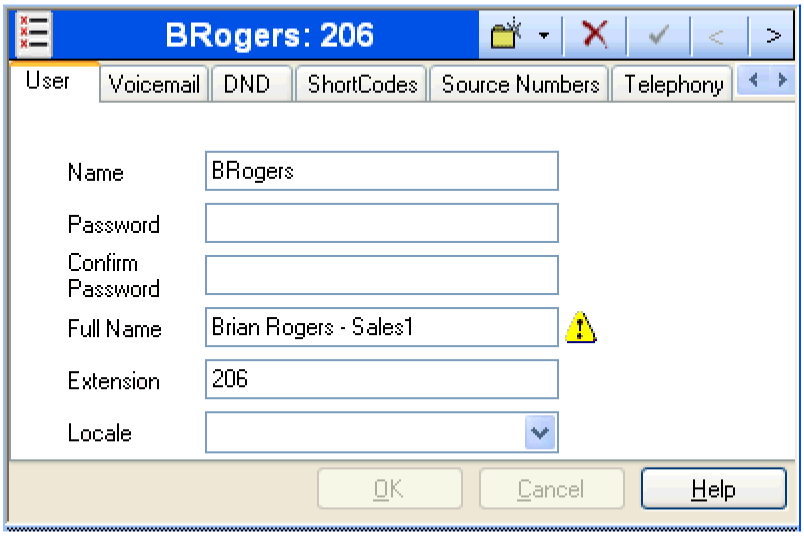
2.8 Using the Details Pane

Whenever a selection is made through the group pane or the navigation toolbar, the settings for the matching entry are shown in the details pane. This pane is also used by Manager in security mode 68 to display entries for security settings.

The details are grouped into tabs. The tabs available may vary depending on what particular type of entry is being viewed. For example, for extension entries the Analog tab is only shown for analog extensions.

Individual settings may also be grayed out. This indicates that they are either for information only or that they cannot be used until another setting is enabled.





The top-left icon indicates the following:

Locked
Indicates that you can view the settings but cannot change them.

Editable
Indicates that you can change the settings if required.

Changed
Indicates that the settings have been changed since the tab was opened. Click OK to save the changes or Cancel to undo.

Various icons may appear adjacent to settings:

Locked Setting
The setting cannot be changed through this tab. This icon appears on user settings where the user is associated with User Rights that controls the setting.

Information
Indicates a value which does not have to be set but may be useful if set.

Warning
A warning indicates a configuration setting value that is not typical and may indicate misconfiguration.

Error
An error indicates a configuration setting value that is not supported by the system. Such settings may cause the system to not operate as expected.

Details Pane Actions

Editing an Entry

1. The method of entering an entry varies as different fields may use different methods. For example text entry boxes or drop down lists.

2. By default when changes are made, they are validated once another field is selected. See File | Preferences | Validation 112.

3. Clicking on OK at the base of the details pane to accept the changes or click on Cancel to undo the changes.

Adding a New Entry

1. Click at the top-right of the details pane.

2. Select the type of entry required. For example, with services you can select from *Normal*, *WAN* or *Intranet*.

Deleting an Entry

1. Click at the top-right of the details pane.

Validating an Entry

1. Click at the top-right of the details pane.

Moving to the Previous or Next Entry

1. Click < or > at the top-right to move to the previous or next entry.

Selecting a Tab

1. To view the detail stored on a particular tab, click on the name of that tab.

2. If the tab required is not shown, use the controls if shown on the right to scroll through the available tabs. The tabs available may vary depending on what particular type of entry is being viewed.

Changing the Position of the Details Pane
When the group pane is visible, the details pane is shown either below it or to its right. This position can be adjusted.

1. Select View and then Details Pane.

2. The current position setting is indicated by a tick mark.

3. To select a position, click on it.

Changing How the Tabs Display

For entries with more than two tabs, you can select whether Manager should use controls or arrange the tabs as multiple rows when necessary.

1. Select Files | Preferences | Visual Preferences.

2. Select Multi-Line Tabs.

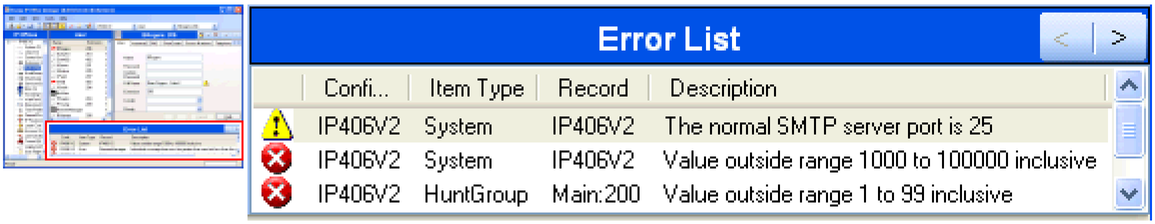
3. Click OK.

2.9 Using the Error Pane

Validation is a process where Manager checks configuration entries for errors or for values for which it regards as requiring a warning. The results of this checking are shown by icons next to the field that caused the error or warning, All errors and warnings are also listed in the Error Pane.

By default validation is performed automatically whenever a configuration file is opened and when any field is edited. However, if required, the use of automatic validation can be controlled through the settings on the File | Preference | Validation 112 tab.

The validation process can be run manually using the icon for the whole configuration or the icon for a particular entry.



The icons used for errors and warnings are:

Error
An error indicates a configuration setting value that is not supported by the system. Such settings are likely to cause the system to not operate as expected.

Warning
A warning indicates a configuration setting value that is not typical and may indicate misconfiguration.

Information
Typically indicates a setting which may be useful to set.

Error Pane Actions

Revalidating Configuration Settings
By default, the configuration is validated when loaded and individual entries are revalidated when changed.

1. To validate the whole configuration, click in the main toolbar.

2. For a particular entry, click in the details pane.

Jumping to an Error or Warning

1. Clicking on an error or warning in the error pane will load the matching entry tab into the details pane.

2. The < and > can be used to move to the next error or warning in the error pane.

Showing or Hiding Panes
The error pane is automatically displayed if a configuration containing errors or warnings is loaded into Manager. However it can be manually shown or hidden using either of the following methods.

1. From the main toolbar, use the icon.

or

1. Select **View**. Those panes currently shown are indicated by a tick mark.

2. To show or hide the error pane, click on its name.

2.10 Altering the Interface

The Manager configuration settings interface can be customized in a number of ways. These changes are remembered the next time Manager is started.

Resizing the Manager Window

When the Manager window is not maximized or minimized, it size can be adjusted.

1. Place the cursor over the edge of the current window.
2. When the cursor changes to a double-headed arrow, click and hold the cursor.
3. Drag the edge to the required position and then release the cursor.

Moving the Border Between the Panes

The border between the visible panes can be adjusted. Note that this is a proportional rather than exact position. If the whole window size is altered, the border position may also move.

1. Place the cursor over the border between two panes.
2. When the cursor changes to a double headed arrow with a bar through it, click and hold the cursor.
3. Drag the border to the required position and release the cursor.

Showing or Hiding Toolbars

The different toolbars can be hidden if not required.

1. Select **View** and then **Toolbars**. Those toolbars currently shown are indicated by a tick mark.
2. To show or hide a toolbar, click on its name.

Moving Toolbars

The position of the Manager toolbars can be moved. Note that when moving a toolbar, the other toolbars and panes may adjust their size or position to ensure that all the toolbar icons remain visible.

1. Place the cursor over the end of the toolbar.
2. When the cursor changes to a four-way arrow, click and hold the cursor.
3. Move the toolbar to the required position and release the cursor.

Showing or Hiding Panes

The details pane cannot be hidden. The navigation pane, group pane and error pane can be shown or hidden. To do this use either of the following methods.

1. From the main toolbar, use the following icons:
   -  **Hide/Show Navigation Pane.**
   -  **Hide/Show Group Pane.**
   -  **Hide/Show Error Pane.**

or

1. Select **View**. Those panes currently shown are indicated by a tick mark.
2. To show or hide a pane, click on its name.

Changing the Position of the Details Pane

When the group pane is visible, the details pane is shown either below it or to its right. This position can be adjusted.

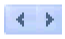
1. Select View and then Details Pane.
2. The current position setting is indicated by a tick mark.
3. To select a position, click on it.

Changing the Size of Configuration Icons

The size of the icons used on the navigation pane and details pane can be adjusted.

1. Select File and then Preferences.
2. Select the Visual Preferences tab.
3. Select the required icon size from Small, Medium or Large.
4. Click OK.

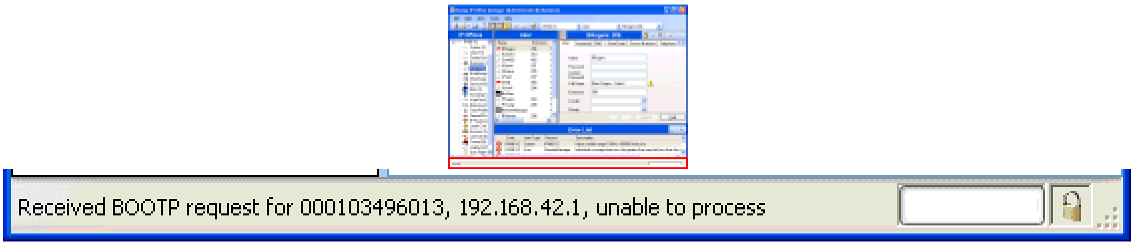
Changing How the Tabs Display

For entries with more than two tabs, you can select whether Manager should use  controls or arrange the tabs as multiple rows when necessary.

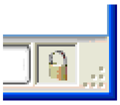
1. Select Files | Preferences | Visual Preferences.
2. Select Multi-Line Tabs.
3. Click OK.

2.11 The Status Bar

The status bar at the base of the Manager screen is used to display icons and messages about communications between Manager and systems. If the Manager is also acting as a BOOTP and TFTP server it will also show BOOTP and TFTP messages.



A padlock icon is displayed whenever the Manager communications settings are set to secure. This indicates all attempted configuration and security settings exchanged will be attempted over a secure TLS link:



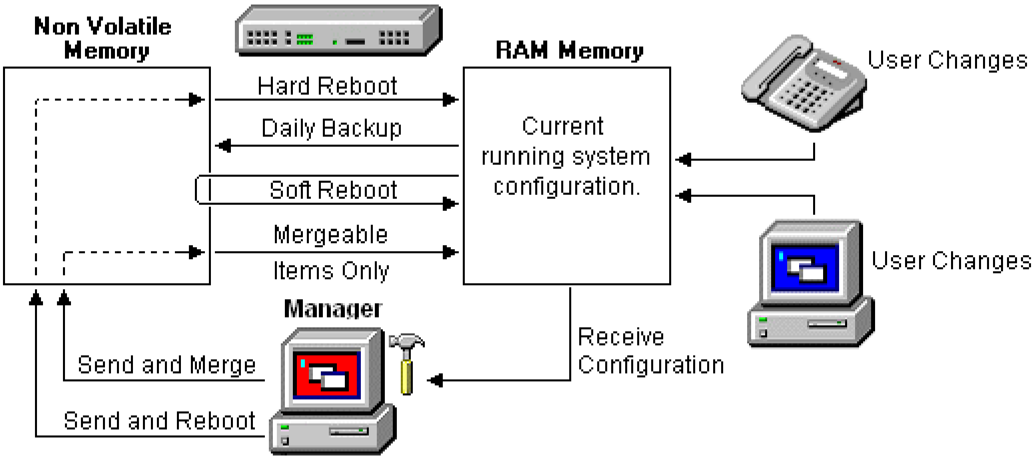
Status bar messages display information about communications the Manager application receives. Some typical status bar messages are listed below.

- Ready
  This message is normally seen when Manager has just started and no configuration has been received.
- Received BOOTP request for 001125465ab2, unable to process
  Manager is acting as a BOOTP server. It has received a BOOTP request that does not match a system listed in its BOOTP entries. The cause may be a device or application, other than an Manager, that also uses BOOTP.
- TFTP: Received TFTP Error "NotFound" from 192.168.42.1
  An attempt to receive settings from or send settings to the system failed. The most probable cause is a name or password error.
- TFTP: Received 17408 bytes for Marks_Test
  Manager has received configuration settings from the named system using TFTP.
- Sent 100% of C:\Program Files\Avaya\IP Office\Manager\b10d01b2_3.bin
  Manager has sent the indicated file in response to a BOOTP request.

2.12 Editing Configuration Settings

Before editing the system's configuration settings, it is important to understand how those settings are stored and used by the system.

- The control unit holds copies of its configuration in both its internal non-volatile and RAM memory. A copy is also held on the System SD card *(IP500v2)*.
- The copies in non-volatile memory and System SD card, are retained even if power to the control unit is removed.
- During power up, the system loads the configuration file stored on the System SD card into its RAM memory. Other systems load the configuration stored in non-volatile memory into RAM memory. The copy in RAM memory is then used to control the system's operation.
  - If the system encounters a problem using the configuration file in its System SD card's /primary folder, it attempt to use the copy in its non-volatile memory. For fully details of the IP500v2 boot process and SD card usage refer to the Manager Installation Manual.
- Users actions such as changing their forward destinations or mailbox passcode are written to the configuration in RAM memory.
- Changes made using Manager are written to the configuration in non-volatile memory and then copied into the RAM memory and System SD.
- Between 00:00 and 00:30, a daily backup occurs which copies the configuration in the system's operation RAM memory back into its non-volatile memory and, on IP500v2 system's, the System SD card. For IP Office Release 6.1 and higher: On IP500v2 system, the contents of the system memory cards /primary folder can then also be automatically copied to the /backup folder by enabling System | System | Automatic Backup Command 143.
- When the system is shutdown using the correct shutdown method 114, the configuration in RAM memory is copied to the non-volatile memory and System SD card.



Using Manager

When using Manager to edit the configuration settings, the following need to be remembered:

- Manager receives the current configuration settings from RAM memory. Therefore the configuration is receives includes any changes made by users up to that time. However it will not contain any subsequent changes made by users.
- When sending the configuration settings back to the system, Manager allows two choices, reboot or merge.
  - Reboot sends the configuration to the system's non-volatile memory along with an instruction to reboot. Following the reboot, the new configuration in non-volatile memory is copied to the RAM memory and used.
  - Merge sends the configuration to the system's non-volatile memory without rebooting. The system then copies those changes that are mergeable into the RAM memory. A key point here is that not all configuration settings are mergeable, see the Mergeable Settings 51 list.

As a result of the above, it is important to bear the follow scenarios in mind:

- Changes made by users after a configuration is received by Manager may be lost when the configuration is sent back from Manager. Therefore it is preferable to always edit a recently received copy of the configuration rather than one that has been open for a period of time.
- If a merge is attempted with non-mergeable changes, those items will be written to the non-volatile memory but will not be copied to RAM memory. If a daily backup occurs, they will then be overwritten by the RAM. If a power loss reboot occurs, they will be written to RAM memory.

2.12.1 Mergeable Settings

The table below shows the configuration entries for which changes can be merged and those that require a system reboot. The Send Configuration menu shown when sending a configuration to the system automatically indicates when the configuration is mergeable.

Mergeable	3.2+	Pre-3.2
System 142	–	–
- System 143	✓ *1	✗
- LAN1/LAN2 148	✗	✗
- DNS 156	✗	✗
- Voicemail 157	✓ *2	✗
- Telephony 160	✓ *3	✗
- VoIP 150	✗	✗
- LDAP 169	✗	✗
- System Events 172	✗	✗
- CDR/SMDR 179	✓	✗
- Twinning 177	✓	–
Line 185	✗	✗
Control Unit 250	✗	✗
Extension 251	✗	✗ *4
User 262	✓	✓
Hunt Group 304	✓	✓
Short Code 330	✓	✓
Service 331	✓	✓
RAS 340	✓	✓
Incoming Call Route 342	✓	✓

Mergeable	3.2+	Pre-3.2
WAN Port 350	✗	✗
Directory 358	✓	✓
Time Profile 360	✓	✗
Firewall Profile 362	✓	✓
IP Route 367	✓	✓
Least Cost Route 370	✓	✓
Account Code 375	✓	✓
License 378	✓	✓
Tunnel 380	✗	✗
Logical LAN 385	✗	✗
Wireless 386	✗	✗
User Rights 389	✓	✓
Auto Attendant 400	✓	✗
Authorization Code 404	✓	✗
ARS 407	✓	–
E911 System 421	✗	✗

- *1 - 3.2+ | System | System
  Changes to Locale, License Server IP Address and Favor RIP Routes over Static require a reboot.
- *2 - 3.2+ | System | Voicemail
  Changes to Voicemail Type require a reboot.
- *3 - 3.2+ | System | Telephony
  Changes to Companding LAW and Busy Tone Detection require a reboot.
- *4 - 4.1+ | Extension
  Base Extension and Disable Speakerphone are mergeable.

2.12.2 Configuration Sizes

There are maximum size limits to the configuration file that can be loaded into a control unit. They are:

Control Unit	Maximum Configuration File Size
Small Office Edition	192KB
IP403	192KB
IP406 V1	192KB
IP406 V2	384KB
IP412	1.0MB

IP500	1.0MB
IP500v2	2.0MB

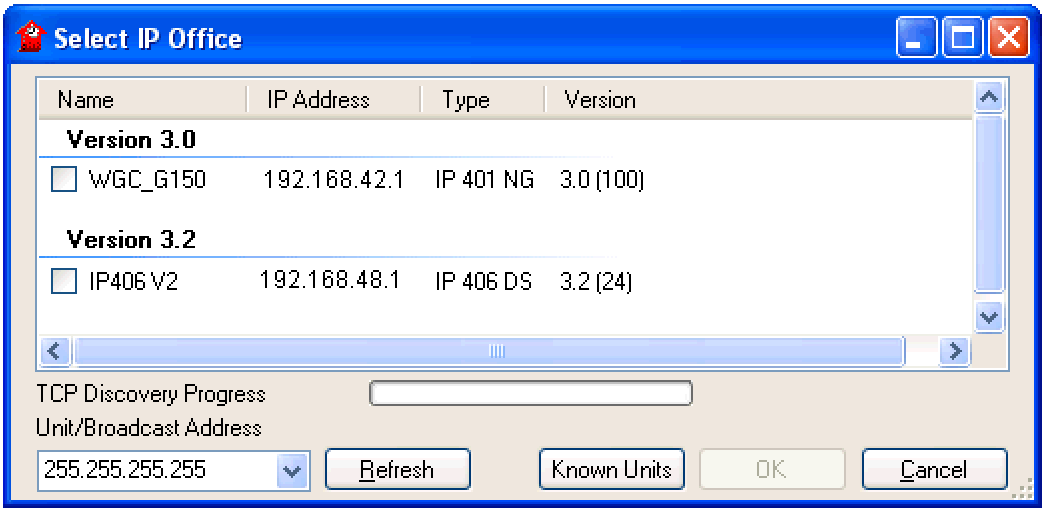
- When you attempt to save a configuration that is too large, you will be prompted and the save is canceled.
- During normal operation, additional configuration entries can be added to the configuration without using Manager (for example call log entries and directory entries made from phones). If, during the overnight backup to flash memory 49, the configuration if found to be too large, entries will be removed until the configuration is sufficiently small to be backed up. The entries removed are call log records, system directory records and then personal directory in that order. Note that those entries will still exist in the configuration running the system in its RAM memory, however if the system is restarted they will disappear as the configuration is reloaded from the Flash memory.

Figures for all individual entries in the configuration cannot be given as they vary. The list below gives typical values, in bytes, for common entries:

Physical Extension: 70.	Intranet Service: 240.	Firewall Profile: 40.
IP Extension: 70.	WAN Service: 400.	Custom Firewall Entry: 80.
User: 170.	RAS Service: 110.	IP Route (Static): 30.
User Short Code: 40.	Incoming Call Route: 30.	License Key: 40.
DSS Button: 20.	WAN Port (PPP): 70.	Account Code: 40.
Hunt Group: 100.	WAN Port (FR): 120.	Logical LAN: 60.
Hunt Group member: 10.	Directory Entry: 70.	Tunnel (L2TP): 200.
System Short Code: 10.	Time Profile: 40.	Tunnel (IPSec): 110.
Normal Service: 220.	Time Profile Entry: 20.	

2.12.3 Setting the Discovery Addresses

By default, when  or File | Open configuration is selected, Manager's Select IP Office menu appears. It performs a UDP broadcast to the address 255.255.255.255. This broadcast will only locate systems that are on the same network subnet as the PC running Manager.



The process above is called discovery. A UDP broadcast will not be routed to other networks and subnets. Therefore to find systems not located on the same subnet as the Manager PC, the following other options are supported.

- Specific Addressing
  The Unit/Broadcast Address shown on the Select IP Office menu can be changed to the specific IP address of the required system. A single address is routable and so can be used to discover a system on another subnet.

- TCP Discovery Address Ranges
  IP Office 3.2+ systems support discovery by TCP as well as UDP. A set of TCP addresses and address ranges can be specified for use by the Select IP Office discovery process.

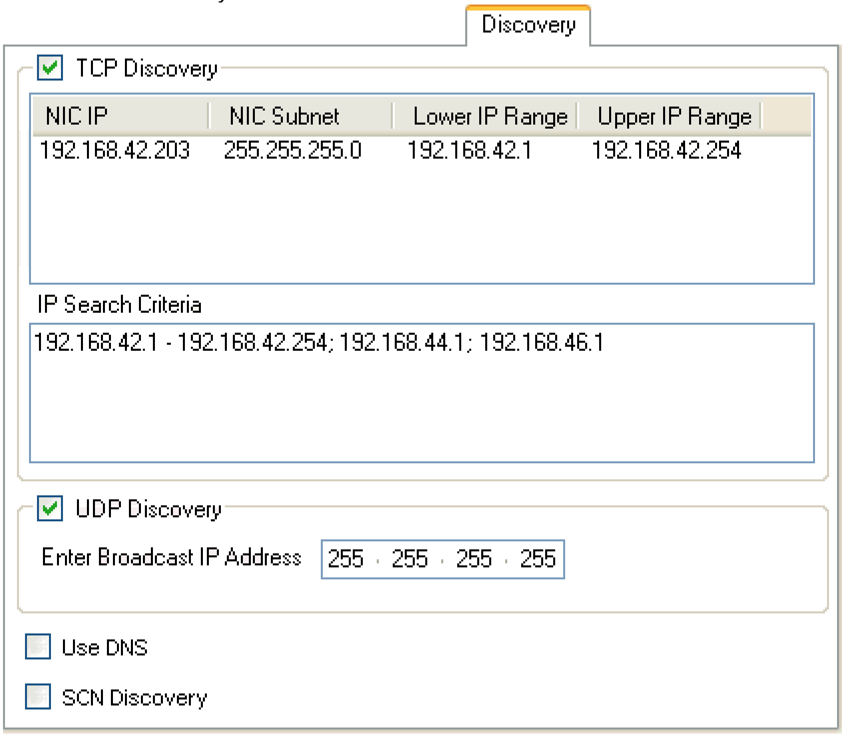
- Known System Discovery 59
  Manager can write the details of systems it discovers to a file. The list of systems in that file can then be used for access to those systems. See Known System Discovery 59.

- DNS Lookup
  Manager can be configured to locate systems using DNS name lookup. This requires the systems on a customer network to be added as names on the customer's DNS server and the Manager PC to be configured to use that server for DNS name resolution. The use of DNS is configured through File | Preferences | Discovery 108.

Changing the Initial Discovery Settings

The Discovery tab of the Manager Preferences menu can be used to set the UDP and TCP addresses used by the discovery process run by the Select IP Office menu.

1. Select File | Preferences menu.
2. Select the Discovery tab.

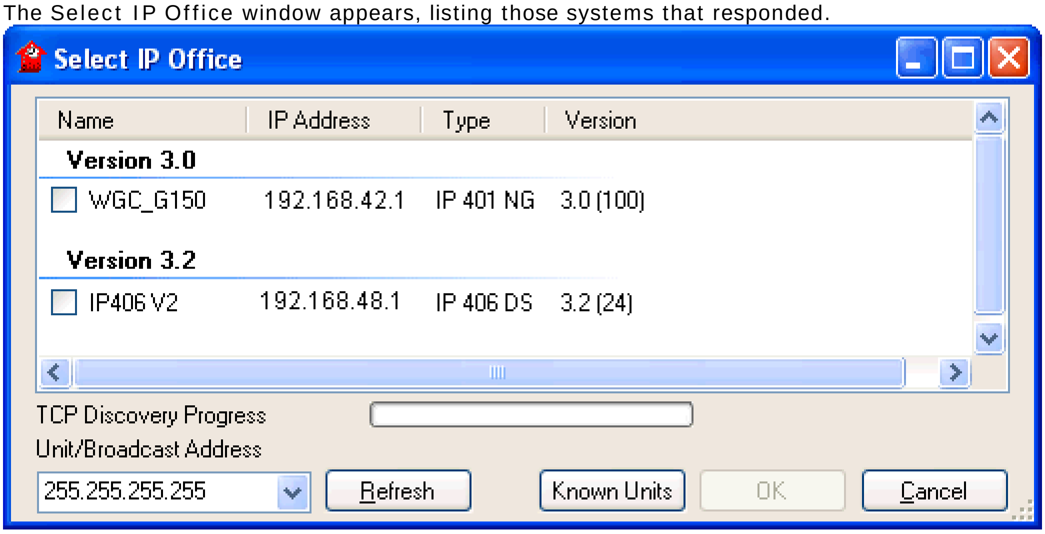


- TCP Discovery: *Default = On. Software level = 3.2+.*
  This setting controls whether Manager uses TCP to discover systems. The addresses used for TCP discovery are set through the IP Search Criteria field below.
  - NIC IP/NIC Subnet
    This area is for information only. It shows the IP address settings of the LAN network interface cards (NIC) in the PC running Manager. Double-click on a particular NIC to add the address range it is part of to the IP Search Criteria. Note that if the address of any of the Manager PC's NIC cards is changed, the Manager application should be closed and restarted.
  - IP Search Criteria
    This section is used to enter TCP addresses to be used for the TCP discovery process. Individual addresses can be entered separated by semi-colons, for example *135.164.180.170; 135.164.180.175*. Address ranges can be specified using dashes, for example *135.64.180.170 - 135.64.180.175*.
- UDP Discovery: *Default = On*
  This settings controls whether Manager uses UDP to discover systems. Pre-3.2 systems only respond to UDP discovery. By default IP Office 3.2 and higher systems also respond to UDP discovery but that can be disabled through the system's security settings.
  - Enter Broadcast IP Address: *Default = 255.255.255.255*
    The broadcast IP address range that Manager should used during UDP discovery. Since UDP broadcast is not routable, it will not locate systems that are on different subnets from the Manager PC unless a specific address is entered.
- Use DNS: *Software level = Manager 6.2+.*
  Selecting this option allows Manager to use DNS name (or IP address) lookup to locate a system. Note that this overrides the use of the TCP Discovery and UDP Discovery options above. This option requires the system IP address to be assigned as a name on the users DNS server. When selected, the Unit/Discovery Address field on the Select IP Office 55 dialogue is replaced by a Enter Unit DNS Name or IP Address field.
- SCN Discovery: *Software level = Manager 8.1+.*
  If enabled, when discovering systems, the list of discovered systems will group systems in the same Small Community Network and allow them to be loaded as a single configuration. At least one of the systems in the SCN must be running Release 6.0 or higher software. See SCN Management 817. This does not override the need for each system in the SCN to also be reachable by the TCP Discovery and or UDP Discovery settings above and accessible by the router settings at the Manager location.

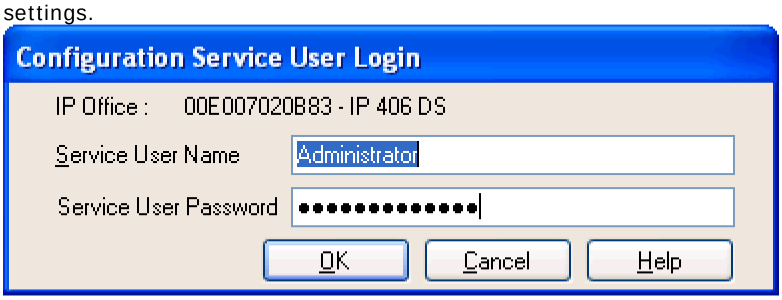
2.12.4 Opening a Configuration from a System

The initial IP address ranges in which Manager searches for systems are set through the Manager preferences (File | Preferences | Discovery 108). By default it scans the local network of the Manager PC.

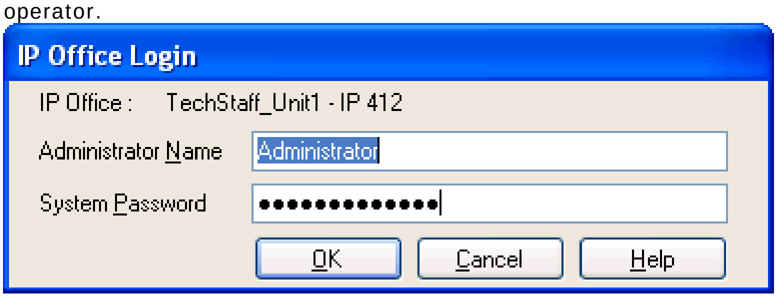
1. Start Manager. If Manager is already started and a configuration is open in it, that configuration must be closed first.
   - If Manager is set to Auto Connect on start up 106, it will scan for systems automatically and either display the list of systems discovered or automatically start login to the only system discovered.
   - Otherwise, click on or select File | Open Configuration.
2. The Select IP Office window appears, listing those systems that responded.



   - If Manager has been set with SCN Discovery 108 enabled, systems in a Small Community Network may be grouped and can be loaded into Manager's SCN management 817 mode.
   - If the system required was not found, the Unit/Broadcast Address used for the search can be changed. Either enter an address or use the drop-down to select a previously used address. Then click Refresh to perform a new search.
   - The address ranges used by Manager for searching can be configured through the File | Preferences | Discover 108 tab.
   - A list of known systems can be stored and used. See Known System Discovery 59.
   - Manager can be configured to search using DNS names. See the Use DNS 108 option.
   - Systems found but not supported by the version of Manager being used will be listed as *Not Supported*.
   - If the IP500v2 system detected is running software other than from its primary folder, a warning icon will be shown next to it. The configuration can still be opened but only as a read-only file.
3. When you have located the system required, check the box next to the system and click OK.
4. The system name and password request is displayed. Enter the required details and click OK.
   - IP Office 3.2 and higher systems:
     The name and password used must match a Service User account configured within the system's security settings.



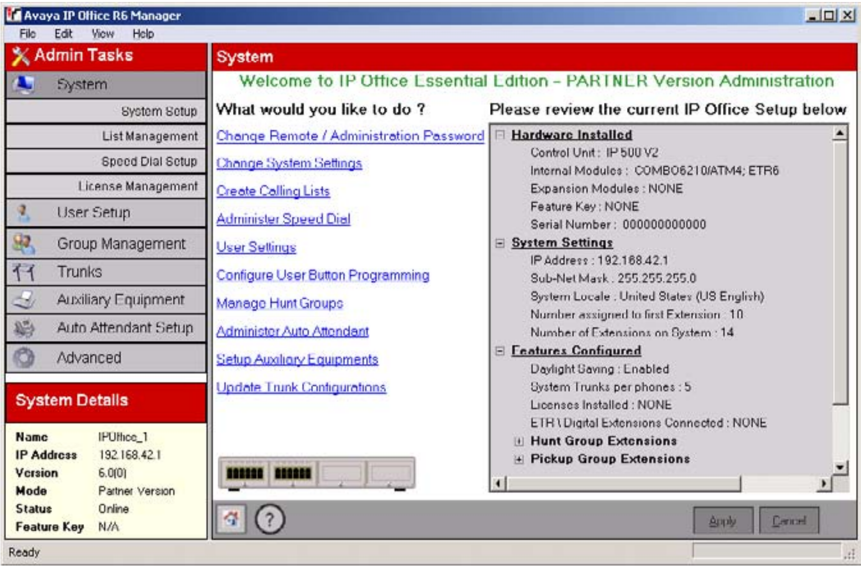
- Pre-3.2 IP Office Systems:
  The name must match a Manager operator and the password must match the system's system password. If the name does not match a Manager operator, the config will still be loaded by using the *Guest (read-only)* operator.



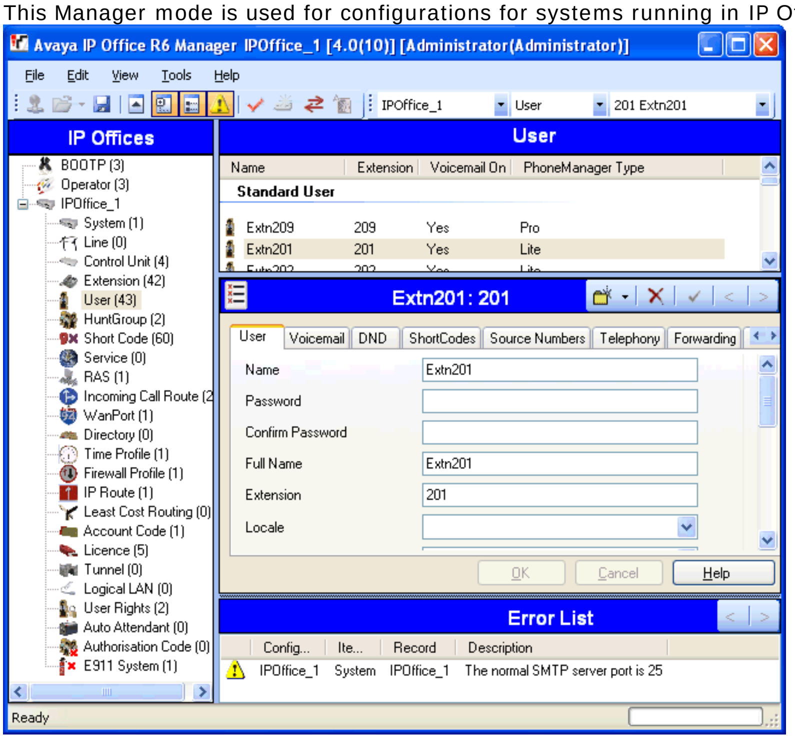
5. Additional messages will inform you about the success or failure of opening the configuration from the system.

6. Following a successful log in, the configuration is opened in Manager. The menus and options displayed will depend on the type of system configuration loaded.

   - PARTNER Version/Norstar Version Configuration
     This Manager mode is used for configurations from systems running in PARTNER Version or Norstar Version modes.



   - IP Office Configuration
     This Manager mode is used for configurations for systems running in IP Office Standard Version mode.



Login Messages

While attempting to login to a system, various additional messages may be displayed.

Configuration Not Loaded Messages

- **Access Denied**
  This is displayed as the cause if the service user name/password were incorrect, or the service user has insufficient rights to read the configuration. The Retry option can be used to log in again but multiple rejections in a 10 minute period may trigger events, such as locking the user account, set by the Password Reject Limit and Password Reject Action options in the systems security settings.

- **Failed to communicate with system**
  This is displayed as the cause if the network link fails, or the secure communication mode is incorrect (for example Manager is set to unsecured, but the system is set to secure only).

- **Account Locked**
  The account of the Service User name and password being used is locked. This can be caused by a number of actions, for example too many incorrect password attempts, passing a fixed expiry date, etc. The account lock may be temporary (10 minutes) or permanent until manually unlocked. An account can be enabled again through the system's security settings.

Additional Messages

- **Your service user account will expire in X days**
  This message indicates that an Account Expiry date has been set on the system service user account and that date is approaching. Someone with access to the system's security settings will be required unlock the account and set a new expiry date.

- **Your password will expire in X days. Do you wish to change it now?**
  This message indicates that password ageing has been configured in the system's security settings. If your password expires, someone with access to the system's security settings will be required to unlock the account.

- **Change password**
  Through the system's security settings, a service user account can be required to change their password when logging in. The menu provides fields for entering the old password and new password.

- **Contact Information Check - This configuration is under special control**
  This message will appear if a Manager user with administrator rights has entered their contact information into the configuration. For example to indicate that they do not want the configuration altered while a possible problem is being diagnosed. The options available are:

- **Cancel**
  Select this option to close the configuration without making any changes.

- **Set configuration alteration flag**
  Select this option if the configuration is being opened because some urgent maintenance action. When the configuration is next opened, the fact that it has been altered will be indicated on the System | System 143 tab.

- **Delete Contact Information**
  Select this option to take the system out of special control.

- **Leave contact information and flags unchanged *(Administrators only)***
  This option is only available to service users logging in with administrator rights.

2.12.5 Opening a Configuration Stored on PC

A configuration file previously saved on the PC can be reopened in Manager. This method of access does not require entry of a Service User name and password. All parts of the configuration are visible.

Use either of the following processes to load a saved configuration file:

1. Click the main toolbar or select **File | Offline | Open File** from the menu bar. If the files is one that has previously been opened offline, click the ▾ symbol next to in the main toolbar
2. An **Open configuration file** window appears. Use this to browse to the required configuration file.
3. Select the file and click **Open**.

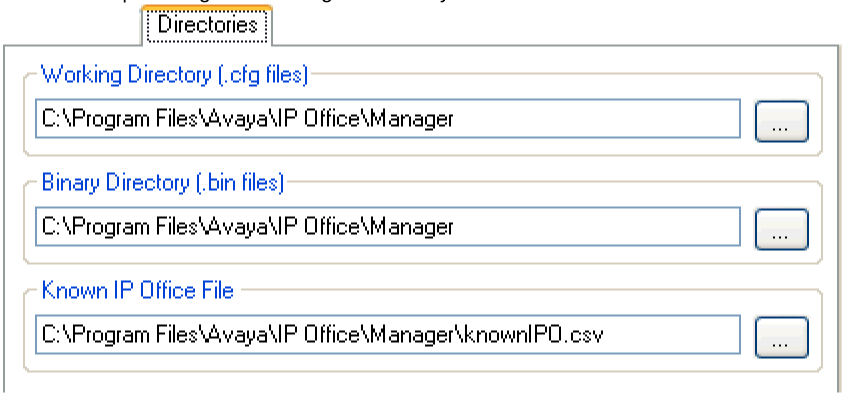
2.12.6 Known System Discovery

The Manager Select IP Office menu normally displays systems discovered by Manager using either UDP broadcast and or TCP requests (see Setting the Discovery Addresses 53). Manager can be configured to also record details of discovered units and then display a list of those previously discovered ('known') systems.

Configuring Manager for Known System Discovery

Use of known systems discovery is not enabled by default. The Manager must be configured for the feature with a file location to which it can store and retrieve know system details.

1. Select File | Change Working Directory.

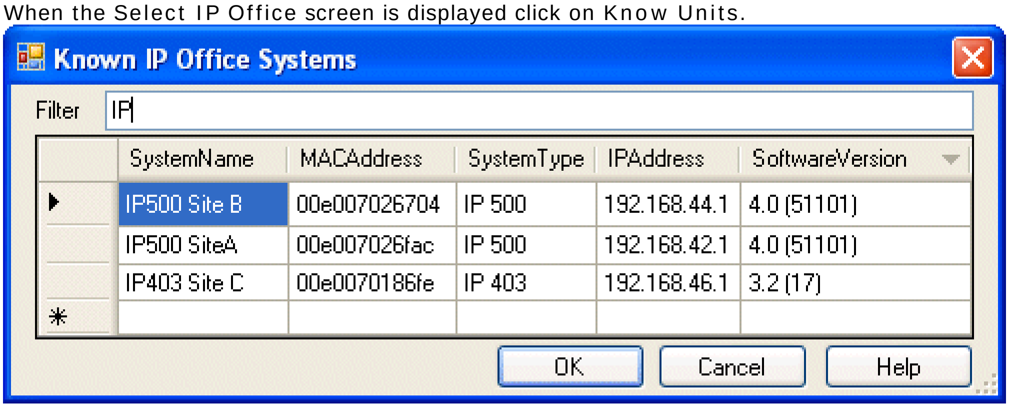


2. In the Known IP Office File field, enter the directory path and file name for a CSV file into which Manager can write details of the systems it discovers. If the file specified does not exist it will be created by Manager.

3. Click OK.

Using Know System Discovery

1. When the Select IP Office screen is displayed click on Know Units.



2. The screen displays the list of systems previously discovered and stored in the CSV file.

   - To select an control unit, highlight the row containing unit data and click OK. The selected unit will appear in the Select IP Office window.
   - To filter displayed units, type the first few characters of the unit name in the Filter field. Any unit whose name does not match the filter will be temporarily hidden.
   - Each discovery appends data to the known unit list. It is possible that details of some entries in the list may be out of date. Right clicking on the leftmost (grey) column of any row will bring up a floating menu offering the options of Refresh and Delete.
   - A new entry may be manually added without having to access the system first through normal discovery. Enter the IP address of the new system in the IP Address column of the blank row shown with a * and select Refresh from the floating menu. This will update the Known Units file with data relating to the unit with the specified address.
   - Select Cancel to return to the Select IP Office menu.

Note:

- The key used by the Known Systems CSV file is the IP address. The file cannot contain entries for separate systems that use the same IP address for access.
- The file can be made read only. In that case any attempts using Manager to update the file will be ignored.

2.12.7 Creating a Offline Configuration

Manager can be used to create a new configuration without connecting to a system. This allows the creation of a configuration prior to installation of the real system and so can be used to speed up installation.

- The configuration created must match the physical equipment in the system into which the configuration will be loaded. Doing otherwise may cause the system to reset and experience other problems.
- The Create Configuration tool includes all control units, external expansion modules and trunk cards supported. It is you responsibility to confirm what equipment is supported in your locale.

Creating a New Configuration

1. Click in the main toolbar or select File | Offline | Create New Config from the menu bar.
2. Select the Locale for the system. This defines a range of features such as default telephony settings. Click Next.
3. Fixed Length Numbering is supported for Manager 4.1+. The value can be *None* or *3* to *9*. Click Next.
    - If a value is selected, all default extension, user and hunt group extension numbers created by Manager will be that length. in addition Manager will display a warning if an extension number of a different length is entered when editing the configuration.
4. Select the type of control unit. Click Next.
5. Select the additional cards to include in the control unit. The number and type of cards selectable will depend on the control unit type. Click Next.
6. Select the external expansion modules to also include in the system. Click Next.
7. Click Finish.
8. The configuration is created and loaded into Manager.
9. Once this configuration has been edited as required it can be saved on the PC. In order to send it to the matching system, File | Offline | Send Configuration has to be used.

2.12.8 Importing and Exporting Settings

Manager can import configuration settings created elsewhere. This can be useful when setting up a new system or sharing common settings such as a directory between systems.

- The system supports LDAP 169 (System | Directory Services | LDAP 169) for automatic importation of directory records.
- IP Office Release 5.0+: The system also supports HTTP 171 (System | Directory Services HTTP 171) for automatic importation of directory records.

Settings are imported and exported in the following formats:

- Binary Files (.exp)
  These are non-editable files. During import and export it is possible to select what types of entries should be included in the file. During import the whole file is imported.
- Comma Separated Variable Text Files (.csv)
  These are plain text files. In addition to being exported from a system these files can be created and edited using programs such as WordPad or Excel.
  - When opening a .csv file in Excel it will alter the way some data is displayed, automatically changing the display format of dates and long numbers such as phone numbers.
  - UTF-8 Character Encoding
    Manager imports and exports CSV files using UTF-8 character encoding which uses a multiple bytes to support characters with diacritic marks such as ä. Other applications, such as Excel, may, depending on the user PC settings, use different encoding which will cause such characters to be removed or corrupted. Care should be taken to ensure that any tool used to create or edit the CSV supports all the characters expected and uses UTF-8 format.
    - Importing into Manager from Excel
      From Excel save the file as a .csv. This file will use ANSI character encoding. Open the file in Notepad and use the Save As option to rename the file and select UTF-8 encoding. Import the UTF-8 version of the file into Manager.
    - Exporting from Manager into Excel
      Do not double-click on the file exported from Manager. Start Excel and use File | Open to select the file. Excel will recognize that the file uses UTF-8 encoding and will start its text file importation wizard. Follow the wizard instructions and select comma as the field delimiter.

CSV File Formats

The format is CSV using commas as field separator, no text delimiters and no header row. The simplest way to check the required format for a CSV file prior to import, is to export and a file from an existing system configuration.

File Name	Fields in Order
Directory	Name, Number.
Hunt Group	Name, Extension, Group, Hunt, Rotary, Longest Waiting, Queuing On, Voicemail On, Broadcast, Voicemail Email.
Short Code	Code, Telephone Number, Feature.
User	Name, Extension, User Restriction/Rights, Voicemail Email address, Full Name *(IP Office Release 5.0+)* .
Configuration	Proprietary format. Note that this does not contain all configuration fields.

License	License *(ignored on import)*, License Key

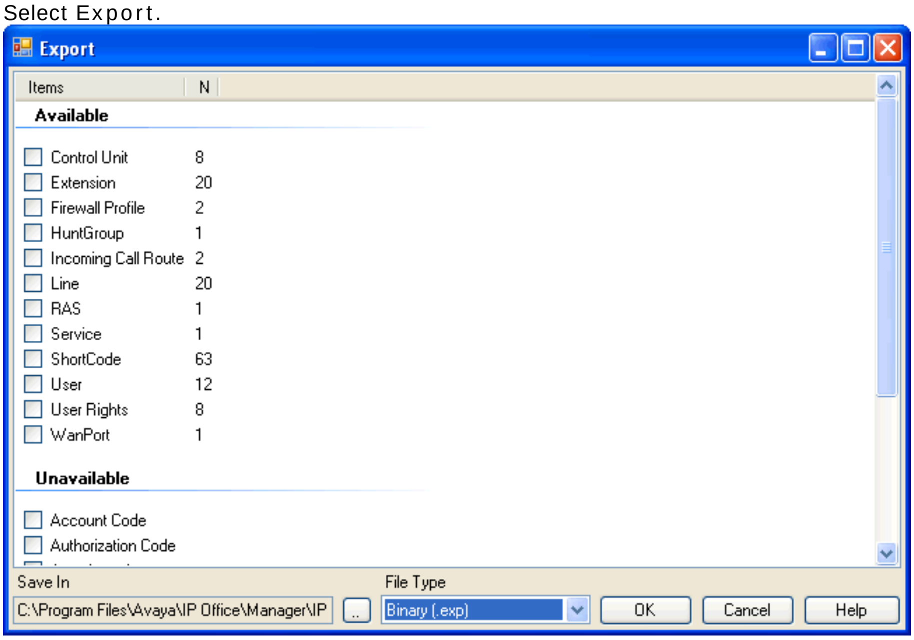
Notes:

- Hunt Group: Apart from Name, Extension and Voicemail Email, the fields use a 1 or 0 value for on or off.
- License:
  - The License field is for information only and is ignored during import.
  - Following import the License name will appear as invalid with Manager. To resolve this save and then reload the configuration file.
- The format of the system CSV is too complex to be described. It is a full export of all the system's configuration settings. This file format should only be used for export and import between systems and not for any offline editing.

Exporting Settings

1. Select File | Import/Export... from the menu bar.

2. Select Export.



3. Select the type of file. The list of exportable entry types will change to match the file type.

4. Select the types of items that should be exported.

5. Use the Save In path to select the location for the exported files. The default location used is sub-directory of the Manager application directory based on system name of the currently loaded system.

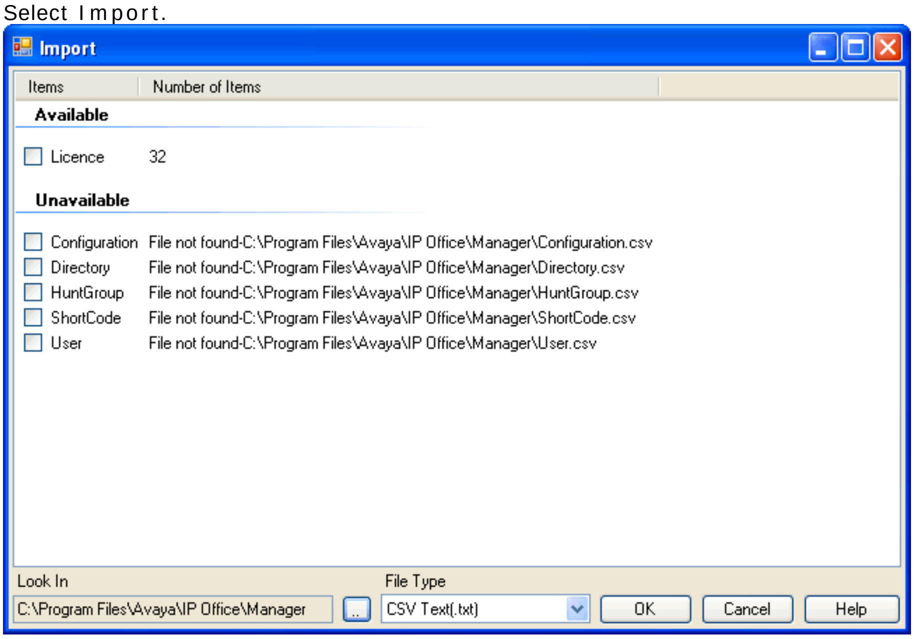
6. Click OK.

Importing Settings

Importing settings will overwrite any existing entries that match an entry being imported.

1. Select File | Import/Export... from the menu bar.

2. Select Import.



3. Select the type of file. The list of items will change to match the type of file selected and whether a matching file or files is found in the current file path.

4. Use Look In to adjust the file path. The default location used is sub-directory of the Manager application directory based on system name of the currently loaded system.

5. Select the types of items that should be imported.

6. Click OK.

2.12.9 Copying and Pasting

Manager supports the normal Windows methods of cutting, copying, pasting and deleting entries and settings. These can be accessed through the Edit menu in the menu bar or using the standard Windows keyboard shortcuts for those actions. They can also be accessed by selecting an entry or text field and then right-clicking.

Copy and paste can be used with the navigation and group panes to create a new entry with the same settings as the original. The copy will be renamed as *Copy of ...* to avoid conflicting with the original.

When using copy and paste between individual settings fields, whether on the same entry or a different entry, care should be taken to ensure that the fields use the same type of data. Similarly copying an entry in the navigation or group pane and then pasting it into the details pane will prompt Manager to paste the copied entries data into the first field of the current entry in the details pane. As a general rule, cut and paste actions should be used with the same pane and within similar entry types.

For users and user rights, a number of controls have been provided to copy settings between a user and a user right or vice versa. See User Rights Overview 389 in the Configuration Settings section.

2.12.10 Saving a Configuration Offline

The system configuration settings shown within Manager can be saved as a .cfg file on the Manager PC. These files can be used as backups or sent to other persons to aid problem diagnostics. Note however that an offline configuration file does not include the Audit Trail records for the system.

Automatically Saving Sent Configurations

By default, Manager creates a file copy of the configuration before it is sent to the system. This copy is stored in Manager's Working Directory using the system name and .cfg. This behavior is controlled by the Backup File on Send (File | Preferences | Security) 110 option.

The number of backups of each systems configuration can be limited to a set number of the most recent copies.

Saving a Configuration Received from a System

1. Select File | Save Configuration as from the menu bar.

Saving a Configuration opened on the PC

1. Click  in the main toolbar or select File | Save Configuration from the menu bar.

2.12.11 Sending a Configuration

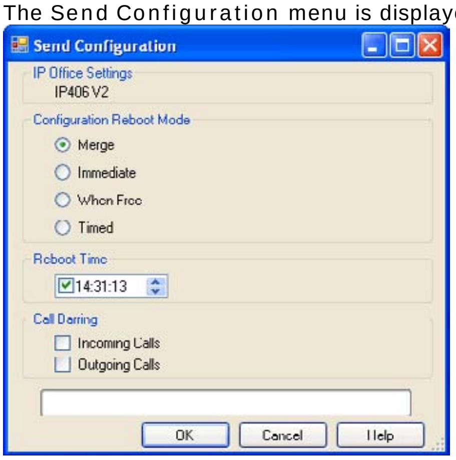
The current configuration settings open within Manager can be sent to the system.

1. The first steps of this process depend on whether you are sending a configuration received from the system or sending one opened offline/created new.

   - **A Configuration Opened from a System**
     Click in the main toolbar or select **File | Save Configuration** from the menu bar.

   - **A Configuration Created Offline or Opened from a PC File**
     Select **File | Offline | Send Config** from the menu bar.

2. The **Send Configuration** menu is displayed.



   - **Password - Pre-3.2 Systems Only**
     This field appears for pre-3.2 system. The system password should be entered. If sending the configuration to an IP Office 3.2+ system, a Service User name and password are requested when OK is clicked.

   - **Configuration Reboot Mode**
     If Manager thinks the changes made to the configuration settings are mergeable, it will select Merge by default, otherwise it will select **Immediate**.

     - **Merge**
       Send the configuration settings without rebooting the system. This mode should only be used with settings that are mergeable. Refer to Mergeable Settings 51.

     - **Immediate**
       Send the configuration and then reboot the system.

     - **When Free**
       Send the configuration and reboot the system when there are no calls in progress. This mode can be combined with the **Call Barring** options.

     - **Timed**
       The same as When Free but waits for a specific time after which it then wait for there to be no calls in progress. The time is specified by the **Reboot Time**. This mode can be combined with the **Call Barring** options.

   - **Reboot Time**
     This setting is used when the reboot mode **Timed** is selected. It sets the time for the system reboot. If the time is after midnight, the system's normal daily backup is canceled.

   - **Call Barring**
     These settings can be used when the reboot mode When Free is selected. They bar the sending or receiving of any new calls.

3. Click **OK**. A Service User name and password may be requested.

   - If the service user name or password used do not valid, *"Access Denied"* is displayed.

   - If the service user name used does not have rights to send a configuration or to request a reboot or merge, "**Insufficient service user rights**" is displayed.

   - If the service user name used does not have operator rights to make the changes that have been made to the configuration, *"Insufficient operator rights. Operator cannot modify xxxx records"* is displayed.

   - The warning will appear if the configuration being sent contain any errors indicated by a icon in the error pane. The configuration can still be sent by selected **Yes**.

   - IP500v2: The message *Failed to save the configuration data. (Internal error)* may indicate that the IP500v2 system has booted using software other than that in its System SD card's primary folder.

2.12.12 Erasing the Configuration

The system configuration settings can be erased. During this process, the system is rebooted and starts with a set of default settings 65.

This process does not erase the security settings of the system. Those can only be reset by a separate process detailed in the Manager Installation Manual.

1. Select File | Advanced | Erase Configuration (Default).
2. Enter a valid user name and password.
3. The system will be rebooted.

2.12.13 Default Settings

The following applies to new control units and those defaulted using the Erase Configuration 65 command. It also applies to IP500 and IP500v2 control units defaulted using the reset button on the rear of the unit (refer to the Manager Installation manual for details of using the reset button).

Mode

IP500v2 control units can operate in a number of modes. The initial mode is determined by the type of System SD card fitted and the level of software.

- **IP Office A-Law**
  A system fitted with this type of card will default to A-Law telephony. The system will default to IP Office Standard Version mode.
- **IP Office U-Law**
  A system fitted with this type of card will default to U-Law telephony. The system will default to IP Office Standard Version mode.
- **IP Office Partner Edition**
  A system fitted with this type of card will default to A-Law telephony and IP Office Essential Edition - PARTNER® Version mode operation.
- **IP Office Norstar Edition**
  A system fitted with this type of card will default to U-Law telephony and IP Office Essential Edition - Norstar Version mode operation.
- **Avaya Branch Gateway**
  Use this option for an SD card intended to be used with a system running in Avaya Branch Gateway mode. There is a separate SD card for Avaya Branch Gateway. The Avaya Branch Gateway SD card can only be used for Avaya Branch Gateway operation and cannot be used to change modes to IP Office. The label on the SD card includes BRANCH GW to indicate it is an Avaya Branch Gateway SD card. You also cannot use or change an IP Office SD card for use with an Avaya Branch Gateway system.
  - **! Warning**
    Do not re-purpose an Avaya Branch Gateway SD card for use with any other IP Office mode. Doing so may damage the SD card and make it unusable for your Avaya Branch Gateway system.

- **A-Law or U-Law**
  PCM (Pulse Code Modulation) is a method for encoding voice as data. In telephony, two methods of PCM encoding are widely used, A-Law and U-Law (also called Mu-Law or µ-Law). Typically U-Law is used in North America and a few other locations while A-Law is used elsewhere. As well as setting the correct PCM encoding for the region, the A-Law or U-Law setting of a system when it is first started affects a wide range of regional defaults relating to line settings and other values.
  - For IP400 systems, each control units was manufactured as either an A-Law variant or a U-Law variant.
  - For IP500 and IP500v2 systems, the encoding default is set by the type of Feature Key installed when the system is first started. The cards are either specifically A-Law or U-Law. PARTNER Version cards are U-Law. Norstar Version cards are A-Law.

Default Short Codes

IP400 control units are manufactured as A-Law and U-Law variants. For IP500 and IP500v2 control units, A-Law or U-Law variant operation is determined by the Feature Key dongle installed in the system. Depending on the variant, a default standard mode system will use different sets of default short codes. See the Default System Short Code List 437.

Default Data Settings

When a new or defaulted control unit is switched on, it requests IP address information from a DHCP Server on the network. This operation will occur whether the LAN cable is plugged in or not.

- If a DHCP server responds within approximately 10 seconds, the control unit defaults to being a DHCP client and uses the IP address information supplied by the DHCP server.
- If no DHCP Server responds, the control unit still defaults to being the DHCP client but assumes the following default LAN addresses:
  - For its LAN1 it allocates the IP address 192.168.42.1 and IP Mask 255.255.255.0.
  - For its LAN2 if supported, it allocates the IP address 192.168.43.1 and IP Mask 255.255.255.0.

- ! Once an IP500v2 control unit has obtained IP address and DHCP mode settings, it will retain those settings even if rebooted without a configuration file present on the System SD card. To fully remove the existing IP address and DHCP mode setting, the system must be defaulted using Manager.

Default Security Settings

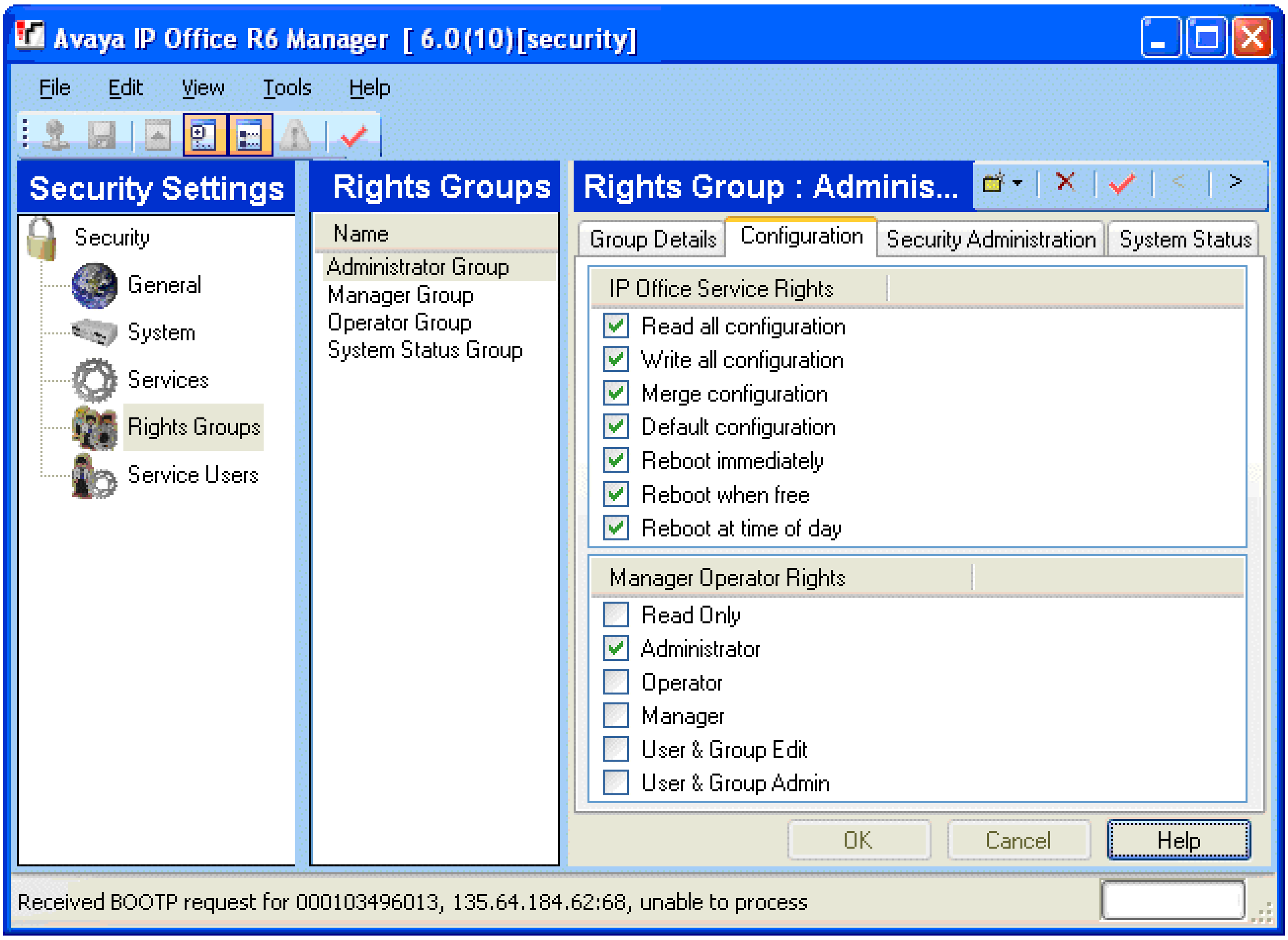
Security settings are held separately from the configuration settings and so are not defaulted by actions that default the configuration. To return the security settings to their default values the separate Erase Security Settings 119 command should be used.

Chapter 3.
Security Mode

3. Security Mode

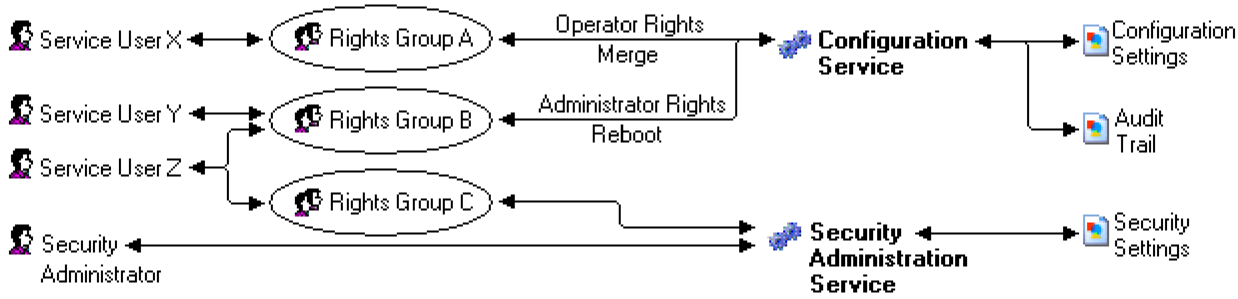
These menus are used to edit the security settings of a system. They are not used for systems running in IP Office Essential Edition - PARTNER® Version or IP Office Essential Edition - Norstar Version mode.

The security settings are stored on the system and are separate from the system's configuration settings. To change a system's security settings, Manager must first be switched to security mode by selecting File | Advanced | Security Settings from the menu bar.



3.1 Security Settings

The following applies for IP Office 3.2 and higher. Access to system settings is controlled by Service Users and Rights Groups. stored in the control unit's security settings. These are stored separately from the system's configuration settings. All actions involving communications between Manager and the system require a Service User name and password. That Service User must be a member of a Rights Group with permissions to perform the required action.



In the example illustrated above:

- Service User X can read and write the configuration. However they can only edit Operator settings and can only make changes that can be merged.
- Service User Y can read and write the configuration, edit all settings and make changes that require reboots.
- Service User Z can read and write the configuration, edit all settings and make changes that require reboots. They can also access the security settings.
- The Security Administrator can only access the security settings.

Security Administrators
By default the security administrator is the only user who can access the system's security settings using Manager's security mode.

Service Users
Each Service User has a name, a password and is a member of one or more Rights Groups.

Rights Groups
The Rights Groups to which a Service User belongs determine what actions they can perform. Actions available to Rights Groups include configuration actions, security actions and system status actions:

Configuration	Security	System Status
• Read the configuration. • Write the configuration. • Merge the configuration. • Default the configuration. • Reboot immediately. • Reboot when free. • Reboot at time of day.	• Read all security settings. • Write all security settings. • Reset all security settings. (Manager 4.1+) • Write own password. (Manager 4.1+)	• System Status Access. • Read All Configuration.

Where a Service User has been configured as a member of more than one Rights Group, they combine the functions available to the separate Rights Groups.

Operator Rights

Each Rights Group has a Manager Operator Rights setting. This setting controls what types of configuration entries Manager will allow members of the Rights Group to view and what actions they can perform with those types of entries.

Operator	View/Edit/New/Delete	Configuration Entry Types
Administrator	All	View, edit create and delete all configuration entries.
Manager	View	View all except WAN Port.
	Edit	Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, Directory, Time Profile, Firewall Profile, IP Route, Least Cost Route, Account Code, ARS, E911 System.
	New	
	Delete	As edit except Short Code.
Operator	View	View all except WAN Port.
	Edit	Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, Time Profile, Firewall Profile, IP Route, Least Cost Route, Account Code, License, ARS, E911 System.
	New	None.
	Delete	Delete Incoming Call Route and Directory.
User & Group Edit	View	User and Hunt Group entries only.
	Edit	
	New	None
	Delete	
User & Group Admin	All	User and Hunt Group entries only.
Dir & Account Admin	All	Directory and Account Code entries only.
Time & Attendant Admin	All	Time Profile and Auto Attendant entries only.
ICR & User Rights Admin	All	Incoming Call Route and User Rights entries only.
Read Only	View	View all configuration entries.
	Edit	None.
	New	
	Delete	

3.2 Default Security Users

This section lists the default Rights Groups and Service Users.

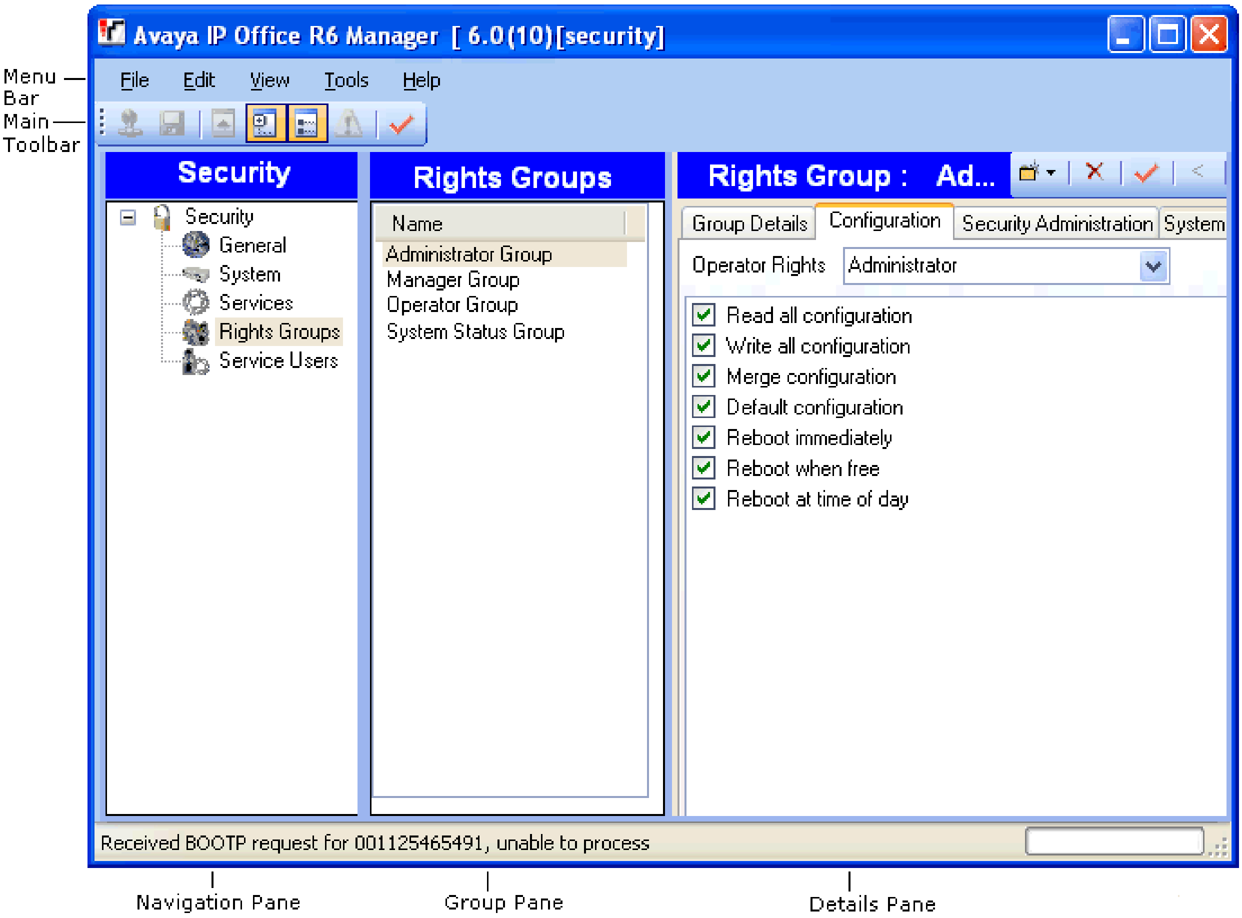
- ⚠ WARNING: Change Passwords
  These settings must be changed to make the system secure. At minimum you must change the default passwords of the Security Administrator and the default Service Users. Failure to do so will render the system unsecure.

Unique Security Control Unit					
Enabled	✓				
Name	security				
Password	securitypwd				
Default Service Users					
Name	Administrator	Manager	Operator	EnhTcpaService	IPDECTService
Password	Administrator	Manager	Operator	EnhTcpaPwd1	–
Rights Group Membership					
- Administrator Group	✓	✗	✗	✗	
- Manager Group	✗	✓	✗	✗	
- Operator Group	✗	✗	✓	✗	
- System Status Group	✓	✓	✓	✗	
- TCPA Group	✗	✗	✗	✓	
- IPDECT Group	✗	✗	✗	✗	

Default Rights Groups		Administrator Group	Manager Group	Operator Group	System Status Group	TCPA Group	IPDECT Group
Operator Rights	Operator View	Administrator	Manager	Operator	–	–	–
Service Rights	Read Configuration	✓	✓	✓	✗	✗	✗
	Write Configuration	✓	✓	✓	✗	✗	✗
	Default Configuration	✓	✓	✓	✗	✗	✗
	Merge	✓	✓	✓	✗	✗	✗
	Reboot Immediately	✓	✓	✓	✗	✗	✗
	Reboot When Free	✓	✓	✓	✗	✗	✗
	Reboot at Time of Day	✓	✓	✓	✗	✗	✗
Security Administration	Read All Security Settings	✗	✗	✗	✗	✗	✗
	Write All Security Settings	✗	✗	✗	✗	✗	✗
	Reset All Security Settings (IP Office 4.1+)	✗	✗	✗	✗	✗	✗
	Write own service user password (IP Office 4.1+)	✗	✗	✗	✗	✗	✗
System Status (IP Office 4.0+)	System Status Access	✗	✗	✗	✓	✗	✗
	Read all configuration	✗	✗	✗	✓	✗	✗
	System Control (IP Office 6.0+)	✗	✗	✗	✓	✗	✗
Enhanced TSPI (IP Office 5.0+)	Enhanced TSPI Access	✗	✗	✗	✗	✓	✗
HTTP	DECT R4 Provisioning	✗	✗	✗	✗	✗	✓

3.3 The Security Mode Interface

Manager can be switched to security mode. This mode it is used to load and edit the security settings of a system. How the controls operate is similar to Manager in configuration mode.



Switching Manager to Security Mode

1. Select File | Advanced | Security Settings.

Switching Manager Back to Configuration Mode

1. Select File | Configuration.

Manager Security Mode Screen Elements

- Menu Bar
  Provides commands for loading and saving security settings. See the Menu Bar Commands 102 section.
- Main Toolbar
  The toolbar icons perform the following actions:
  - Get the Security Settings.
  - Save the Security Settings.
  - Not Used in security mode.
  - Show / Hide the Navigation Pane.
  - Show / Hide the Group Pane.
  - Not used in security mode.
  - Not used in security mode.
- Security Settings Pane
  This pane is used to select the type of security entries that should be displayed in the group pane or details pane.
  - General
    Defines general security controls for the system. When selected, the settings are displayed in the details pane.
  - System
    Defines security settings for the system such as application access. When selected, the settings are displayed in the details pane.
  - Services
    Secure services supported by the system. Currently these are access to security settings and access to configuration settings.
  - Rights Groups
    Create groups with different access rights. When selected, the existing Rights Groups are displayed in the group pane.
  - Service Users
    Sets the name and password for an administrator. Also allows selection of the Rights Groups to which the user belongs. When selected, the existing Service Users are displayed in the group pane.
- Group Pane
  This pane is used to display the existing Right Groups or Service Users when those options are selected in the security settings pane.
- Details Pane
  This pane shows the settings selected in the security settings pane or the group pane.
- Status Bar
  This bar display messages about communications between Manager and systems. It also displays the security level of the communications by the use of a padlock icon.

3.4 Security Administration

IP Office 4.1 added support for the use of certificates if required. This section also covers a basic introduction to security principles and the security mechanisms.

- NOTE: If administration security is of no concern, the default settings allow modification of all Manager features without restriction. It is recommended as a minimum that default passwords are changed.

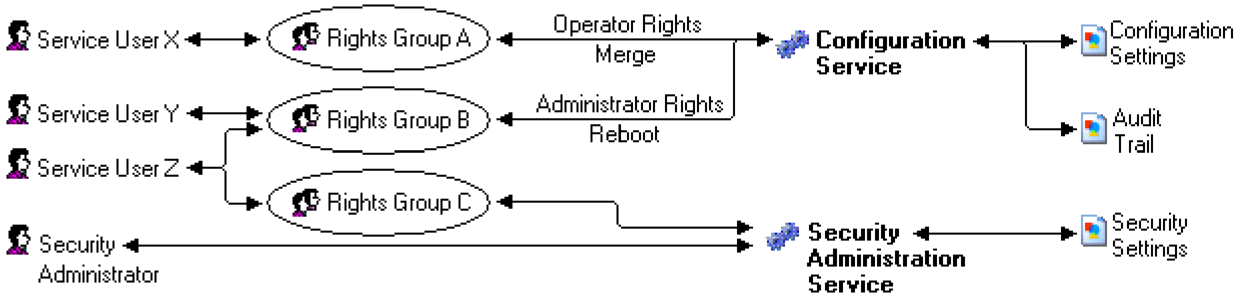
1. Introduction

Administration security is achieved using a number of optional cryptographic elements:

- Access control to prevent unauthorized use.Supported in version 3.2+.
- Encryption to guarantee data remains private. Supported in version 4.1+.
- Message Authentication ensures data has not been tampered with. Supported in version 4.1+.
- Identity assures the source of the data. Supported in version 4.1+.

2. Access Control

Access to configuration, security settings and SSA is controlled by the use of Service Users, passwords and Rights Groups. All actions involving communications between the Manager user and the system require a Service User name and password. That Service User must be a member of a Rights Group configured to perform the required action.



In the example illustrated above:

- Service User X can read and write the configuration. However they can only edit Operator settings and can only make changes that can be merged.
- Service User Y can read and write the configuration, edit all settings and make changes that require reboots.
- Service User Z can read and write the configuration, edit all settings and make changes that require reboots. They can also access the security settings.
- The Security Administrator can only access the security settings.

Security Administrator

By default the security administrator is the only user who can access the system's security settings using Manager's security mode.

Service Users

Each Service User has a name, a password and is a member of one or more Rights Groups.

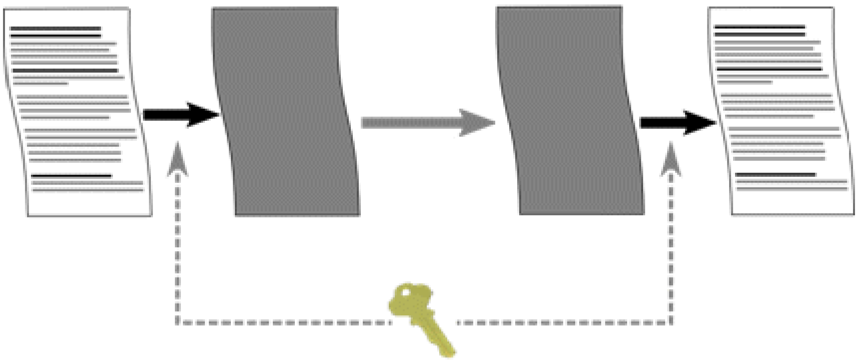
Rights Groups

The Rights Groups to which a Service User belongs determine what actions they can perform. Actions available to Rights Groups include configuration actions, security actions and system status actions.

Where a Service User has been configured as a member of more than one Rights Group, they combine the functions available to the separate Rights Groups.

3. Encryption

Encryption ensures that all data sent by either the system or Manager cannot be 'read' by anyone else, even another copy of Manager. Encryption is the application of a complex mathematical process at the originating end, and a reverse process at the receiving end. The process at each end uses the same 'key' to encrypt and decrypt the data:



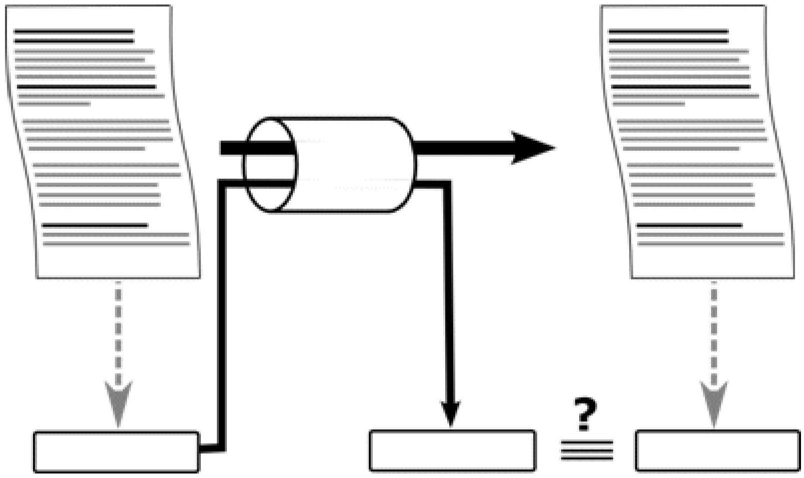
Any data sent may be optionally encrypted using a number of well known and cryptographically secure algorithms:

Algorithm	Effective key size (bits)	Use
DES-40	40	Not recommended.
DES-56	56	'Minimal' security.
3DES	112	'Strong' security.
RC4-128	128	'Strong' security.
AES-128	128	'Very strong' security.
AES-256	256	'Very strong' security.

In general the larger the key size, the more secure the encryption. However smaller key sizes usually incur less processing. The system supports encryption using the Transport Layer Security (TLS) v1.0 protocol. In addition, many cryptographic components of the TLS module have been FIPS 140-2 certified, indicating the accuracy of implementation.

4. Message Authentication

Message authentication ensures that all data sent by either the system or Manager cannot be tempered with (or substituted) by anyone else without detection. This involves the originator of the data producing a signature (termed a hash) of the data sent, and sending that as well. The receiver gets the data and the signature and check both match.



Any data sent may be optionally authenticated using a number of well known and cryptographically secure algorithms:

Algorithm	Effective hash size (bits)	Use
MD5	128	'Minimal' security.
SHA-1	160	'Strong' security.

In general the larger the hash size, the more secure the signature. However smaller hash sizes usually incur less processing.

Manager supports message authentication using the Transport Layer Security (TLS) v1.0 protocol. In addition, many cryptographic components of the TLS module have been FIPS 140-2 certified, indicating the accuracy of implementation.

5. Identity

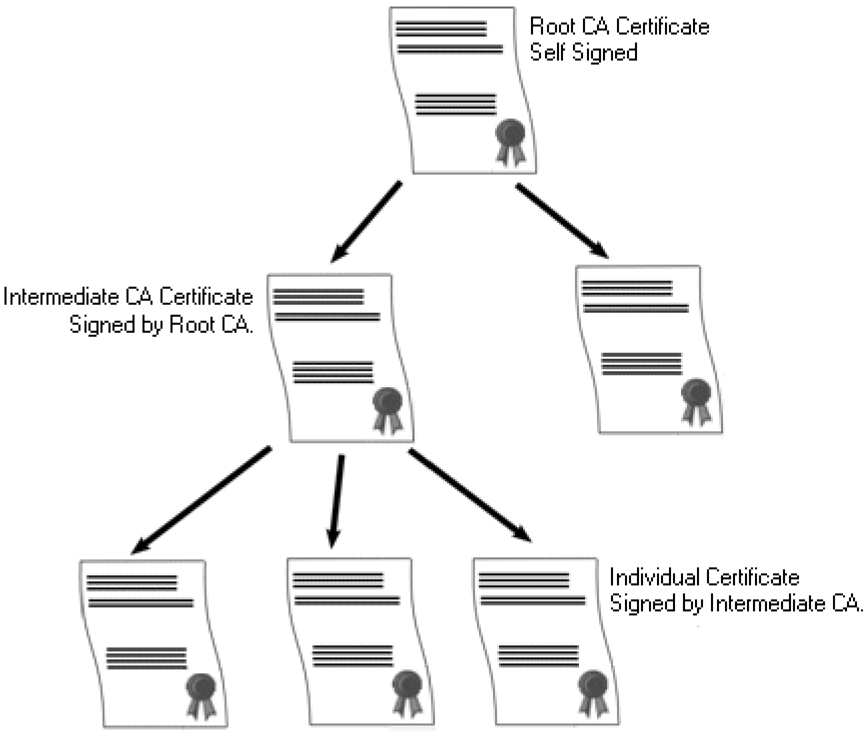
The identity of the equipment or person at each end of the link is achieved by the used of digital certificates – more specifically X.509 v3 certificates. Digital certificates are the preferred mechanism for the majority of internet-based applications including e-commerce and email, and can be thought of as a credential, just like a passport or drivers' license.

A digital certificate contains at least three things:

- A public key.
- Certificate information (Identity information about the user, such as name, user ID, and so on.)
- One or more digital signatures

The purpose of the digital signature on a certificate is to state that the certificate information has been verified to by some other person or entity. The digital signature does not verify authenticity of the certificate as a whole; it vouches only that the signed identity information goes along with, or is bound to, the public key: A certificate essentially is a public key with one or two forms of ID attached, plus a stamp of approval from some other 'trusted individual'.

Trusted individuals (also termed Certificate Authorities) themselves have publicly available certificates, which can contain signatures from their trusted authorities. These can be verified all the way up to a 'self-signed' root certificate from a root certificate authority.



Examples of root certificate authorities' certificates can be found in every web browsers' certificate store.

6. Windows Certificate Store Usage

The certificate store that is used by the Manager to save X509 certificates to and retrieve certificates from is the default one provided by the Windows operating system. This may be accessed for maintenance purposes by a user with sufficient permission via the use of a 'snap-in'.

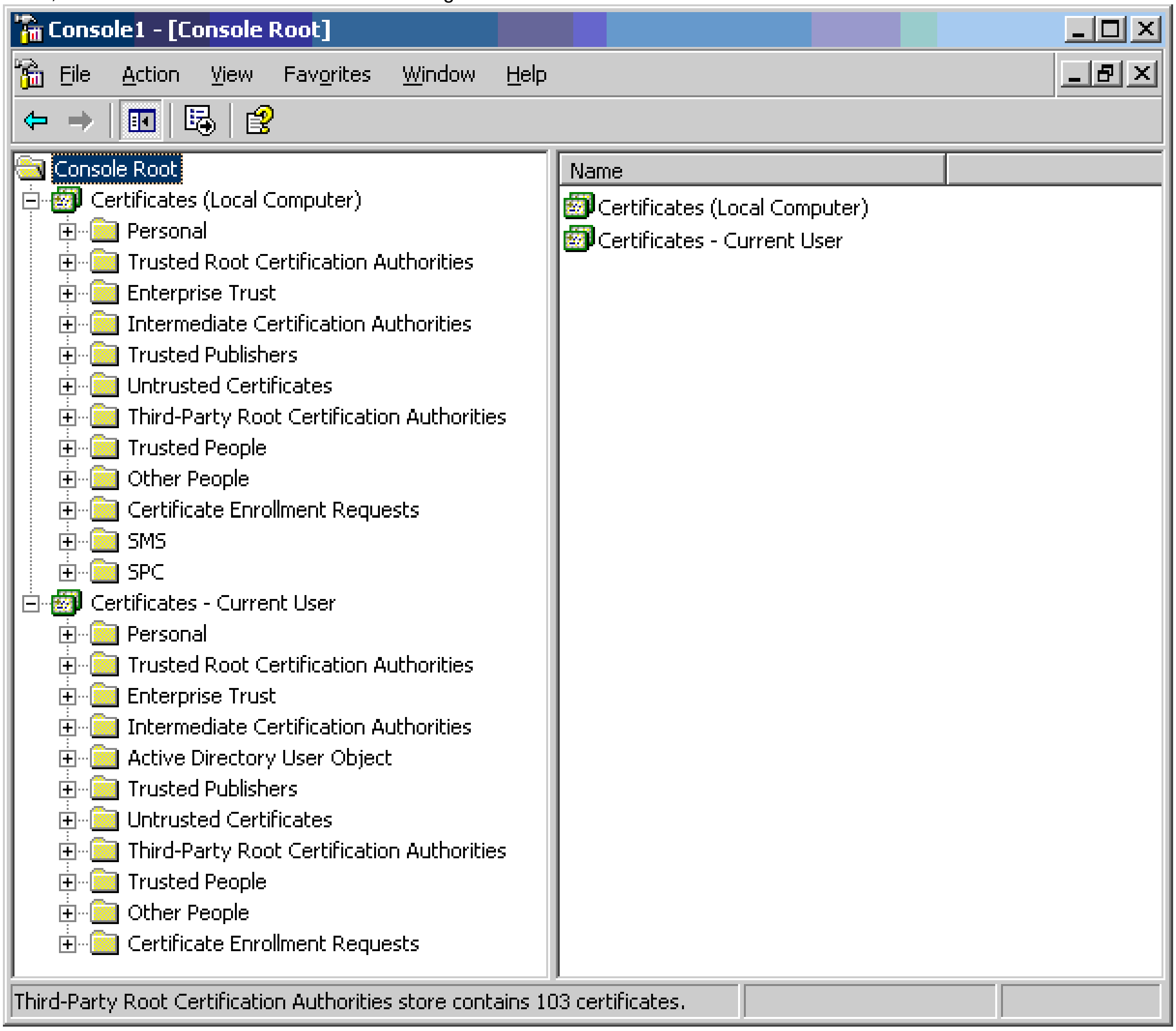
- WARNING
  Avaya accept no responsibility for changes made by users to the Windows operating system. Users are responsible for ensure that they have read all relevant documentation and are sufficiently trained for the task being performed.

If not installed already, the Microsoft Management Console (MMC) Certificates snap-in can be installed by following the relevant instructions. Both 'user account' and 'computer' options should be installed.

- For Windows XP Professional:
  http://www.microsoft.com/resources/documentation/windows/xp/all/proddocs/en-us/sag_cm_addsnap.mspx
- For Windows Server 2003:
  http://technet2.microsoft.com/windowsserver/en/library/4fa4568e-16de-4a64-b65e-12ee14b31dc21033.mspx?mfr=true
- For Windows Vista:
  http://technet2.microsoft.com/WindowsVista/f/?en/library/a8b21b9b-d102-4045-9f36-e4b3430d2f381033.mspx

7. Windows Certificate Store Organization

By default, certificates are stored in the following structure:



Each of the sub folders has differing usage. The Certificates - Current User area changes with the currently logged-in windows user. The Certificate(Local Computer) area does not change with the currently logged-in windows user.

Manager only accesses some of the certificate sub folder:

Certificates (Local Computer) Folder	Manager Use
Personal | Certificates	• Folder searched by Manager 1st for matching certificate to send to the system when requested. Certificate matched by the subject name contained in File | Preferences | Security | Certificate offered to the system. • Folder accessed whenever 'Local Machine certificate store' used for Security Settings. • Folder searched by Manager for matching certificate when certificate received from the system, and File | Preferences | Security | Manager Certificate Checks = Medium or High.
Trusted Root Certification Authorities | Certificates	• Folder searched by Manager for matching parent certificates when non-self signed certificate received from the system, and File | Preferences | Security | Manager Certificate Checks = Medium or High.

Certificates – Current User Folder	Manager Use
Personal | Certificates	• Folder searched by Manager 2nd for matching certificate (subject name) to send to the system when requested. Certificate matched by the subject name contained in File | Preferences | Security | Certificate offered to the system. • Folder accessed whenever 'Current User certificate store' used for Security Settings. • Folder searched by Manager for matching certificate when certificate received from Manager, and File | Preferences | Security | Manager Certificate Checks = Medium or High.
Trusted Root Certification Authorities | Certificates	• Folder searched by Manager for matching parent certificates when non-self signed certificate received from the system, and File | Preferences | Security | Manager Certificate Checks = Medium or High.
Other People | Certificates	• Folder searched by Manager for matching parent certificates when non-self signed certificate received from the system, and File | Preferences | Security | Manager Certificate Checks = Medium or High.

8. Windows Certificate Store Import

In order to use certificates – either for security settings or Manager operation – they must be present in the windows certificate store. Certificates may be placed in the store by the Certificate Import Wizard or the Certificate MMC snap-in

The Certificate Import Wizard can be used whenever a certificate is viewed. In order for Manager to subsequently access this certificate the Place all certificate in the following store option must be selected:

- If the certificate is to subsequently identify the system, the Other People folder should be used.
- If the certificate is to subsequently identify the Manager, the Personal folder should be used, and the associated private key saved as well.

If the saved certificate is to be used by other windows users, the MMC certificate snap-in must be used to move it to the Certificates (Local Computer) folder.

9. Certificate Store Export

Any certificate required outside of the Manager PC required to be first saved in the Certificate store, then exported using the MMC snap-in.

If the certificate is to be used for identity checking (i.e. to check the far entity of a link) the certificate alone is sufficient, and should be saved in PEM or DER format.

If the certificate is to be used for identification (i.e. to identify the near end of a link) the certificate and private key is required, and should be saved in PKCS#12 format, along with a password to access the resultant .pfx file.

10. Implementing Administration Security

This section suggests system security settings that could implement possible security requirements. This section does not cover the general aspects of security policy analysis or definition, or how the system administration security interacts with other security mechanism.

10.1. Negligible Security

If all Manager and system security settings are left at default, no security mechanisms are active, other than the use of default service user names and passwords. In addition, all legacy interfaces are active, and all configuration and security data is sent unencrypted.

It is recommended that at the very least, the default service user passwords are changed.

10.2. Minimum Security

A minimum security scenario could be where configuration data is open, but the security settings are constrained: Any individual with the correct service user name and password can access the configuration from any PC installation of Manager, no logging of access: Passwords can be simple, and will never age.

- Change all default passwords of all service users and Security Administrator
- Set the system Security Administration service security level to Secure, Low.
- Set the system Service User Password Reject Action to None.
- Set the system Client Certificate Checks level to None (default).
- Set the system Minimum Password Complexity to Low (default).
- Set the system Previous Password Limit to zero (default).
- Set the system Password Change Period to zero (default).
- Set the system Account Idle Time to zero (default).
- Set certificate check level to low in Manager Security Preferences (default).

In addition, any PC installation of Manager can manage any Manager.

10.3. Medium Security

A medium security scenario could be where both configuration and security settings are constrained and a level of logging is required: Any individual with the correct service user name and password can access the configuration from any PC installation of Manager: Passwords cannot be simple, and will age.

- Change all default passwords of all service users and Security Administrator
- Set the system Security Administration service security level to Secure, Medium.
- Set the system Configuration service security level to Secure, Medium.
- Set the system Service User Password Reject Action to Log to Audit Trail (default).
- Set the system Client Certificate Checks level to None (default).
- Set the system Minimum Password Complexity to Medium.
- Set the system Previous Password Limit to non zero.
- Set the system Password Change Period to non zero.
- Set the system Account Idle Time to zero (default).
- Disable all the system Unsecured Interfaces.
- Set certificate check level to low in Manager Security Preferences (default).

10.4. Maximum Security

A maximum security scenario could be where both configuration and security settings are constrained and a full level of logging is required: Certified individuals with the correct service user name and password can access the configuration from specific PC installations of Manager: Passwords cannot be simple, and will age: Manager can managed specific systems.

- Change all default passwords of all service users and Security Administrator
- Set the system Security Administration service security level to Secure, High.
- Set the system Configuration service security level to Secure, High.
- Set the system Service User Password Reject Action to Log and Disable Account.
- Set the system Client Certificate Checks level to High.
- Set the system Minimum Password Complexity to High.
- Set the system Minimum Password Length to >8.
- Set the system Previous Password Limit to non zero (>5).
- Set the system Password Change Period to non zero.
- Set the system Account Idle Time to non zero.
- Set the system Session ID Cache to zero.
- Install valid, 1024 bits+, non self signed certificates (+private key) in all Manager server certificates, derived from a trusted certificate authority.
- Install the corresponding trusted CA certificate in each of the Manager's windows certificate stores.
- Install valid, 1024 bits+, non self signed certificate (+ private key) in all Manager Certificate Stores.
- Install the corresponding certificates in all the system Certificate Stores of all permissible Manager entities, and the trusted CA certificate.
- Disable all the system Unsecured Interfaces.
- Set Manager Certificate Checks level to high in Manager Security Preferences.
- Set Certificate offered to the system in Manager Security Preferences.

The above essentially locks the systems and corresponding Managers together. Only recognized (by strong certificate) entities may communicate successfully on the service interfaces. All services use strong encryption and message authentication.

The use of intermediate CA certificates can be used to overcome the limit of 6 maximum certificates in each system Certificate Store.

3.5 Editing Security Settings

Security settings can only be loaded directly from a system. These settings cannot be saved as a file on the local PC, nor do they appear as a temporary file at any time.

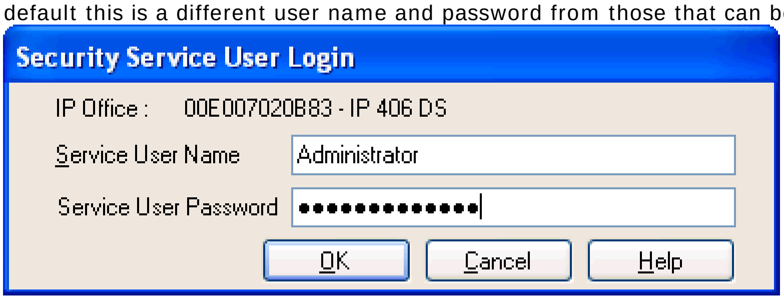
For IP Office 4.1 and higher, you can optionally secure the link between the system and Manager for configuration and security settings exchanges. By default Manager and the system will always attempt to use the original, unsecured link. The control of which mechanism is used by Manager is determined Manager preferences.

When secure communications mode is selected a padlock icon is present on the Manager status bar.

Loading Security Settings

The address ranges in which Manager searches for systems are set through the Manager preferences (File | Preferences | Discovery 106). The security mechanism used for security settings transfer between Manager and a system are set through the Secure Communications attribute of Manager preferences (File | Preferences | Security).

1. If not already done, switch Manager to security mode by selecting File | Advanced | Security Settings.
    - Note: If the system's configuration settings have already been loaded using a Service User name and Password that also has security access, then the security settings are automatically loaded when Manager is switched to security mode.
2. If already in security mode, click in the main toolbar or select File | Open Security Settings from the menu bar.
3. The Select IP Office window appears, listing those systems that responded. The list can be sorted by clicking on the column names.
4. If the system required was not found, the address used for the search can be changed. Enter or select the required address in the Unit/Broadcast Address field and then click Refresh to perform a new search.
5. When the system required is located, check the box next to the system and click OK.
6. The user name and password request for the system is then displayed. Enter the required details and click OK. By default this is a different user name and password from those that can be used for configuration access.



7. If the security settings are received successfully, they appear within Manager.
    - If the service user name/password is incorrect, or the service user has insufficient rights to read the security settings, *"Access Denied"* is displayed.
    - If the network link fails, or the secure communication mode is incorrect (for example Manager is set to unsecured, but the system is set to secure only), *"Failed to communicate with IP Office"* is displayed.

Editing Security Settings

Editing security settings differ from editing configuration settings in a number of ways:

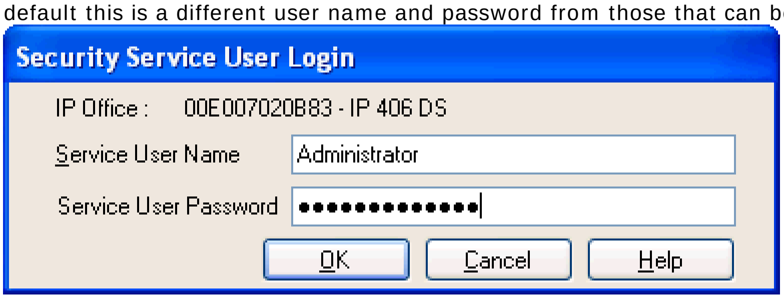
1. Editing of security settings may only be done online to a system. No offline saving or editing is allowed for security purposes.
2. No errors in the security settings are allowed to persist. This prevents the system becoming inaccessible through operator error.
3. Sets of changes to security objects may be made without the need for the OK button to be selected every time. This allows a coordinated set of changes to be accepted or canceled by the operator.

Saving Security Settings

1. Click in the Main Toolbar or select File | Save Security Settings from the menu bar. These options are only available when some change has been made.
2. The user name and password request for the system is then displayed. Enter the required details and click OK. By default this is a different user name and password from those that can be used for configuration access.

Resetting a System's Security Settings *(IP Office 4.1+)*

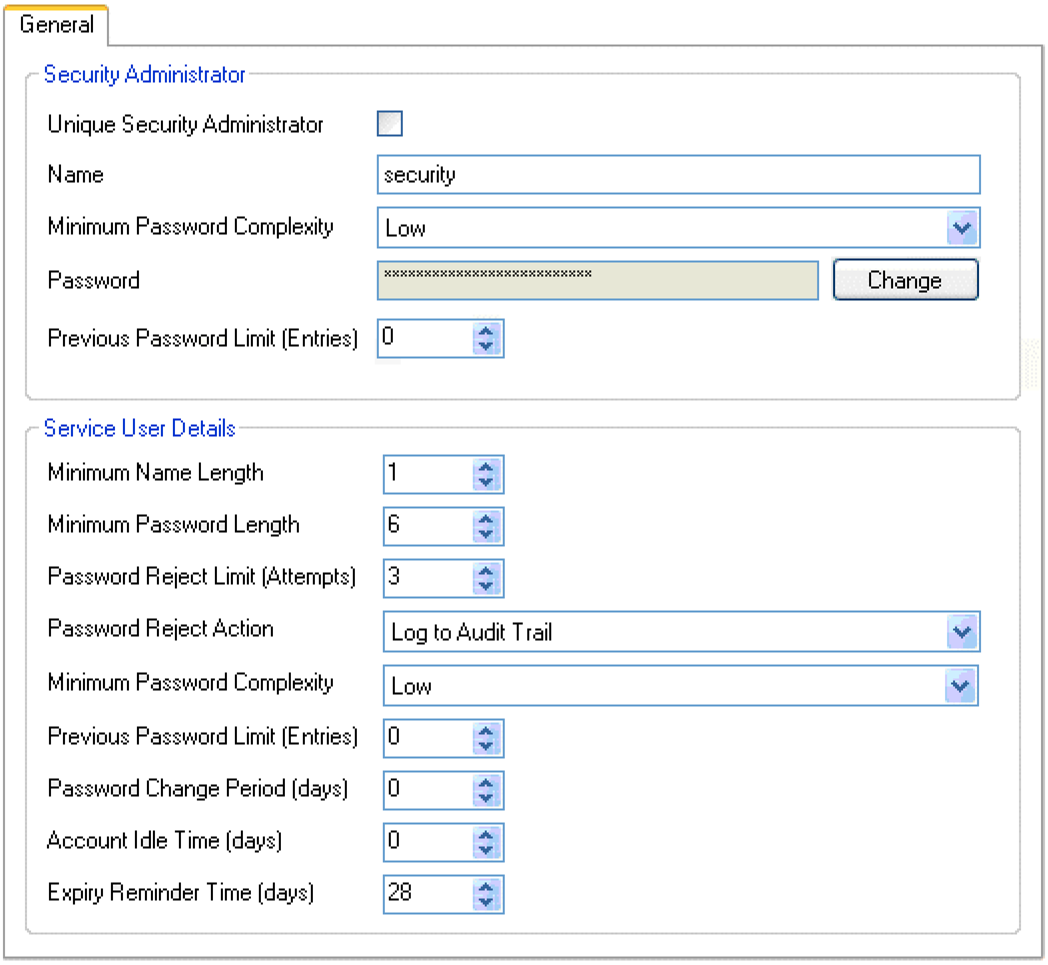
1. Select File | Reset Security Settings (if in security mode), or File | Advanced | Erase Security Settings (if in configuration mode).
2. The Select IP Office window appears, listing those systems that responded. The list can be sorted by clicking on the column names.
3. When the system required is located, check the box next to the system and click OK.
4. The user name and password request for the system is then displayed. Enter the required details and click OK. By default this is a different user name and password from those that can be used for configuration access.



5. Manager will indicate if the security settings are reset successfully.

3.5.1 General Settings

These settings are displayed when  General is selected in the navigation pane.



- Security Administrator
  The security administrator is a special service user who does not belong to any Rights Groups. The security administrator is able to access the system's security settings but cannot access its configuration settings. By default they are the only service user able to access to the security settings.

  - Unique Security Administrator: *Default = Off*
    When selected, only the Security Administrator is able to access the system's security settings. When this is selected, the security options for Rights Groups are disabled. When not selected, the ability to access security settings can also be assigned to Rights Groups.

  - Name: *Default = 'security'. Range = 6 to 31 characters.*
    The name for the Security Administrator.

  - Password: *Default = 'securitypwd'. Range = 6 to 31 characters.*
    The password for the Security Administrator. In order to change the Security Administrator password, the current password must be known.

  - Minimum Password Complexity: *Default = Low. Software level = 4.1+.*
    The password complexity requirements for the Security Administrator. This setting is active for attempted password changes on both Security Manager and the system.

    - Low
      Any password characters may be used without constraint.

    - Medium
      The password characters used must include characters from at least 2 of the 'code point sets' listed below. For example a mix of lower case and upper case. In addition, there should not be any adjacent repeated characters of any type.

      1. Lower case alphabetic characters.
      2. Upper case alphabetical character.
      3. Numeric characters.
      4. Non-alphanumeric characters, for example # or *.

* High
  The password characters used must include characters from at least 3 of the 'code point sets' listed above. For example a mix of lower case, upper case and numbers. In addition, there should not be any adjacent repeated characters of any type.

* Previous Password Limit (Entries): *Default = 0. Range = 0 (Off) to 10 entries. Software level = 4.1+.*
  The number of previous password to check for duplicates against when changing the password. When set to *0*, no checking of previous passwords takes place. This setting is active for attempted password changes on both Security Manager and the system.

* Service User Details
  These settings control Service User names and password/account policies.

  * Minimum Name Length: *Default = 6, Range 1 to 31 characters.*
    This field sets the minimum name length for Service User names.

  * Minimum Password Length: *Default = 6, Range 1 to 31 characters.*
    This field sets the minimum password length for Service User passwords.

  * Password Reject Limit: *Default = 3, Range 0 to 255 failures.*
    Sets how many times an invalid name or password is allowed within a 10 minute period before the Password Reject Action is performed. Selecting 0 indicates never perform the Password Reject Action.

  * Password Reject Action: *Default = Log to Audit Trail*
    The action performed when a user reaches the Password Reject Limit. Current options are:

    * No Action

    * Log to Audit Trail
      Log to Audit Trail creates an entry indicating the service user account name and time of last failure.

    * Log and Disable Account: *Software level = 4.1+.*
      Log and Disable Account creates an audit trail entry and additionally permanently disables the service user account. This account may only be enabled using the Security Manager Service User settings.

    * Log and Temporary Disable: *Software level = 4.1+.*
      Log and Temporary Disable creates an audit trail entry and additionally temporarily disables the service user account for 10 minutes. This account may additionally be enabled using the Security Manager Service User settings.

  * Minimum Password Complexity: *Default = Low. Software level = 4.1+.*
    The password complexity requirements for all Service Users. This setting is active for attempted password changes on both Security Manager and the system.

    * Low
      Any password characters may be used without constraint.

    * Medium
      The password characters used must cover two 'code point sets'. For example lower case and upper case. In addition, Medium and High do not allow more than 2 repeated characters of any type.

    * High
      The password characters used must cover three 'code point sets'. For example lower case plus upper case and numbers.

  * Password Change Period: *Default = 0 (Off), Range 0 to 999 days. Software level = 4.1+.*
    Sets how many days a newly changed password is valid. Selecting 0 indicates any password is valid forever. This setting is active for password changes through this form or prompted by Manager. Note that the user must be a member of a Rights Group that has the Security Administration 97 option Write own service user password enabled. If this timer expires, the service user account is locked. The account may only be re-enabled using the Service User Settings 99. To prompt the user a number of days before the account is locked set a Expiry Reminder Time (see below).

    * Whenever this setting is changed and the OK button is clicked, the Security Manager recalculates all existing service user password timers.

  * Account Idle Time: *Default = 0 (Off), Range 0 to 999 days. Software level = 4.1+.*
    Sets how many days a service user account may be inactive before it becomes disabled. Selecting 0 indicates an account may be idle forever. If this timer expires, the service user account is permanently disabled. The account may only be re-enabled using the Service User Settings 99. The idle timer is reset whenever a service user successfully logs in.

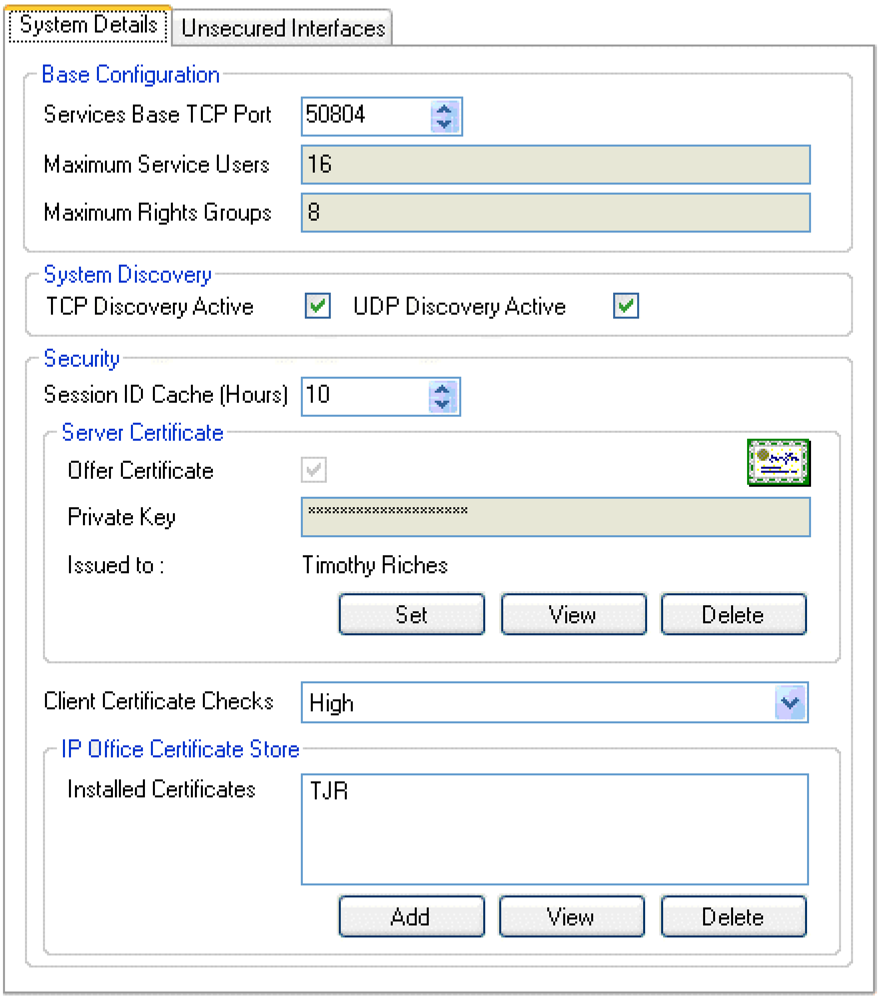
    * Whenever this setting is changed and the OK button is clicked, the Security Manager recalculates all existing service user idle timers.

  * Expiry Reminder Time: *Default = 28, Range 0 (Off) to 999 days. Software level = 4.1+.*
    Sets the period before password or account expiry during which a reminder indication if the service user logs in. Selecting 0 prevents any reminders. Reminders are sent, for password expiry due to the Password Change Period (above) or due to the Account Expiry date (see Service User 99 setting) – whichever is the sooner. Currently Manager displays reminders but System Status does not.

3.5.2 System

3.5.2.1 System Details

This tab is accessible when System is selected in the navigation pane.



- Base Configuration
  - Base TCP Port: *Default = 50804, Range = 49152 to 65526.*
    This is the base TCP port for services provided by the system. It sets the ports on which the system listens for requests to access those services, using its LAN1 IP address. Each service uses a port offset from the base port value. If this value is changed from its default, the Manager application must be set to the same Base TCP Port through its Services Base TCP Port setting (File | Preferences 106).

Service	Method	Port Used	Default
Configuration	Unsecure *(IP Office 3.2+)*	Base TCP Port	50804
	Secure *(IP Office 4.1+)*	Base TCP Port plus 1.	50805
System Status Interface	Unsecure *(IP Office 4.0+)*	Base TCP Port plus 4.	50808
Security Administration	Unsecure *(IP Office 3.2+)*	Base TCP Port plus 8.	50812
	Secure *(IP Office 4.1+)*	Base TCP Port plus 9.	50813
Enhanced TSPI	Unsecure *(IP Office 5.0+)*	Base TCP Port plus 10.	50814

  - When changing the base port, exercise caution that the selected port and those offset from it do not conflict with any ports already in use by other applications.

- Maximum Service Users: *Default = 16.*
  This is a fixed value for indication purposes only. This value is the maximum number of Service Users that can be stored in a system's security settings.

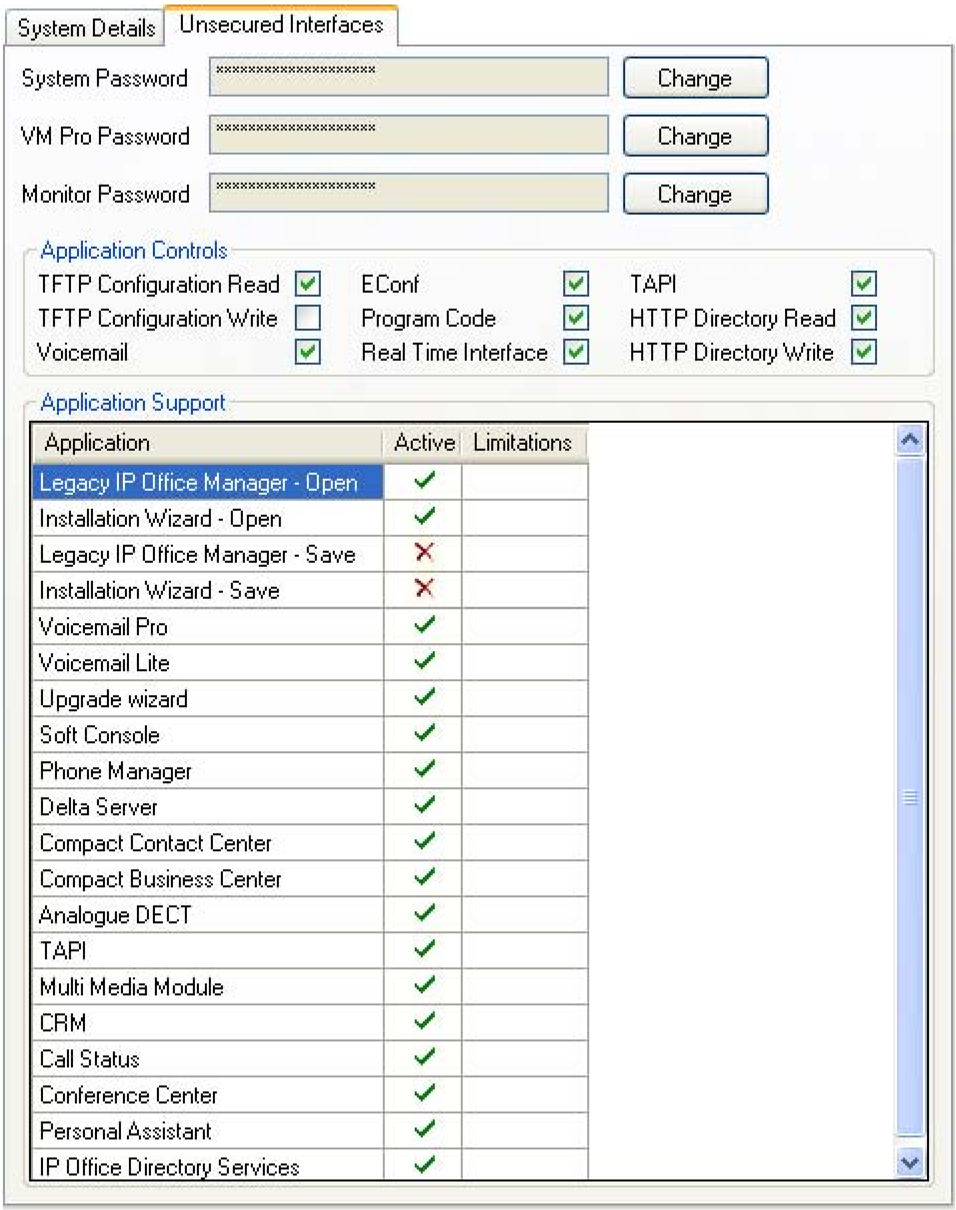
- Maximum Rights Groups: *Default = 8.*
  This is a fixed value for indication purposes only. This value is the maximum number of Rights Groups that can be stored in a system's security settings.

* System Discovery
System discovery is the processes used by applications to locate and list available systems. The Manager can be disabled from responding to this process if required. If this is done, access to the Manager requires its specific IP address to be used.
  * TCP Discovery Active: *Default = On. Software level = IP Office 5.2+.*
  Selecting TCP Discovery Active allows the system to respond to those requests.
  * UDP Discovery Active: *Default = On.*
  Selecting UDP Discovery Active allows the system to respond to those requests.
* Security: *Software level = 4.1+.*
These settings cover the per-system security aspects, primarily TLS settings.
  * Session ID Cache: *Default = 10 hours, Range 0 to 100 hours.*
  This sets how long a TLS session ID is retained by the system. If retained, the session ID may be used to quickly restart TLS communications between the system and a re-connecting application. When set to *0*, no caching takes place and each TLS connection must be renegotiated.
  * Allow HTTPS: *Default = Off. Software level = 6.0-6.1.*
  Allow HTTPS connection to the system for applications such as the IP Office Softphone.
  * Offer Server Certificate: *Default = On.*
  This is a fixed value for indication purposes only. This sets whether the system will offer a certificate in the TLS exchange when the Manager is acting as a TLS server, which occurs when accessing a secured service.
  * Server Private Key: *Default = None.*
  This is a fixed value for indication purposes only. This indicate whether the system has a private key associated with the Server Certificate.
  * Server Certificate: *Default = None.*
  The Server Certificate is an X.509v3 certificate that identifies the system to a connecting client device (usually a PC running a application). This certificate is offered in the TLS exchange when the system is acting as a TLS server, which occurs when accessing a secured service. By default the system's own self-generated certificate is used (see note below), but set can be used to replace this with another certificate.
    * The Server Certificate may be generated by the system itself, and can take up to 5 minutes to generate. This occurs when any of the Service Security Level 92 is set to a value other than *Unsecure Only*. During this time, normal system operation is suspended.
    * Set
    Sets the current Server Certificate and associated private key. The certificate and key must be a matching pair. The source may be:
      * Current User Certificate Store.
      * Local Machine Certificate Store.
      * File in PKCS#12 (.pfx), DER (.cer), or password protected DER (.cer) format.
      * Pasted from clipboard in PEM format, including header and footer text.
    * View
    View the current Server Certificate. The certificate (not the private key) may also be installed into the local PC certificate store for export or later use when running the manager in secured mode.
    * Delete
    Delete the current Server Certificate. When sent to the system will generate a new Server Certificate when next required. This can take up to 5 minutes to generate. During this time, normal system operation is suspended.
  * Client Certificate Checks: *Default = None.*
  When a Service Security Level is set to *High*, a certificate is requested of Manager.The received certificate is tested according to the Client Certificate Checks level:
    * None
    No extra checks are made (The certificate must be in date).
    * Low
    Certificate minimum key size 512 bits, in date.
    * Medium
    Certificate minimum key size 1024 bits, in date, match to store, no reflected.
    * High
    Certificate minimum key size 1024 bits, in date, match to store, no self signed, no reflected.
  * Client IP Office Certificate Store: *Default = Empty.*
  The certificate store contains a set of trusted certificates used to evaluate received client certificates. Up to six X.509v3 certificates may be installed. The source may be:
    * Current User Certificate Store.
    * Local Machine Certificate Store.

- File in PKCS#12 (.pfx), DER (.cer), or password protected DER (.cer) format.
- Pasted from clipboard in PEM format, including header and footer text.

3.5.2.2 Unsecured Interfaces

This tab is accessible when System is selected in the navigation pane. These features relate to applications that access the system configuration settings using older security methods.



- System Password: *Default = 'password', Range = 0 to 31 characters.*
  This password is required by some legacy applications such as Monitor and Call Status. It is also used for control unit software upgrades.

- VM Pro Password: *Default = '', Range = 0 to 31 characters.*
  This password is required if a matching password is also set through the Voicemail Pro client application. Typically no password is set.

- Monitor Password: *Default = '', Range = 0 to 31 characters.*
  This password, if set, is used by the System Monitor and Call Status applications. If this password is not set, those applications use the system password. If changing this password with no previous password set, enter the system password as the old password.

- Applications Controls: *Default = All selected except TFTP Configuration Write.*
  These check boxes control which actions the system will support for legacy applications. Different combinations are used by the different applications. A summary of the applications affected by changes is listed in the Application Support list.

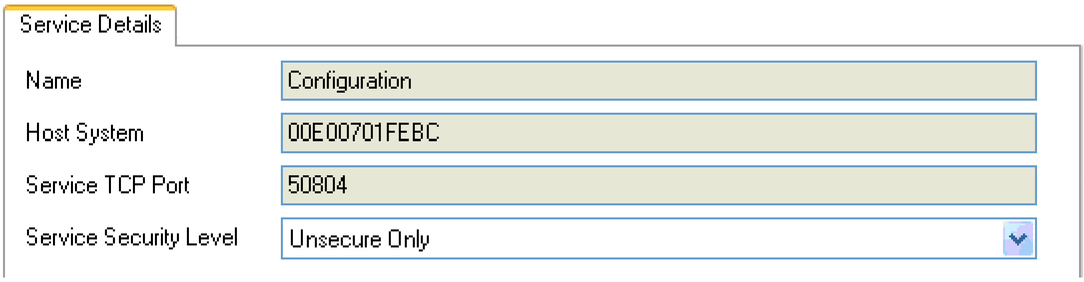
  - TFTP Configuration Read: *Default = On.*
  - TFTP Configuration Write: *Default = Off.*
  - Voicemail: *Default = On.*
  - EConf: *Default = On.*

- Program Code: *Default = On.*
- Real Time Interface: *Default = On.*
  If disabled, IP Office Delta Server and related applications cannot change the status of agents.
- TAPI: *Default = On.*
- HTTP Directory Read: *Default = On. Software level = 5.0+.*
  Allow the system's current directory records to be accessed using HTTP.
- HTTP Directory Write: *Default = On. Software level = 5.0+.*
  Allow HTTP import 171 to be used to place temporary directory entries into the directory.

- Application Support
  This panel is shown for information only. It indicates the effect on various applications of the Application Controls selections.

3.5.3 Services Settings

This tab is accessible when Service is selected in the navigation pane. It shows details of the services that the system runs to which Service Users can communicate.



- Name
  The name of the service. This is a fixed value for indication purposes only.
- Host System
  This field shows the system's name. This is a fixed value for indication purposes only.
- TCP Base Port
  This is the TCP port on which the system listens for attempts to access the service. The routing of traffic to this port may need to be enabled on firewalls and network devices between the Service Users and the system. The TCP Base Port for each service is offset by a fixed amount from the Base TCP Port set in System Settings.

Service	Method	Port Used	Default
Configuration	Unsecure *(IP Office 3.2+)*	Base TCP Port	50804
	Secure *(IP Office 4.1+)*	Base TCP Port plus 1.	50805
System Status Interface	Unsecure *(IP Office 4.0+)*	Base TCP Port plus 4.	50808
Security Administration	Unsecure *(IP Office 3.2+)*	Base TCP Port plus 8.	50812
	Secure *(IP Office 4.1+)*	Base TCP Port plus 9.	50813
Enhanced TSPI	Unsecure *(IP Office 5.0+)*	Base TCP Port plus 10.	50814

- Service Security Level: *Default = 'Unsecure Only'. Software level = 4.1+.*
  Sets the minimum security level the service will support. See File | Preferences | Security 110 for the corresponding Manager application setting, which must be changed to match the appropriate service access security settings.
  - WARNING
    If the system does not already have an X509 security certificate, selecting a setting other than Unsecure Only will cause the system to stop responding for a period (less than a minute) while the system generates its own unique security certificate.

  - *Unsecure Only*
    This option allows only unsecured access to the service. The service's secure TCP port is disabled.
  - *Unsecure + Secure*
    This option allows both unsecured and secure (Low) access. In addition, TLS connections are accepted without encryption, just authentication.
  - *Secure, Low*
    This option allows secure access to that service using TLS, and demands weak (for example DES_40 + MD5) encryption and authentication or higher. The service's unsecured TCP port is disabled.
  - *Secure, Medium*
    This option allows secure access to that service using TLS, and demands moderate (for example DES_56 + SHA-1) encryption and authentication or higher. The service's unsecured TCP port is disabled.
  - *Secure, High*
    This option allows secure access to that service using TLS and demands strong (for example 3DES + SHA-1) encryption and authentication, or higher. In addition, a certificate is required from the client (usually Manager). See System Details | Client Certificate Checks 87 for tests made on the received certificate. The service's unsecured TCP port is disabled.

Services

The different services are used as follows:

- **Configuration**
  This service is used access to control unit configuration settings by the Manager application. If set to a mode other than secure, the Manager's Security preferences 110 also need to changed to match.

- **System Status Interface**
  This service is used by connections to the System Status Application and the IP Office Customer Call Reporter applications.

- **Security Administration**
  This service is used for access to control unit security settings by the Manager applications.

- **Enhanced TSPI**
  This service is used by connections to the one-X Portal for IP Office application.

- **HTTP**
  The HTTP service affects all HTTP connections provided by the system. Changing its setting will affect applications other than just the DECT R4. The default Service Security Level is *Secure + Unsecure*, meaning both http and https can be used.

  - **Unsecure** = HTTP port 80 available.
    This is used for phone files, embedded file manager, system file upgrade, one-X Portal directory services, DECT R4 provisioning, IPO softphone provisioning.

  - **Secure** = HTTPS port 443 available.
    This can be used for DECT R4 provisioning, IPO softphone provisioning.

3.5.4 Rights Groups

The Rights Groups to which a service user belongs define what the service user is able to do after that login to the system.

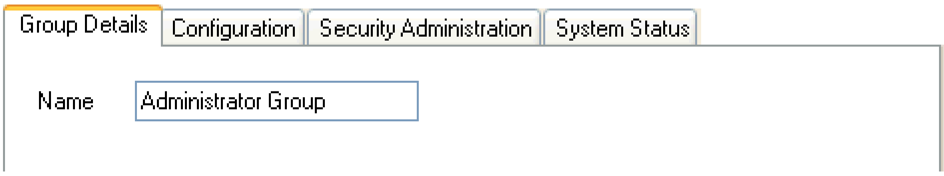
Default Rights Groups		Administrator Group	Manager Group	Operator Group	System Status Group	TCPA Group	IPDECT Group
Operator Rights	Operator View	Administrator	Manager	Operator	–	–	–
Service Rights	Read Configuration	✓	✓	✓	✗	✗	✗
	Write Configuration	✓	✓	✓	✗	✗	✗
	Default Configuration	✓	✓	✓	✗	✗	✗
	Merge	✓	✓	✓	✗	✗	✗
	Reboot Immediately	✓	✓	✓	✗	✗	✗
	Reboot When Free	✓	✓	✓	✗	✗	✗
	Reboot at Time of Day	✓	✓	✓	✗	✗	✗
Security Administration	Read All Security Settings	✗	✗	✗	✗	✗	✗
	Write All Security Settings	✗	✗	✗	✗	✗	✗
	Reset All Security Settings (IP Office 4.1+)	✗	✗	✗	✗	✗	✗
	Write own service user password (IP Office 4.1+)	✗	✗	✗	✗	✗	✗
System Status (IP Office 4.0+)	System Status Access	✗	✗	✗	✓	✗	✗
	Read all configuration	✗	✗	✗	✓	✗	✗
	System Control (IP Office 6.0+)	✗	✗	✗	✓	✗	✗
Enhanced TSPI (IP Office 5.0+)	Enhanced TSPI Access	✗	✗	✗	✗	✓	✗
HTTP	DECT R4 Provisioning	✗	✗	✗	✗	✗	✓

- Administrator Group/Manager Group/ Operator Group:
  These default groups are used for configuration access. They differ only in the level of rights to view and amend configuration 96 settings.
- System Status Group:
  This is the default group intended for use with the System Status Application (SSA). It is also used by the Customer Call Reporter application unless a separate rights group is created for that application.
- TCPA Group: *Software level = 5.0+*
  This rights group is used by the one-X Portal for IP Office application for connection to the system.

Some of the groups above have been added in different software releases. For new systems and systems with default security settings, the new rights groups are created automatically when the system is upgraded to the new release. For systems which have customized (non default) security settings, these rights groups may have to be created manually.

3.5.4.1 Group Details

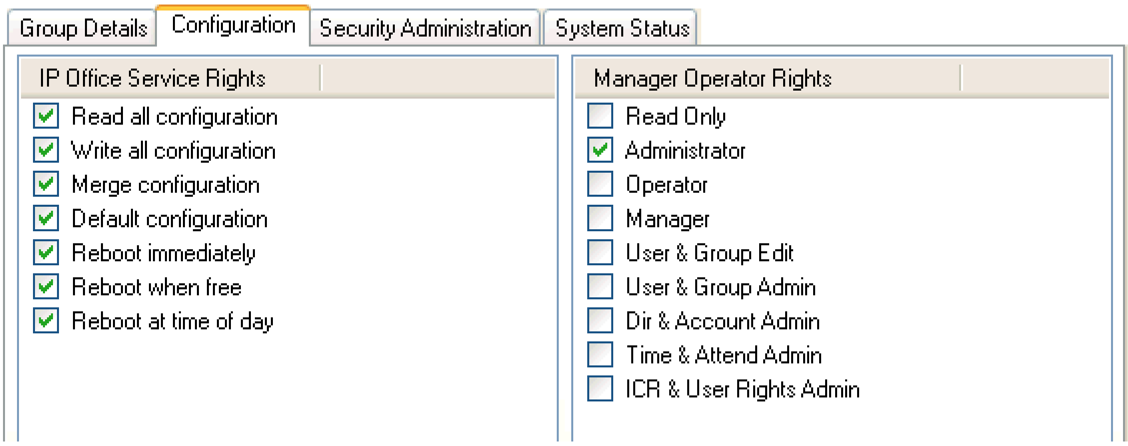
These settings are displayed when Rights Groups is selected in the navigation pane. This tab sets the name of the Rights Group.



- Name: *Range = Up to 31 characters*
  The name for the Rights Group should be unique.

3.5.4.2 Configuration

These settings are displayed when Rights Groups is selected in the navigation pane. This tab sets the configuration settings access for Service User's who are members of this Rights Group.



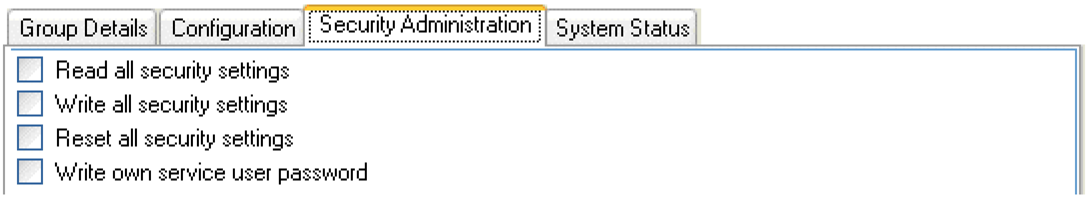
- **IP Office Service Rights**
  This setting controls what action on the system can be performed by members of the Rights Group.

- **Manager Operator Rights**
  This setting controls what types of configuration entries Manager will allow members of the Rights Group to viewed and what actions they can perform with those types of entries.

Operator	View / Edit / New / Delete	Configuration Entry Types
Administrator	All	View, edit create and delete all configuration entries.
Manager	View	View all except WAN Port.
	Edit	Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, Directory, Time Profile, Firewall Profile, IP Route, Least Cost Route, Account Code, ARS, E911 System.
	New	
	Delete	As edit except Short Code.
Operator	View	View all except WAN Port.
	Edit	Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, Time Profile, Firewall Profile, IP Route, Least Cost Route, Account Code, License, ARS, E911 System.
	New	None.
	Delete	Delete Incoming Call Route and Directory.
User & Group Edit	View	User and Hunt Group entries only.
	Edit	
	New	None
	Delete	
User & Group Admin	All	User and Hunt Group entries only.
Dir & Account Admin	All	Directory and Account Code entries only.
Time & Attendant Admin	All	Time Profile and Auto Attendant entries only.
ICR & User Rights Admin	All	Incoming Call Route and User Rights entries only.
Read Only	View	View all configuration entries.
	Edit	None.
	New	
	Delete	

3.5.4.3 Security Administration

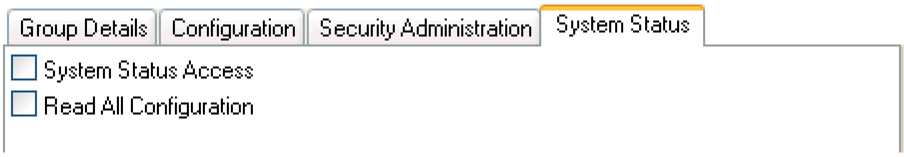
These settings are displayed when Rights Groups is selected in the navigation pane. This tab sets the security settings access for Service user's who are members of this Rights Group. These settings are ignored and greyed out if a Unique Security Administrator has been enabled in General Settings 85.



- Read all security settings
  Members of the Rights Group can view the system's security settings.
- Write all security settings
  Members of the Rights Group can edit and return changes to the system's security settings.
- Reset all security settings: *Software level = 4.1+.*
  If selected, members of the Rights Group can reset the security settings to default values.
- Write own service user password: *Software level = 4.1+.*
  If selected, members of the Rights Group can change their own password when requested to do so by the system. That request may be the result of a Password Change Period 85, Force new password 99 or Account Expiry 99. The new password change is requested automatically at login time.

3.5.4.4 System Status

These settings are displayed when Rights Groups is selected in the navigation pane. This tab sets whether members of the group can access the system using the System Status Application (SSA). That application is only supported by Manager 4.0 and higher systems.



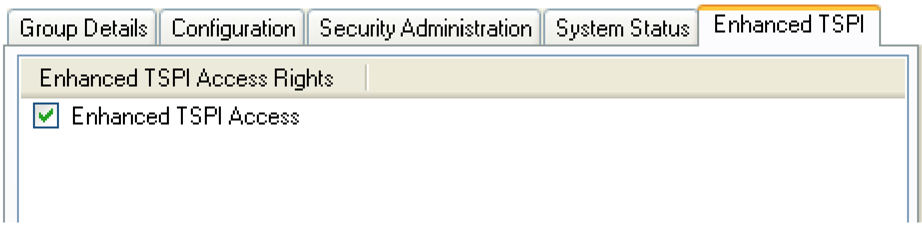
- System Status Access
  If selected, members of the Rights Group can view the system's current status and resources using the System Status Application (SSA).

- Read all configuration
  The System Status application includes tools to take a snapshot of the system for use by Avaya for diagnostics. That snapshot can include a full copy of the system's configuration settings. This setting must be enabled for the SSA user to include a copy of the configuration in the snapshot.

- System Control: *Software level = 6.0+*
  If enabled, the SSA user is able to use SSA to initiate system shutdowns and memory card shutdown/restarts.

3.5.4.5 Enhanced TSPI

These settings are displayed when Rights Groups is selected in the navigation pane. This tab sets whether members of the group can access the system using the Enhanced TSPI application interface. This interface is only supported by IP Office Release 5.0 and higher systems.

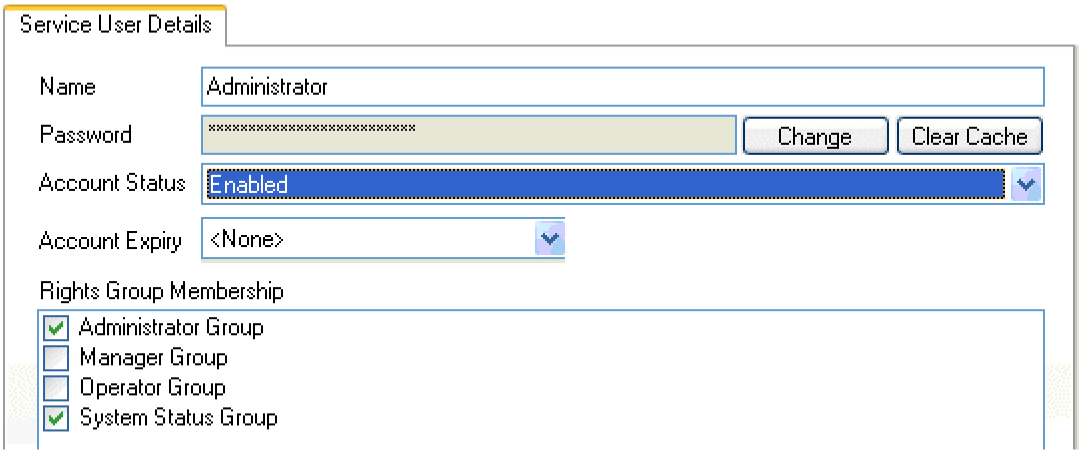


- Enhanced TSPI Access
  If selected, applications in this rights group are able to uses the system's Enhanced TSPI interface. This interface is currently used by the one-X Portal for IP Office application server for its connection to the system.

3.5.5 Service User Settings

These settings are displayed when Service Users is selected in the navigation pane and a particular Service User is selected in the group pane.

Users can be created and deleted using the and icons at the top-right of the details pane. The maximum number of Service Users is 16.



- Name: *Range = Up to 31 characters.*

  Sets the Service User's name. The minimum name length is controlled through General settings 85.

  - Note: If changing the user name and/or password of the current service user used to load the security settings, after saving the changes Manager should be closed. Not closing Manager will cause error warnings when attempting to send any further changes.

- Password: *Range = Up to 31 characters.*

  Sets the Service User's password. The minimum password length and complexity is controlled through General settings 85.

- Account Status: *Default = 'Enabled'. Software level = 4.1+.*
  Displays the current service user account status (correct at the time of reading from the system).

  - *Enabled*
    This status is the normal non-error state of a service user account. This setting can be selected manually to re-enable an account that has been disabled or locked. Note that re-enabling a locked account will reset all timers relating to the account such as Account Idle Time.

  - *Force New Password*
    This status can be selected manually. The service user is then required to change the account password when they next log in. Until a password change is successful, no service access is allowed. Note that the user must be a member of a Rights Group that has the Security Administration 97 option Write own service user password enabled.

  - *Disabled*
    This status prevents all service access. This setting can be selected manually. The account can be enabled manually by setting the Account Status back to *Enabled*.

- Locked – Password Error
  This status indicates the account has been locked by the Password Reject Action option *Log and Disable Account* on the security General Settings 85 tab. The account can be enabled manually by setting the Account Status back to *Enabled*.

- *Locked - Temporary*
  This status indicates the account is currently locked temporarily by the Password Reject Action option *Log and Temporary Disable* on the security General Settings 85 tab. The account can be enabled manually by setting the Account Status back to *Enabled*, otherwise the service user must wait for the 10 minute period to expire.

- *Locked - Idle*
  This status indicates the account has been locked by passing the number of days set for the Account Idle Time on the security General Settings 85 tab without being used. The account can be enabled manually by setting the Account Status back to *Enabled*.

- *Locked - Expired*
  This status indicates the account has been locked after passing the Account Expiry date set below. The account can be enabled manually by setting Account Status back to *Enabled*, and resetting the Account Expiry date to a future date or to *No Account Expiry*.

- *Locked – Password Expired*
  This status indicates the account has been locked after having not been changed within the number of days set by the Password Change Period option on the security General Settings 85 tab. The account can be enabled manually by setting the Account Status back to *Enabled*.

- Account Expiry: *Default = <None> (No Expiry). Software level = 4.1+.*
  This option can be used to set a calendar date after which the account will become locked. The actual expiry time is 23:59:59 on the selected day. To prompt the user a number of days before the expiry date, set an Expiry Reminder Time on the security General Settings 85 tab.

- Rights Group Membership
  The check boxes are used to set the Rights Groups to which the user belongs. The user's rights will be a combination of the rights assigned to the groups to which they belong.

Chapter 4.
Menu Bar Commands

4. Menu Bar Commands

The commands available through the Manager's menu bar change according to the mode in which Manager is running. Commands may also be grayed out if not currently applicable. The following sections outline the functions of each command. The Edit and Help menus are not included.

Configuration Mode

File	Open Configuration... 104		
	Close Configuration 104		
	Save Configuration 104		
	Save Configuration As... 104		
	Change Working Directory... 105		
	Preferences 106		
	Offline	Create New Config 112	
		Open File... 112	
		Send Config... 112	
		Receive Config... 112	
	Advanced	Erase Configuration (Default) 113	
		Reboot 113	
		System Shutdown... 114	
		Upgrade.. 115	
		Change Mode... 116	
		Audit Trail... 118	
		Security Settings... 119	
		Erase Security Settings... 119	
		Embedded File Management 119	
		Format IP Office SD Card 121	
		Recreate IP Office SD Card 122	
		Memory Card Command 125	Shutdown... 125
			Start Up... 125
		Voicemail Pro 124	
		System Status 124	
		LVM Greeting Utility 124	
	Backup/Restore	Backup Binaries and Configurations 125	
		Restore Binaries and Configurations 125	
	Import/Export	Import 127	
		Export 126	
	Exit 127		
View	Toolbars 128		
	Navigation Pane 128		
	Group Pane 128		
	Details Pane 128		
	Error Pane 128		
	Simplified View 128		
	TFTP Log 129		
Tools	Extension Renumber 130		
	Line Renumber 130		
	Connect To 130		
	SCN Service User Management 130		
	Busy on Held Validation 131		
	MSN 132 Configuration 132		
	Print Button Labels 133		

Security Mode

File	Open Security Settings 134 Close Security Settings 134 Save Security Settings 134 Reset Security Settings 134 Preferences 134 Configuration 134 Exit 134
View	Toolbars 128 Navigation Pane 128 Group Pane 128 Details Pane 128

Embedded File Management

File	Open File Settings 119 Close File Settings 119 Refresh File Settings 135 Upload File 119 Upload System Files 135 Backup System Files 135 Restore System Files 135 Upgrade Binaries 135 Upgrade Configuration 135 Upload Voicemail Files 136 Upload Phone Files 136 Copy System Card 136 Preferences 106 Configuration 104 Exit 127
View	Toolbars 128 Tiles 119 Icons 119 List 119 Details 119

4.1 Configuration Mode

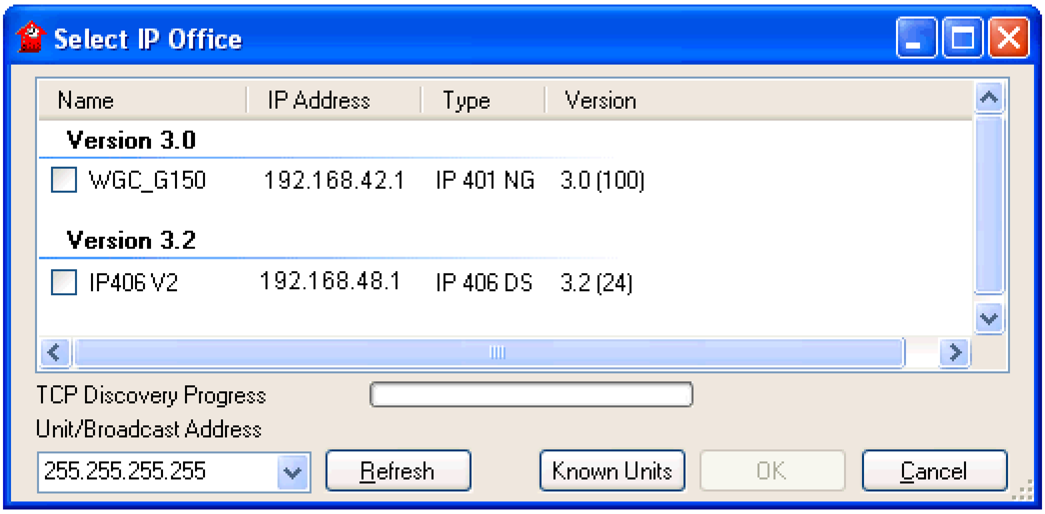
These commands are available when the Manager is in configuration mode.

4.1.1 File Menu

4.1.1.1 Open Configuration

This command displays the Select IP Office menu used to receive a systems configuration settings. See Loading a Configuration 55.

The same action is performed by the icon in the Main Toolbar.



The Select IP Office menu is also used for other actions such as reboot and sending a configuration. If the unit required is not found, the Unit/Broadcast Address can be changed and then Refresh clicked. To change the TCP addresses scanned, select File | Preferences | Discovery 106 and enter the required addresses in the IP Search Criteria.

Known Units is not available unless configured, see Known System Discovery 59.

4.1.1.2 Close Configuration

This command closes the currently loaded configuration without saving it.

4.1.1.3 Save Configuration

The File | Save command saves the amended configuration.

If the configuration has been received from a system, the Send Config menu is displayed. See Sending a Configuration 64.

If the configuration file has been opened offline or created from new, the file is saved to disk only.

4.1.1.4 Save Configuration As

The File | Save As command allows you to save a configuration a file on the Manager computer. The command displays the Save File As dialog box. You can enter the new file name, including the drive and directory.

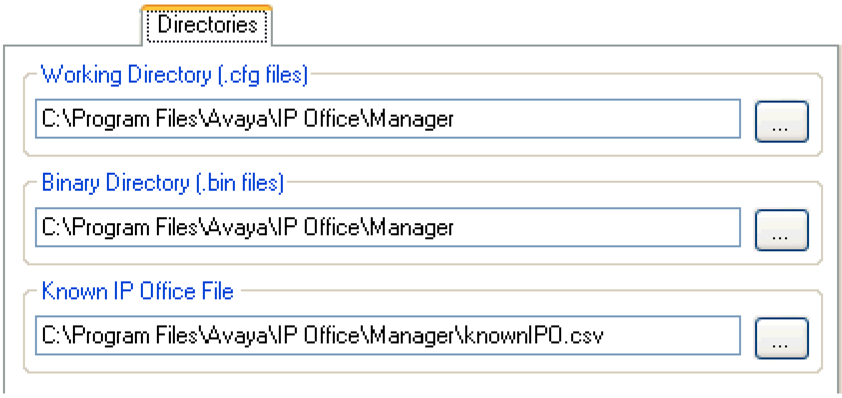
Configurations saved onto the PC in this way can be reopened using the icon or the File | Offline | Open File 112 command.

Note that dynamic configuration data, for example advertised hunt groups on other systems, is not included in a configuration file saved onto PC and then reopened.

4.1.1.5 Change Working Directory

This command allows you to change the default locations where Manager looks for and saves files.

These fields set the default location where Manager will look for and save files. This tab is also accessed by the File | Change Working Directory command.



- Working Directory (.cfg files)
  Sets the directory into which Manager saves *.cfg* files. By default this is the Manager application's program directory.
  - This folder location is also used for the Recreate IP Office SD Card command. It uses sub-folders of the \Memory Cards folder at the specified locations. If the Working Directory is changed to a location without an appropriate set of *\Memory Cards* sub-folders, the required set of files will not be copied onto the SD card.
- Binary Directory (.bin files)
  Sets the directory in which the Manager upgrade wizard, HTTP, TFTP and BOOTP functions look for firmware files requested by phones, expansion module, control units and other hardware components. That includes *.bin* file, *.scr* files and *.txt* files. By default this is the Manager application's program directory.
  - Note that in the Upgrade Wizard 115, right-clicking and selecting Change Directory also changes this setting.
- Known IP Office File: *Software level = 4.0 Q2 2007 maintenance release+.*
  Sets the file and directory into which Manager can record details of the systems it has discovered. Once a file location has been specified, a Known Units 59 button becomes available on the discovery menu used for loading system configuration. Pressing that button displays the known units file as a list from which the required system can be selected. It also allows sorting of the list and entries to be removed.

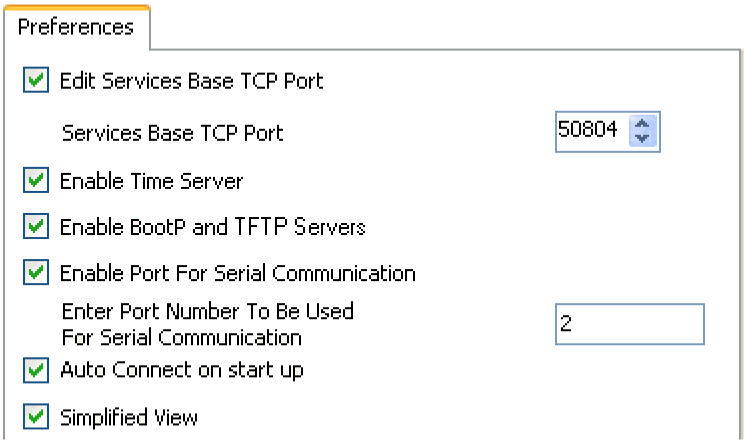
4.1.1.6 Preferences

This command displays a menu for configuring various aspects of Manager's operation. The menu is divided into a number of tabs.

- Preferences 106
- Directories 107
- Visual Preferences 109
- Discovery 108
- Validation 112
- Security 110

4.1.1.6.1 Preferences

This tab is accessed through File | Preferences and then selecting the Preferences sub-tab.

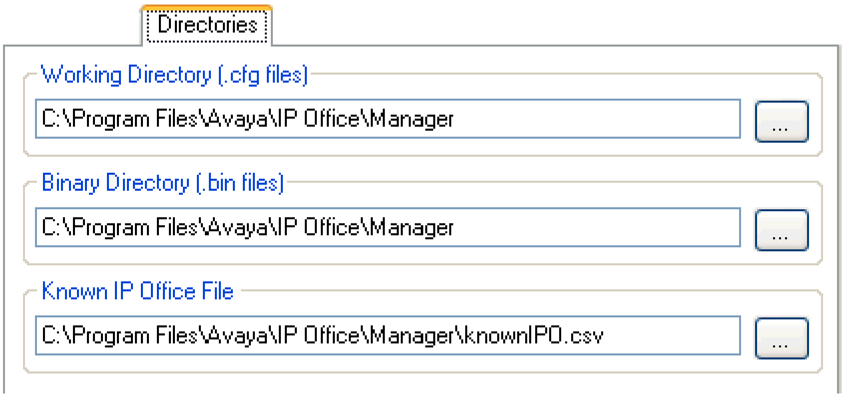


- Edit Services Base TCP Port: *Default = On.*
  This field shows or hides the Service Base TCP Port setting.
  - Service Base TCP Port: *Default = 50804. Software level = 3.2+.*
    Access to the configuration and security settings on a system requires Manager to send its requests to specific ports. This setting allows the TCP Base Port used by Manager to be set to match the TCP Base Port setting of the system. The system's TCP Base Port is set through its security settings.
- Enable Time Server: *Default = On.*
  This setting allows Manager to respond to RFC868 Time requests from systems. It will provide the system with both the UTC time value and the local time value of the PC on which it is running. See Date and Time 727.
- Enable BootP and TFTP Servers: *Default = On.*
  This setting allows Manager to respond to BOOTP request from systems for which it also has a matching BOOTP entry. It also allows Manager to respond to TFTP requests for files.
- Enable Port for Serial Communication
  Not used. This is a legacy feature for some older control units that were managed via the serial port rather than the LAN.
  - Enter Port Number to be used for Serial Communication
    Used with the setting above to indicate which serial port Manager should use.
- Auto Connect on start up: *Default = On*
  If on, when Manager is started it will automatically launch the Select IP Office menu and display any discovered systems. If only one system is discovered, Manager will automatically display the login request for that system or load its configuration if the security settings are default.
- Simplified View: *Default = On*
  If on, the Manager will start in simplified view 18 mode if no configuration is loaded.

4.1.1.6.2 Directories

This tab is accessed through File | Preferences and then selecting the Directories sub-tab.

These fields set the default location where Manager will look for and save files. This tab is also accessed by the File | Change Working Directory command.

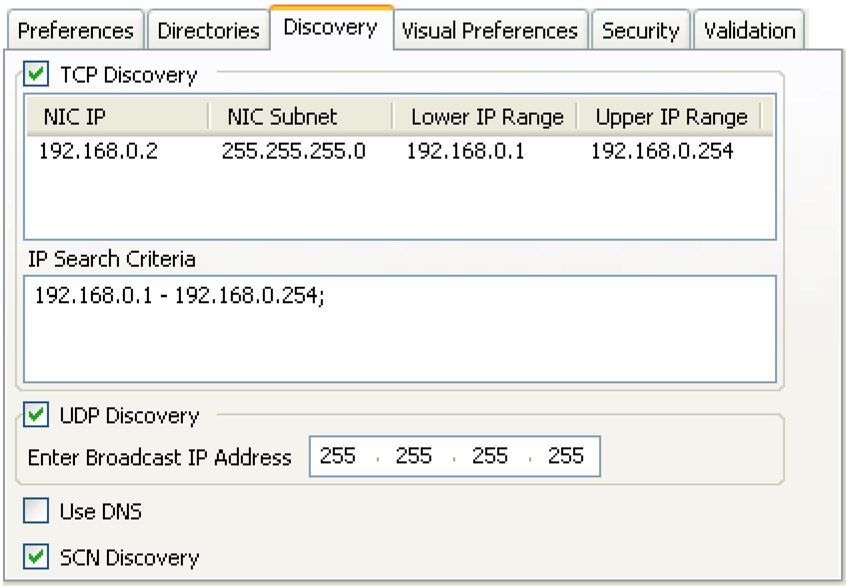


- Working Directory (.cfg files)
  Sets the directory into which Manager saves *.cfg* files. By default this is the Manager application's program directory.
  - This folder location is also used for the Recreate IP Office SD Card command. It uses sub-folders of the \Memory Cards folder at the specified locations. If the Working Directory is changed to a location without an appropriate set of *\Memory Cards* sub-folders, the required set of files will not be copied onto the SD card.
- Binary Directory (.bin files)
  Sets the directory in which the Manager upgrade wizard, HTTP, TFTP and BOOTP functions look for firmware files requested by phones, expansion module, control units and other hardware components. That includes *.bin* file, *.scr* files and *.txt* files. By default this is the Manager application's program directory.
  - Note that in the Upgrade Wizard 115, right-clicking and selecting Change Directory also changes this setting.
- Known IP Office File: *Software level = 4.0 Q2 2007 maintenance release+.*
  Sets the file and directory into which Manager can record details of the systems it has discovered. Once a file location has been specified, a Known Units 59 button becomes available on the discovery menu used for loading system configuration. Pressing that button displays the known units file as a list from which the required system can be selected. It also allows sorting of the list and entries to be removed.

4.1.1.6.3 Discovery

This tab is accessed through File | Preferences and then selecting the Discovery sub-tab.

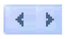
These settings affect the Select IP Office menu used by Manager to discovery systems. By default IP Office 3.2 systems respond to both UDP and TCP discovery. Pre-3.2 systems only support UDP discovery.



- TCP Discovery: *Default = On. Software level = 3.2+.*
  This setting controls whether Manager uses TCP to discover systems. The addresses used for TCP discovery are set through the IP Search Criteria field below.
  - NIC IP/NIC Subnet
    This area is for information only. It shows the IP address settings of the LAN network interface cards (NIC) in the PC running Manager. Double-click on a particular NIC to add the address range it is part of to the IP Search Criteria. Note that if the address of any of the Manager PC's NIC cards is changed, the Manager application should be closed and restarted.
  - IP Search Criteria
    This section is used to enter TCP addresses to be used for the TCP discovery process. Individual addresses can be entered separated by semi-colons, for example *135.164.180.170; 135.164.180.175*. Address ranges can be specified using dashes, for example *135.64.180.170 - 135.64.180.175*.
- UDP Discovery: *Default = On*
  This settings controls whether Manager uses UDP to discover systems. Pre-3.2 systems only respond to UDP discovery. By default IP Office 3.2 and higher systems also respond to UDP discovery but that can be disabled through the system's security settings.
  - Enter Broadcast IP Address: *Default = 255.255.255.255*
    The broadcast IP address range that Manager should used during UDP discovery. Since UDP broadcast is not routable, it will not locate systems that are on different subnets from the Manager PC unless a specific address is entered.
- Use DNS: *Software level = Manager 6.2+.*
  Selecting this option allows Manager to use DNS name (or IP address) lookup to locate a system. Note that this overrides the use of the TCP Discovery and UDP Discovery options above. This option requires the system IP address to be assigned as a name on the users DNS server. When selected, the Unit/Discovery Address field on the Select IP Office 55 dialogue is replaced by a Enter Unit DNS Name or IP Address field.
- SCN Discovery: *Software level = Manager 8.1+.*
  If enabled, when discovering systems, the list of discovered systems will group systems in the same Small Community Network and allow them to be loaded as a single configuration. At least one of the systems in the SCN must be running Release 6.0 or higher software. See SCN Management 817. This does not override the need for each system in the SCN to also be reachable by the TCP Discovery and or UDP Discovery settings above and accessible by the router settings at the Manager location.

4.1.1.6.4 Visual Preferences

This tab is accessed through File | Preferences and then selecting the Visual Preferences sub-tab.

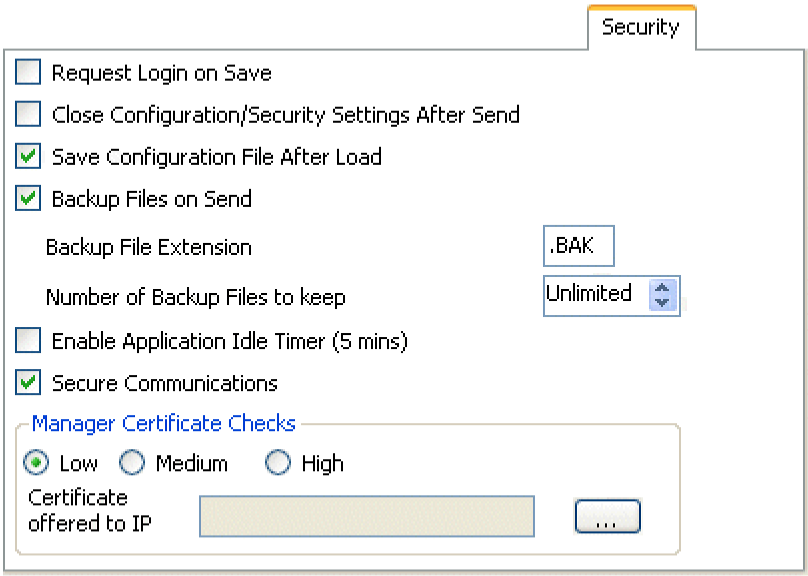
- Icon size
  Sets the size for the icons in the navigation pane between *Small*, *Medium* or *Large*.
- Multiline Tabs: *Default = Off.*
  In the details pane, for entry types with more than two tabs, Manager can either use  buttons to scroll the tabs horizontally or arrange the tabs into multiple rows. This setting allows selection of which method Manager uses.

4.1.1.6.5 Security

This tab is accessed through File | Preferences and then selecting the Security sub-tab.

Controls the various security settings of Manager. To control the security settings of the system, see Security Mode 68.

All settings, except Secure Communications, can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.



- Request Login on Save: *Default = On*
  By default a valid user name and password is required to receive a configuration from a system and also to send that same configuration back to the system. Deselecting this setting allows Manager to send the configuration back without having to renter user name and password details. This does not apply to a configuration that has been saved on PC and then reopened. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

- Close Configuration/Security Settings After Send: *Default = On.*
  When selected, the open configuration file or security settings are closed after being sent back to the system. This setting does not affect SCN management mode which always closes the configuration after saving.

- Save Configuration File After Load: *Default = On.*
  When selected, a copy of the configuration is saved to Manager's working directory 107. The file is named using the system name and the suffix *.cfg*. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

- Backup Files on Send: *Default = On.*
  If selected, whenever a copy of a configuration is sent to a system, a backup copy is saved in Manager's working directory 107. The file is saved using the system name, date and a version number followed by the Backup File Extension as set below. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

- Backup File Extension: *Default = .BAK*
  Sets the file extension to use for backup copies of system configurations generated by the Backup Files on Send option above.

- Number of Backup Files to keep: *Default = Unlimited. Software level = 4.2+.*
  This option allows the number of backup files kept for each system to be limited. If set to a value other then Unlimited, when that limit would be exceeded, the file with the oldest backup file is deleted.

- Enable Application Idle Timer (5 minutes): *Default = Off. Software level = 4.1+.*
  When enabled, no keyboard or mouse activity for 5 minutes will cause the Manager to grey out the application and re-request the current service user password. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

- Secure Communications: *Default = Off. Software level = 4.1+.*
  When selected, any service communication from Manager to the system uses the TLS protocol. This will use the ports set for secure configuration and secure security access. It also requires the configuration and or security service within the system's security configuration settings to have been set to support secure access. Depending on the level of that secure access selected, it may be necessary for the Manager Certificate Checks below to be configured to match those expected by the system for configuration and or security service. See Security Administration 74.

  - When Secure Communications is set to *On*, a padlock icon is displayed at all times in the lower right Manager status field.

- Manager Certificate Checks: *Software level = 4.1+.*
  When the Secure Communications option above is used, Manager will process and check the certificate received from the system. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.
  - *Low*
    Any certificate sent by the system is accepted without question.
  - *Medium*
    Any certificate sent by the system is accepted if it has previously been previously saved in the Windows' certificate store. If the certificate has not been previously saved, the user has the option to review and either accept or reject the certificate.
  - *High*
    Any certificate sent by the system is accepted if it has previously been previously saved in the Windows' certificate store. Any other certificate cause a log in failure.
  - Certificate Offered to IP Office: *Default = none*
    Specifies the certificate used to identify Manager when the Secure Communications option is used and the system requests a certificate. Use the Set button to change the selected certificate. Any certificate selected must have an associated private key held within the store:
    - Select from Current User certificate store - Display certificates currently in the currently logged-in user store.
    - Select from Local Machine certificate store.
    - Remove Selection – do not offer a Manager certificate.

Security – Registry Settings

- WARNING: Changing Windows Registry Settings
  Avaya accept no liability for any issues arising from the editing of a PC's registry settings. If you are in any doubt about how to perform this process you should not proceed. It is your responsibility to ensure that the registry is correctly backed up before any changes are made.

NOTE: Before manually editing any registry entry, the following Microsoft support articles should be read:

- http://support.microsoft.com/kb/256986
- http://www.microsoft.com/resources/documentation/windows/xp/all/proddocs/en-us/regedit_permit_key.mspx

Manager stores it's security preferences in the Windows Registry. The following key affects manager security operation; it's values may only be changed by a configuration or security administrator:

- *HKEY_CURRENT_USER\Software\Avaya\IP400\Manager\Security*

In order to prevent circumvention by manual editing of the Windows Registry, Regedt32.exe, the native registry editor, allows an operator user (with Full Control permissions) to edit permissions on a per key basis.

To prevent a user from manually editing the security preferences, the HKEY_USERS\User GUID\Software\Avaya\IP400\Manager\Security key permission should be set to 'Read' only for that user. Ensure that all child object permissions are replaced as well by using the 'Advanced' button.

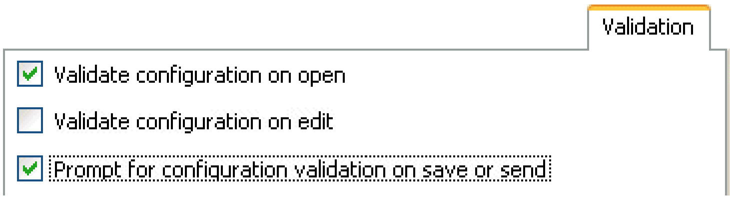
To allows the security policy of all local PC users to be fixed, a set of values in the key HKEY_CURRENT_USER\Software\Avaya\IP400\Manager\Security\ may be created. This is tested and used in preference to any value found under HKEY_CURRENT_USER\Software\Avaya\IP400\Manager\Security\.

This key is not created by the manager application.

4.1.1.6.6 Validation

This tab is accessed through File | Preferences and then selecting the Validation sub-tab.

By default Manager validates the whole configuration when it is loaded and individual fields whenever they are edited. This tab allows selection of when automatic validation should be applied to configuration files loaded into Manager.



- **Validate configuration on open**
  Automatically validate configuration files when they are opened in Manager.

- **Validate configuration on edit**
  Validate the whole configuration when OK is clicked after editing a record. For large configurations, disabling this option removes the delay caused by validating the configuration after every edit.

- **Prompt for configuration validation on save or send**
  If selected, when saving or sending a configuration, a prompt is displayed asking whether the configuration should be validated. If validation is selected and error are found, the send or save process is canceled. This option is disabled if Validate configuration on edit is selected.

4.1.1.7 Offline

4.1.1.7.1 Create New Config

This command starts a dialog that allows you to create a default offline configuration by specifying the system locales, the type of control unit and expansion modules and the trunk cards fitted. See Creating a New Configuration 60.

The same action is performed by the icon in the Main Toolbar.

4.1.1.7.2 Open File

This command allows a configuration file stored on PC to be opened in Manager.

4.1.1.7.3 Send Config

This command is used to send an offline configuration to a system. See Sending a Configuration 64.

4.1.1.7.4 Receive Config

This command displays the Select IP Office menu used to receive a systems configuration settings. See Loading a Configuration 55.

Once the configuration has been received, you are prompted to save it on the PC.

4.1.1.8 Advanced

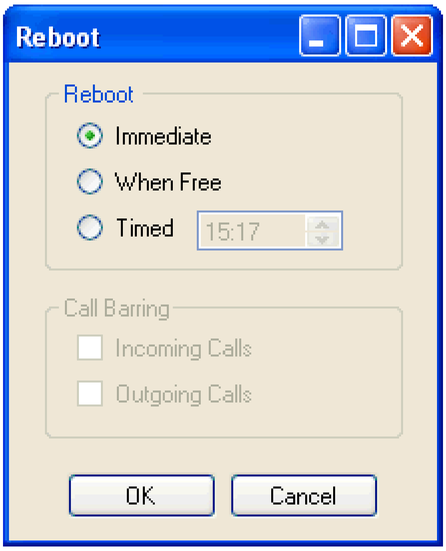
4.1.1.8.1 Erase Configuration

This command returns the configuration settings of a system back to their default values 65. It does not affect the system's security settings or audit trail record.

When this command is used, the Select IP Office menu is displayed. Once a system is selected, a valid configuration user name and password are required to complete the action.

4.1.1.8.2 Reboot

When this command is used, the Select IP Office menu is displayed. Once a system is selected, a valid user name and password are required. The type of reboot can then be selected.



- **Reboot**
  If Manager thinks the changes made to the configuration settings are mergeable, it will select Merge by default, otherwise it will select Immediate.
  - **Immediate**
    Send the configuration and then reboot the system.
  - **When Free**
    Send the configuration and reboot the system when there are no calls in progress. This mode can be combined with the Call Barring options.
  - **Timed**
    The same as When Free but waits for a specific time after which it then wait for there to be no calls in progress. The time is specified by the Reboot Time. This mode can be combined with the Call Barring options.
- **Reboot Time**
  This setting is used when the reboot mode Timed is selected. It sets the time for the Manager reboot. If the time is after midnight, the system's normal daily backup is canceled.
- **Call Barring**
  These settings can be used when the reboot mode When Free is selected. They bar the sending or receiving of any new calls.

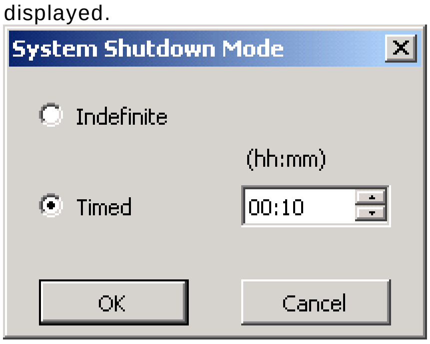
4.1.1.8.3 System Shutdown

IP Office Release 6.0+: This command can be used to shutdown systems. The shut down can be either indefinite or for a set period of time after which the system will reboot.

! WARNINGS

- A shutdown must always be used to switch off the system. Simply removing the power cord or switching off the power input may cause errors.
- This is not a polite shutdown, any users calls and services in operation will be stopped. Once shutdown, the system cannot be used to make or receive any calls until restarted.
- The shutdown process takes up to a minute to complete. When shutdown, the CPU LED and the control unit base card LEDs 1 and 9 (if trunk daughter card fitted) will flash red rapidly. The memory card LEDs are extinguished. Do not remove power from the system or remove any of the memory cards until the system is in this state.
- To restart a system when shutdown indefinitely, or to restart a system before the timed restart, switch power to the system off and on again.

1. Once you have selected the system from the **Select IP Office** menu, the **System Shutdown Mode** menu is displayed.

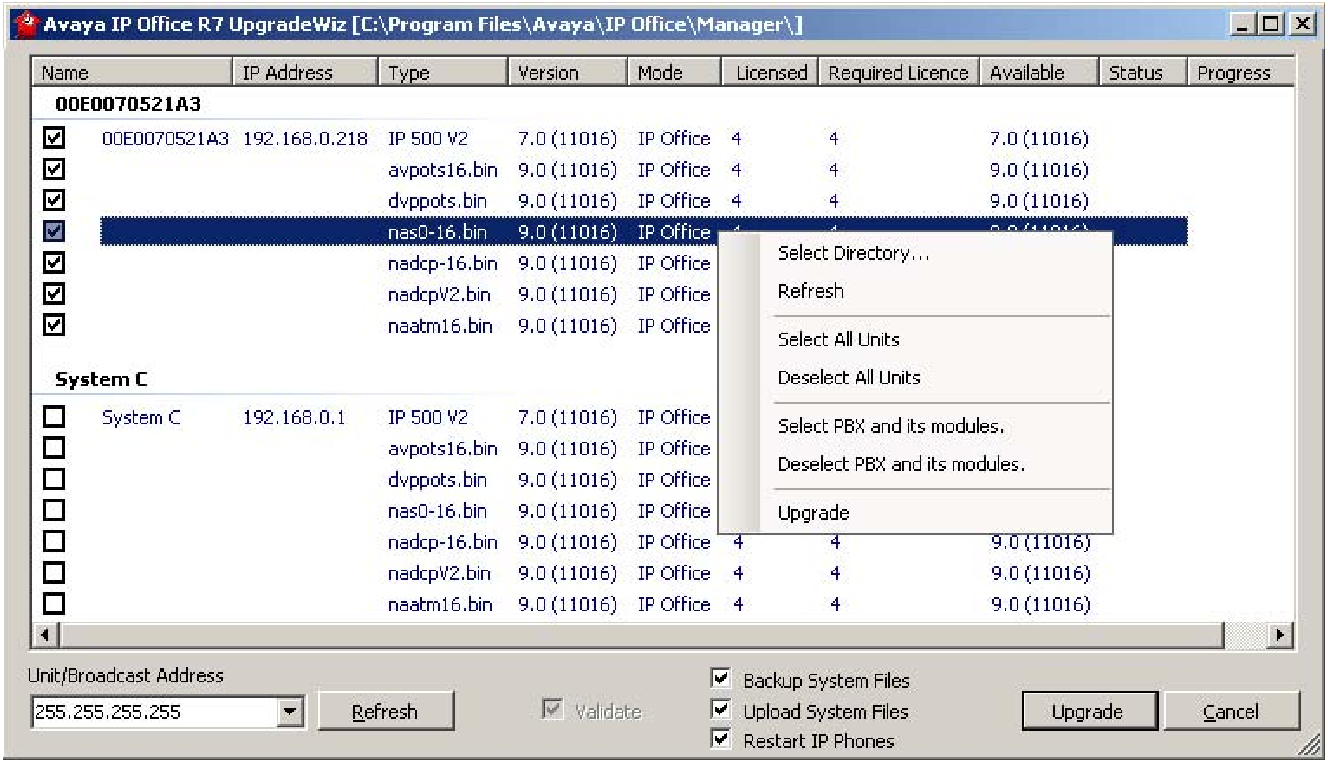


2. Select the type of shutdown required. If **Indefinite** is used, the system can only be restarted by having its power switched off and then on again. If a **Timed** shutdown is selected, the system will reboot after the set time has elapsed.

4.1.1.8.4 Upgrade

This command starts the Upgrade Wizard tool. This tool is used to compare the software level of the control unit and expansion modules within systems against the software level of the .bin binary files Manager has available. The Upgrade Wizard can then be used to select which units to upgrade.

- WARNING
  Incorrect use of the upgrade command can halt system operation and render units in the system unusable. You must refer to the Technical Bulletins for a specific release for full details of performing software upgrades to that release.
- Performing any other actions on a system during an upgrade or closing the upgrade wizard and Manager during an upgrade may render systems unusable.
- During an upgrade the system may restrict calls and services. It will reboot and disconnect all current calls and services.
- The Validate option must remain selected wherever possible. Use of unvalidated upgrades is subject to a number of conditions outlined in the Manager Installation Manual and Technical Bulletins.



The list area shows details of systems found by the Upgrade Wizard and the software currently held by that system.

- The check boxes are used to select which units should be upgraded. Upgrading will require entry of a valid name and password for the selected system.
- The Version column details the current software each unit in the systems is running.
- The Available column shows the version of .bin file Manager has available for that type of unit (a – indicates no file available).
- The Mode column indicates the operation mode of the system.
- The Licensed column indicates the highest value software upgrade license present in the systems configuration. The Required License column indicates the software upgrade license required for the current level of software it is running. It does not refer to the software upgrade license required for the level of software which is available for upgrade. IP500v2 systems, a value of 255 indicates that the control unit is still in its initial 90 days where it can be upgraded to a higher level without requiring a license.
- The Validate option should remain selected wherever possible. When selected, the upgrade process is divided as follows: transfer new software, confirm transfer, delete old software, restart with new software. If Validate is not selected, the old software is deleted before the new software is transferred.
- For any IP500v2 systems being upgraded, the Backup system files option will cause the system to backup its memory card files as part of the upgrade.
- For any IP500v2 system being upgraded, the Upload system files option will upload copies of the new system binaries to the units memory card.

- IP Office Manager 8.1+: The Restart IP Phones option can be used. This will cause those phone to load any upgrade phone firmware included in the system upgrade (if using the system's memory card as their firmware file source).

Search for Particular Systems

The default address used by the Upgrade Wizard is the address shown in the Manager title bar, which is selected through File | Preferences 108. If the unit required is not found, the address used can be changed.

1. Enter or select the required address in the Unit/Broadcast Address field.
2. Click Refresh to perform a new search.

Changing the .bin File Directory Used

The directory in which the Upgrade Wizard looks for .bin files is set through Manager's Binary Directory setting. This can be changed using Files | Change Working Directory 105 or File | Preferences | Directories 107. It can also be changed directly from the Upgrade Wizard as follows.

1. Right-click on the list area.
2. Select Select Directory.
3. Browse to and highlight the folder containing the .bin files. Click OK.
4. The list in the Available column will be updated to show the .bin files in the selected directory that match units or modules listed.

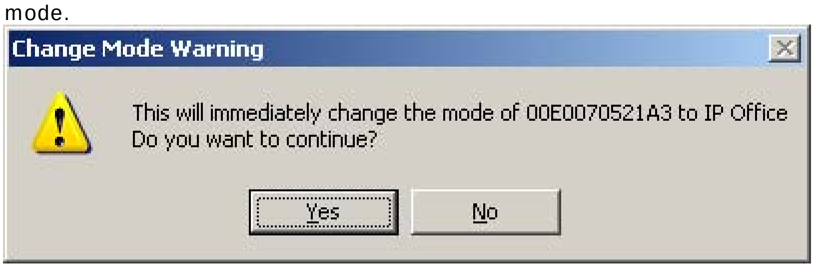
4.1.1.8.5 Change Mode

This command can be used to change the operating mode of an IP500v2 System SD card and thus of the system.

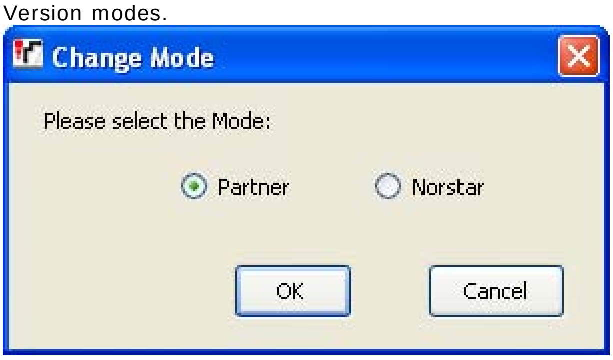
- Using this command will default the existing configuration. Therefore ensure that you have a backup copy of the configuration before using this command in case it is necessary to return to the previous mode.
- Do not use this command if the system includes components not supported by the mode to which you want to switch, the system may not restart correctly if that is the case. For example BRI cards are not supported in IP Office Essential Edition - PARTNER® Version mode.
- In order to use this command, the system security settings must first be defaulted. This can be done using the Erase Security Settings (Default) 119 command.
- Follow the command, the system is restarted.

The menu displayed after selecting a system depends on the current mode of that system:

- This menu is displayed for switching a non-IP Office Standard Version mode system to IP Office Standard Version mode.



- This menu is displayed for switching a IP Office Standard Version mode system to one of the non-IP Office Standard Version modes.



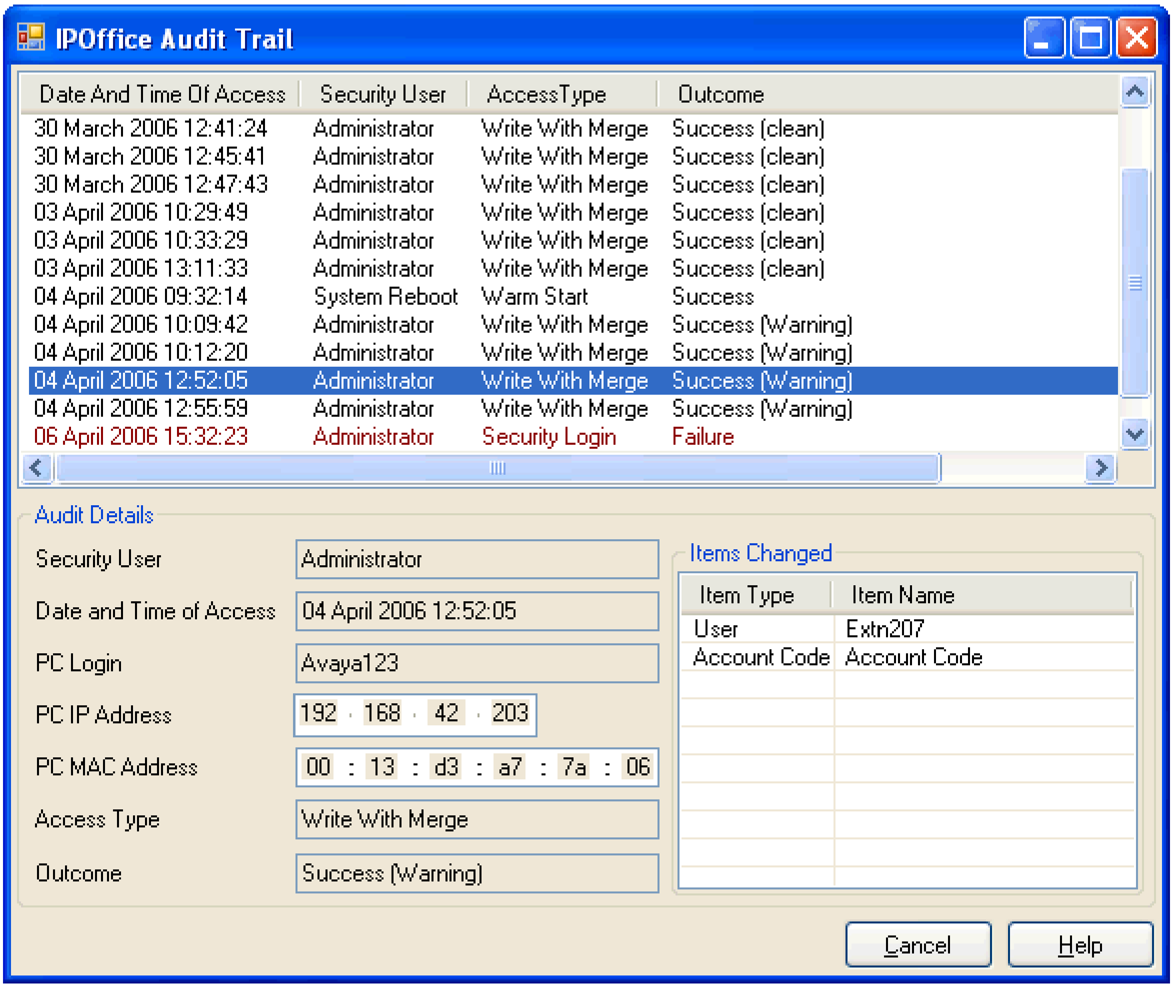
4.1.1.8.6 Audit Trail

The audit trail lists the last 16 actions performed on the system from which the configuration loaded into Manager was received. It includes actions by Service Users such as getting the configuration, sending a configuration back, reboots, upgrades and default the system. The audit trail is not available for systems running pre-3.2 Manager software.

IP Office 4.1+: Audit trail events can be output to a Syslog server through the system's **System | System Events** settings.

The last failed action is always recorded and shown in red. It is kept even if there have been 16 subsequent successful actions.

- The Audit Trail is part of the system configuration file received from the system. If the configuration is kept open between send and reboot operations (ie. if Close Configuration/Security Setting After Send 110 is not selected), the Audit Trail will not show details of those operations. It will only show details of those operations if the configuration is closed and then a new copy of the configuration is received from the system.



- **Audit Details**
  When a specific access event is selected from the list, the following information is shown in the Audit Details section:
  - The **Security User** shows the service user name used for the access action.
  - The **Data and Time of Access** indicate the local system time when the recorded event occurred.
  - The **PC Login** is the computer name of the PC used for the access.
  - The **PC IP Address** and **PC MAC Address** are the IP address and MAC address of the PC used for access.
  - The **Access Type** details the type of action that was performed.
  - The **Outcome** shows the system's response to the access. The outcome *Success (Warning)* refers to the sending of a configuration that contains fields marked as errors or warnings by Manager's validation function. *Success (Clean)* refers to the sending of a configuration that does not contain any validation errors or warnings.
- **Items Changed**
  The Items Changed area summarizes the changes contained in a sent configuration. Where changes to a single entry of a particular type are made, the Item Name field lists the individual entry changed. Where changes are made to several entries of the same type, the Item Name field displays Multiple items.

4.1.1.8.7 Security Settings

This command is used to switch the Manager application to security mode. In that mode, Manager is used to edit the security settings of a system (3.2 or higher only). Refer to the section Security Mode 68.

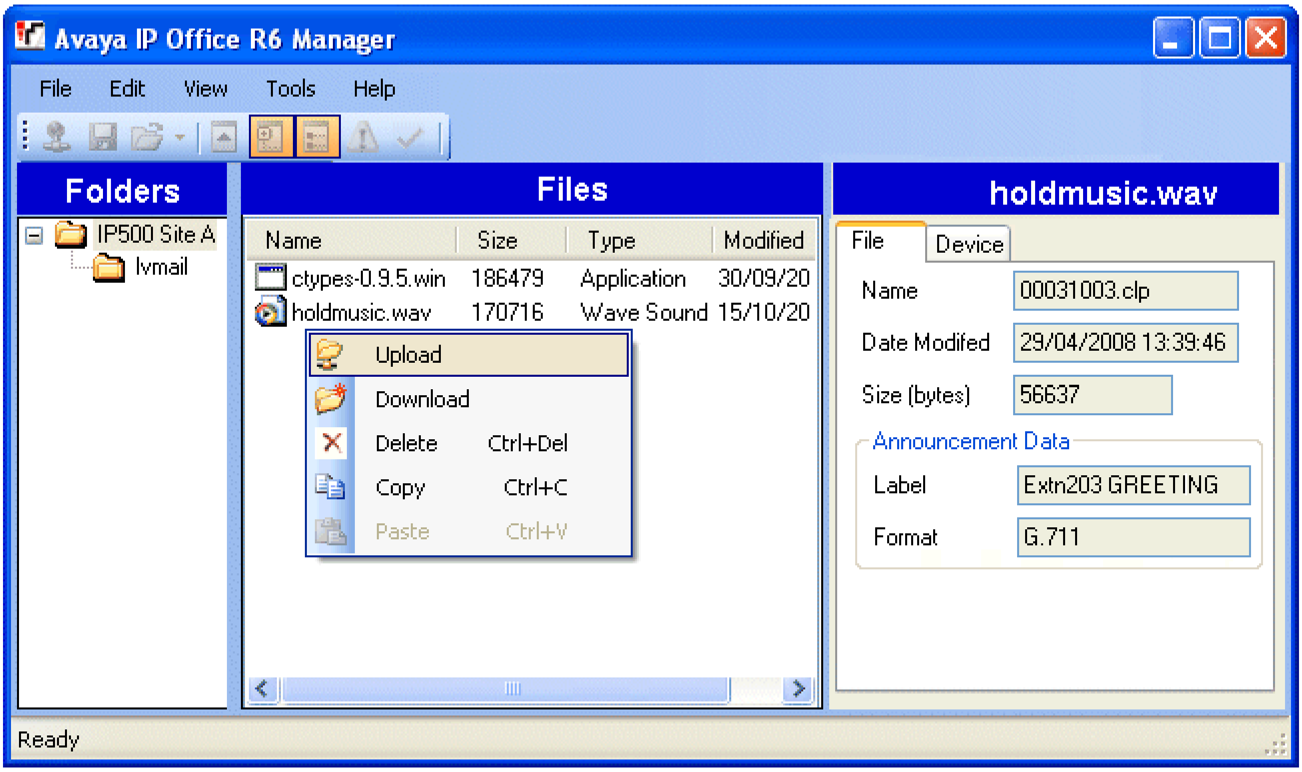
4.1.1.8.8 Erase Security Settings (Default)

IP Office 4.1+: This command returns the security settings of a system back to their default values. This action does not affect the system's configuration or audit trail record.

When this command is used, the Select IP Office menu is displayed. Once a system is selected, a valid user name and password are required to complete the action.

4.1.1.8.9 Embedded File Management

IP Office 4.2+: For control units (except Small Office Edition) with an memory card installed, the contents of the card can be viewed through Manager. This view can also be used to add and remove files from the card. This may be useful when the memory card is being used to store Music on Hold or IP phone firmware files.



- Embedded Voicemail Files
  When viewing the memory card, the files related to Embedded Voicemail are visible, however these files are greyed out (ie. cannot be deleted, downloaded or overwritten).
  - Mailbox greetings and messages are shown as *.clp* files.
  - The language prompts for Embedded Voicemail functions are stored in separate language sub-folders of lvmail . These are either *.c11* (IP500/IP500v2 and IP406 V2) or *.c23* (Small Office Edition) files.
  - Named prompt files for use by Embedded Voicemail auto attendants are stored in the lvmail\AAG folder and use the same *.c11* or *.c23* file formats as the language prompts. These files can be created from standard .wav files before being downloaded to the memory card by using the LVM Greeting Utility 124.
  - IP Office Release 5.0+: The Manager menu option File | Upload Voicemail Files will automatically select and transfer all Embedded Voicemail language prompts onto the card.
- Avaya IP Phone Files
  The memory card can be used as the source of files requested by IP Phones when rebooting. For phones using system DHCP, once the files are loaded onto the card, the TFTP Server IP Address 143 and HTTP Server IP Address 143 on the System | System 143 tab must be set to match the system's LAN address.
  - IP Office Release 5.0+: The Manager menu option File | Upload Phone Files will automatically select and transfer all DS phone, H323 phone and IP DECT binary files from the Manager folder to the embedded memory card.

* Viewing a Memory Card
  When Advanced | Embedded File Management is selected, the Manager will go through normal system discovery but will only allow selection of systems which can support a memory card. When a system is selected, a valid service user name and password for configuration access to that system is requested. If the system selected does not have a memory card installed, the files view remains blank and the message *TFTP:Received TFTP Error "Not Found"* appears in Manager's status bar.

* Changing the Files View
  The type of display used in the Files pane can be changed by selecting from the View menu in the toolbar.

* Adding Files
  Files can be added to the card by dragging and dropping or by right-clicking on the Files pane and selecting Upload or by using File | Upload File.... The system will ask for confirmation if the file already exists on the memory card. The progress of the file upload is then indicated.

* Deleting Files
  Existing files can be deleted by right-clicking on them and selecting Delete.

* Downloading Files
  Files can also be copied from the card by right-clicking on the file and selecting Download. Manager will prompt for the download location. Existing files are overwritten if present.

* To exit this mode of Manager back to normal configuration operation, select File | Configuration from the menu bar. Alternatively to view the card in another system, select File | Close File Settings and then File | Open File Settings.

4.1.1.8.10 Format IP Office SD Card

This command allows suitable SD cards to be formatted by the Manager PC. The system supports SD cards with the following format: SDHC minimum 4GB FAT32 format (Single partition, SDHC, class2+, FAT32, SPI & SD bus). Non-Avaya supplied cards of the same format can be used a system's **Optional** SD slot for additional actions such as backup.

- **! Warning**
  Do not re-purpose an Avaya Branch Gateway SD card for use with any other IP Office mode. Doing so may damage the SD card and make it unusable for your Avaya Branch Gateway system.

- **! WARNING: All File Will Be Erased**
  Note that this action will erase any existing files and folders on the card. If the requirement is just to update the card, use Recreate IP Office SD Card 122 without reformatting. Once a card has been formatted, the folders and files required for operation can be loaded onto the card from the Manager PC using the Recreate IP Office SD Card 122 command.

- **! WARNING:**
  Avaya supplied SD cards should not be formatted using any other method than the format commands within Manager and System Status Application. Formatting the cards using any other method will remove the feature key used for system licensing from the card.

1. Insert the SD card into a reader slot on the Manager computer.
2. Using Manager, select **File | Advanced | Format IP Office SD Card**.
3. Select the type of card. This selection just sets the card label shown when viewing the card details. It does not affect the actual formatting. Select the label that matches the file set you will be placing on the card.
   - **IP Office A-Law**
     A system fitted with this type of card will default to A-Law telephony. The system will default to IP Office Standard Version mode.
   - **IP Office U-Law**
     A system fitted with this type of card will default to U-Law telephony. The system will default to IP Office Standard Version mode.
   - **IP Office Partner Edition**
     A system fitted with this type of card will default to A-Law telephony and IP Office Essential Edition - PARTNER® Version mode operation.
   - **IP Office Norstar Edition**
     A system fitted with this type of card will default to U-Law telephony and IP Office Essential Edition - Norstar Version mode operation.
   - **Avaya Branch Gateway**
     Use this option for an SD card intended to be used with a system running in Avaya Branch Gateway mode. There is a separate SD card for Avaya Branch Gateway. The Avaya Branch Gateway SD card can only be used for Avaya Branch Gateway operation and cannot be used to change modes to IP Office. The label on the SD card includes BRANCH GW to indicate it is an Avaya Branch Gateway SD card. You also cannot use or change an IP Office SD card for use with an Avaya Branch Gateway system.
     - **! Warning**
       Do not re-purpose an Avaya Branch Gateway SD card for use with any other IP Office mode. Doing so may damage the SD card and make it unusable for your Avaya Branch Gateway system.
4. Browse to the card location and click OK.
5. The status bar at the bottom of Manager will display the progress of the formatting process.
6. When the formatting is complete, you can use the Recreate IP Office SD Card 122 command to load the system folders and files onto the card from the Manager PC.

- **A-Law or U-Law**
  PCM (Pulse Code Modulation) is a method for encoding voice as data. In telephony, two methods of PCM encoding are widely used, A-Law and U-Law (also called Mu-Law or µ-Law). Typically U-Law is used in North America and a few other locations while A-Law is used elsewhere. As well as setting the correct PCM encoding for the region, the A-Law or U-Law setting of a system when it is first started affects a wide range of regional defaults relating to line settings and other values.
  - For IP400 systems, each control units was manufactured as either an A-Law variant or a U-Law variant.
  - For IP500 and IP500v2 systems, the encoding default is set by the type of Feature Key installed when the system is first started. The cards are either specifically A-Law or U-Law. PARTNER Version cards are U-Law. Norstar Version cards are A-Law.

4.1.1.8.11 Recreate IP Office SD Card

This command can be used with a read-writeable SD card on the Manager PC. It copies the files and folders used by an IP500v2 system when starting. It updates the card with the version of those files installed with the Manager application. It includes the binary files for the system, external expansion modules and phones. It also includes the prompt files for embedded voicemail operation.

This process just replaces existing files and adds new files. It does not delete files, so for example, any existing embedded voicemail messages and greetings are retained. If the card contains dynamic system files such as SMDR records, they are temporarily backed up by Manager and then restored after the card is recreated.

For the card to be used in a system's System SD slot the card must be Avaya SD Feature Key card. The card must be correctly formatted (see Format IP Office SD card 121), however a reformat of an existing working card is not necessary before using recreate to update the card contents.

- The source for the files copied to the SD card are the sub-folders of the *\Memory Cards* folder under Manager's Working Directory 107 (normally *C:\Program Files Avaya\IP Office\Manager*). However, if the Working Directory is changed to a location without an appropriate set of *\Memory Cards* sub-folders, the required set of files will not be copied onto the SD card.

- ! Warning
  Do not re-purpose an Avaya Branch Gateway SD card for use with any other IP Office mode. Doing so may damage the SD card and make it unusable for your Avaya Branch Gateway system.

1. Note: This process can take up to 20 minutes depending on the PC. Once started, the process should not be interrupted.

2. Insert the SD card into a reader slot on the Manager computer.

3. Using Manager, select File | Advanced | Recreate IP Office SD Card.

4. Select the type of system for which the card is intended. This selection will affect how the systems operates when defaulted with this card present in its System SD card slot.

   - IP Office A-Law
     A system fitted with this type of card will default to A-Law telephony. The system will default to IP Office Standard Version mode.

   - IP Office U-Law
     A system fitted with this type of card will default to U-Law telephony. The system will default to IP Office Standard Version mode.

   - IP Office Partner Edition
     A system fitted with this type of card will default to A-Law telephony and IP Office Essential Edition - PARTNER® Version mode operation.

   - IP Office Norstar Edition
     A system fitted with this type of card will default to U-Law telephony and IP Office Essential Edition - Norstar Version mode operation.

   - Avaya Branch Gateway
     Use this option for an SD card intended to be used with a system running in Avaya Branch Gateway mode. There is a separate SD card for Avaya Branch Gateway. The Avaya Branch Gateway SD card can only be used for Avaya Branch Gateway operation and cannot be used to change modes to IP Office. The label on the SD card includes BRANCH GW to indicate it is an Avaya Branch Gateway SD card. You also cannot use or change an IP Office SD card for use with an Avaya Branch Gateway system.

     - ! Warning
       Do not re-purpose an Avaya Branch Gateway SD card for use with any other IP Office mode. Doing so may damage the SD card and make it unusable for your Avaya Branch Gateway system.

5. Browse to the card location and click OK.

6. Manager will start creating folders on the SD card and copying the required files into those folders.

7. Do not remove the card until the process is completed and Manager displays a message that the process has been completed.

- A-Law or U-Law
  PCM (Pulse Code Modulation) is a method for encoding voice as data. In telephony, two methods of PCM encoding are widely used, A-Law and U-Law (also called Mu-Law or µ-Law). Typically U-Law is used in North America and a few other locations while A-Law is used elsewhere. As well as setting the correct PCM encoding for the region, the A-Law or U-Law setting of a system when it is first started affects a wide range of regional defaults relating to line settings and other values.

  - For IP400 systems, each control units was manufactured as either an A-Law variant or a U-Law variant.

- For IP500 and IP500v2 systems, the encoding default is set by the type of Feature Key installed when the system is first started. The cards are either specifically A-Law or U-Law. PARTNER Version cards are U-Law. Norstar Version cards are A-Law.

4.1.1.8.12 LVM Greeting Utility

This command launches a utility that can be used to convert .wav files to the formats used by embedded voicemail (*.c23* for Small Office Edition and *.c11* for others). The source file must be in the standard format used for all system applications: PCM, 8kHz 16-bit, mono.

The resulting named greeting files can then be transferred to the embedded voicemail memory card and selected as auto attendant greetings. That is done using the Recording Name 402 field on the Auto Attendant | Auto Attendant 402 tab. The same named greeting file can be used in several auto attendants.

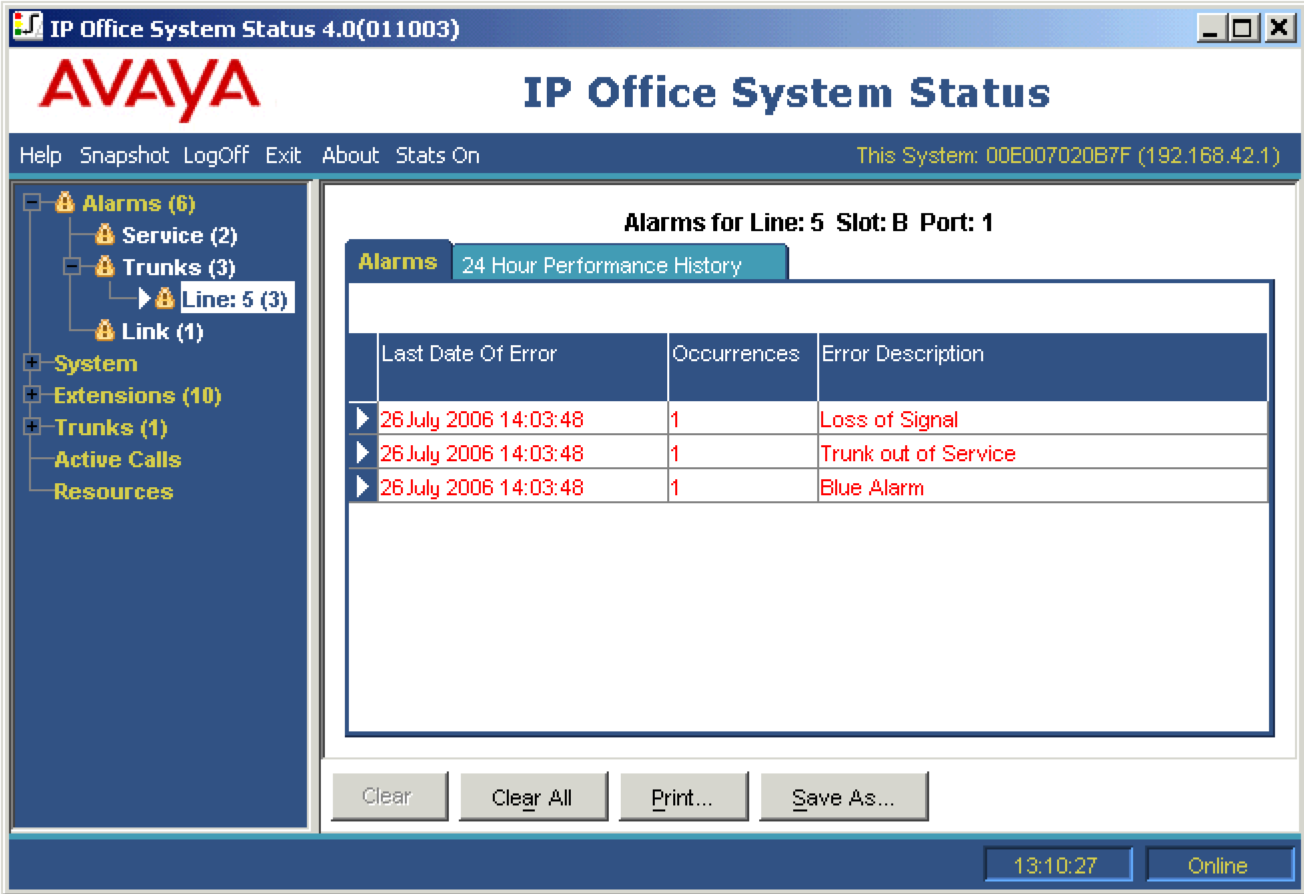
The utility can be run separately using the file LVMGreeting.exe found in the LVMGreeting sub-folder of the Manager application.

4.1.1.8.13 Voicemail Pro

If the Voicemail Pro client is installed on the same PC and Manager, this link can be used to launch the Voicemail Pro client. This can also be done by clicking on the icon in the Manager toolbar.

4.1.1.8.14 System Status

IP Office 4.0+: System Status is an application that can be used to monitor and report on the status of a system.



This is a separate application from Manager but if installed on the same PC, it can be started using the File | Advanced | System Status link within Manager. Use of the application requires a service user name and password configured on the system for System Status Access within the system's security settings.

4.1.1.8.15 Memory Card Command

These commands are used with the memory cards installed in IP500/IP500v2 control units.

4.1.1.8.15.1 Shutdown

This command can be used to shutdown operation of IP500/IP500v2 memory cards.

This action or a system shutdown 114 must be performed before a memory card is removed from the unit. Removing a memory card while the system is running may cause file corruption.

Shutting down the memory card will disable all services provided by the card including embedded voicemail. For IP500v2 systems, features licensed by the memory card will continue to operate for up to 2 hours.

Card services can be restarted by either reinserting the card or using the Start Up 125 command.

4.1.1.8.15.2 Start Up

This command can be used to restart operation of an IP500/IP500v2 memory card that has been shut down 125.

The command will start the Select IP Office discovery process for selection of the system.

4.1.1.9 Backup/Restore

4.1.1.9.1 Backup Binaries and Configuration

This command copies of all configuration files (.cfg) and software files (.bin) stored in Manager's working directory to a selected folder.

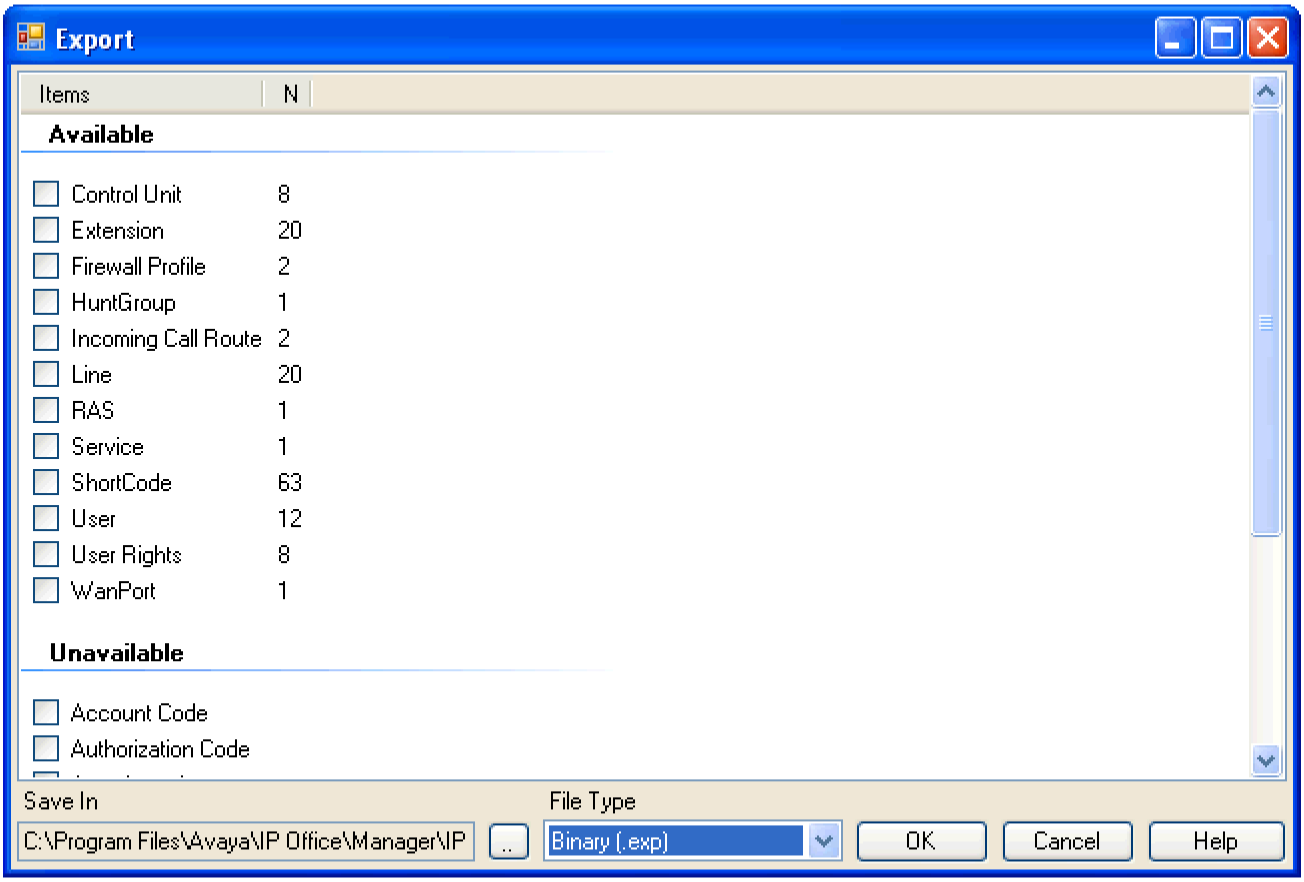
4.1.1.9.2 Restore Binaries and Configuration

This command copies all configuration files (.cfg) and software files (.bin) stored in a selected folder to the Manager's working directory.

4.1.1.10 Import/Export

4.1.1.10.1 Export

This command allows you to export the selected parts of the configuration to either a set of CSV text files (.csv) or a single binary file (.exp). See Importing and Exporting Settings 61.

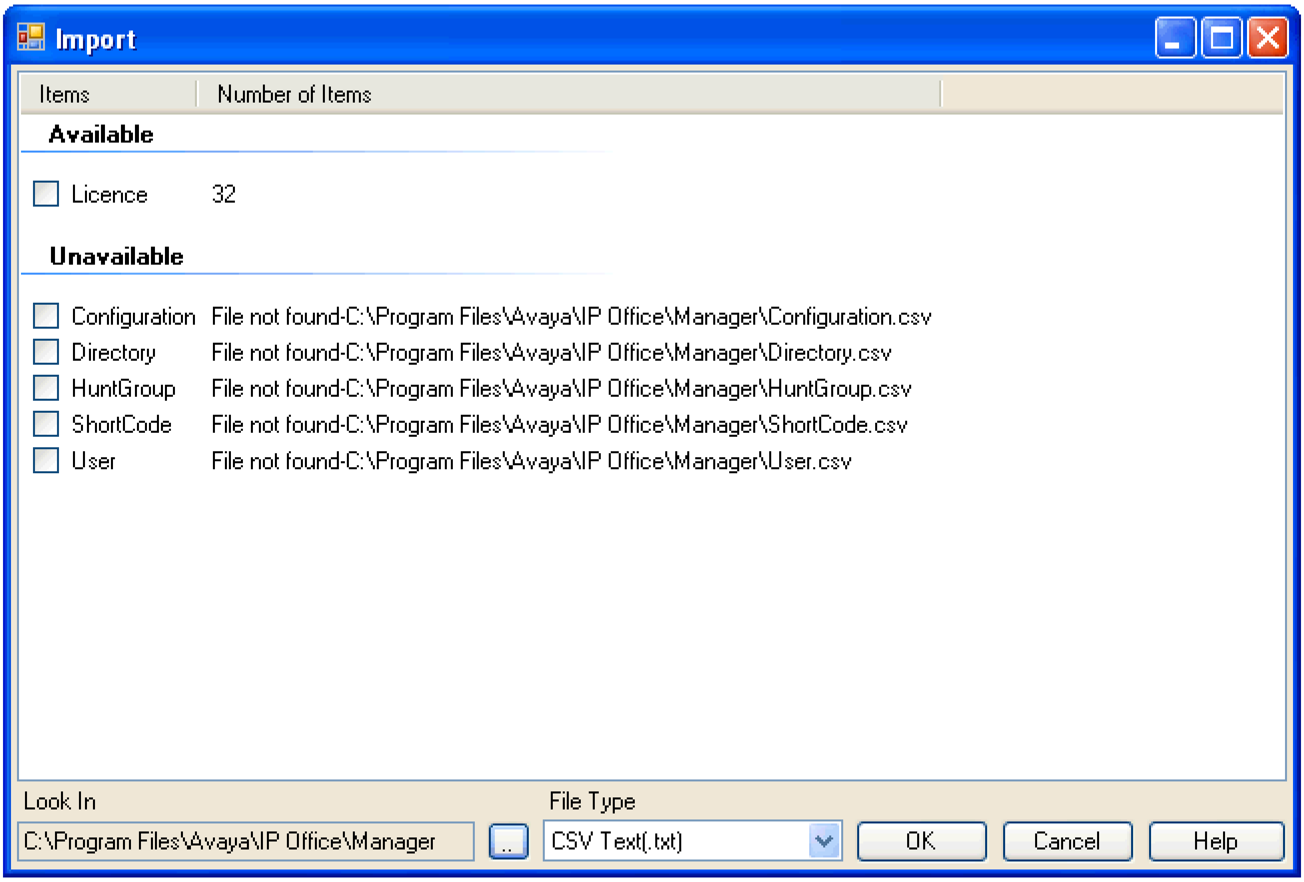


The display shows those exportable entry types for which the configuration contains entries. The File Type and the Save In path can be selected at the base. The default location used is sub-directory of the Manager application directory based on system name of the currently loaded system.

Manager imports and exports CSV files using UTF8 character encoding which uses a double byte to support characters with diacritic marks such as ä. Other applications such as Excel, depending on the user PC settings, may use different single-byte encoding which will cause such characters to be removed. Care should be taken to ensure that any tool used to create or edit a CSV supports all the characters expected and is compatible with UTF8.

4.1.1.10.2 Import

This command allows you to import configuration settings. Two formats are supported. Binary files (.exp) are settings previously exported from a system using File | Import /Export | Export 126. CSV text files (.csv) can also be exported from a system or can be created using a plain text editor. See Importing and Exporting Settings 61.



For the selected File Type and the Look In path, the window displays the file or files found. The default location used is sub-directory of the Manager application directory based on system name of the currently loaded system.

Manager imports and exports CSV files using UTF8 character encoding which uses a double byte to support characters with diacritic marks such as ä. Other applications such as Excel, depending on the user PC settings, may use different single-byte encoding which will cause such characters to be removed. Care should be taken to ensure that any tool used to create or edit a CSV supports all the characters expected and is compatible with UTF8.

4.1.1.11 Exit

The File | Exit command exits the Manager application.

4.1.2 View

4.1.2.1 Toolbars

This command allows selection of which toolbars should be shown or hidden in configuration mode. A tick mark is displayed next to the name of those toolbars that are currently shown.

4.1.2.2 Tooltip

This setting control whether additional tooltips are displayed when Manager is running in Partner Version mode.

4.1.2.3 Navigation Pane

This command shows or hides the Navigation Pane. A tick mark appears next to the command when the pane is shown.

4.1.2.4 Group Pane

This command shows or hides the Group Pane. A tick mark appears next to the command when the pane is shown.

4.1.2.5 Details Pane

This command set the location of the Details Pane when the Group Pane is also shown. The Details Pane can be placed either below or to the right of the Group Pane.

4.1.2.6 Error Pane

This command shows or hides the Error Pane. A tick mark appears next to the command when the pane is shown.

4.1.2.7 Advance View

This command causes Manager to switch from its simplified view to advanced view mode. Manager automatically switches to advanced view mode if an IP Office Standard Version configuration is loaded.

4.1.2.8 Hide Admin Tasks

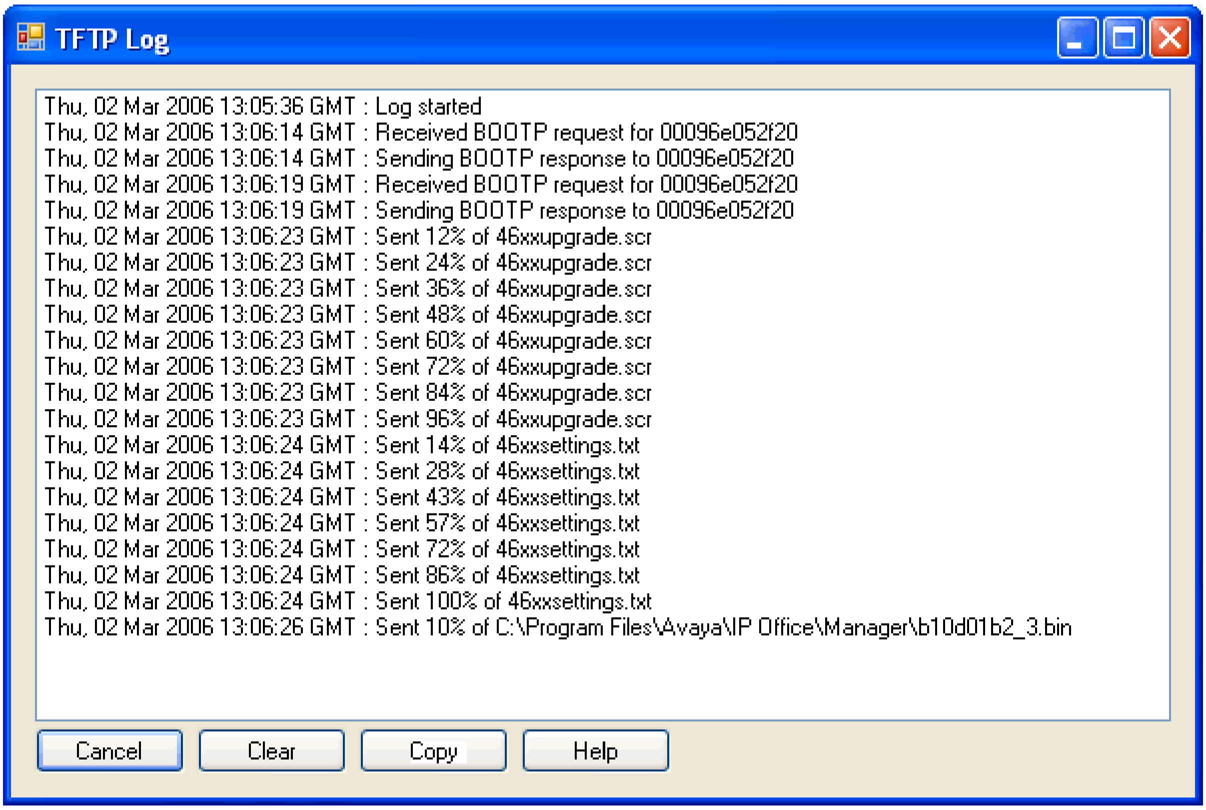
This settings shows or hides the Admin Tasks List available when Manager has a IP Office Essential Edition - PARTNER® Version or IP Office Essential Edition - Norstar Version configuration loaded.

4.1.2.9 Simplified View

If Manager has no configuration loaded, this command switches it from advanced view to simplified view.

4.1.2.10 TFTP Log

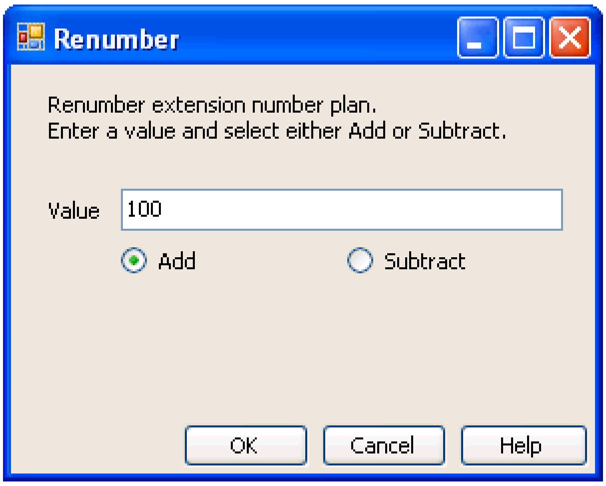
This command displays the TFTP Log window. This window shows TFTP traffic between Manager and devices that uses TFTP to send and receive files. For example, the TFTP Log below shows an Avaya IP phone requesting and then being sent its software files.



4.1.3 Tools Menu

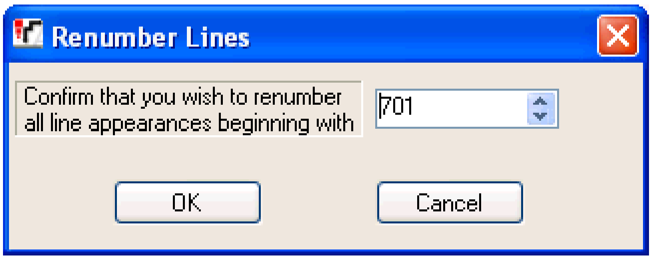
4.1.3.1 Extension Renumber

This command allows the extension numbering of user extensions to be raised or lowered by a specified amount. The command does not alter the extension number used for hunt groups but does adjust the extension numbers of hunt group members.



4.1.3.2 Line Renumber

On external trunks Line appearance ID numbers can be assigned to each channel supported in order to allow that channel or line to be associated with a Line Appearance button on phones that support button programming. By default all lines are automatically numbered from 701 upwards when added to the system. This command allows the lines to be renumbered from a different starting point.



4.1.3.3 Connect To

This option can be used to create Small Community Network connections between two systems, one being the system with its configuration currently loaded in Manager, the other being selected from a discovery dialog.

- ! IMPORTANT
  This process will require the systems to be rebooted.

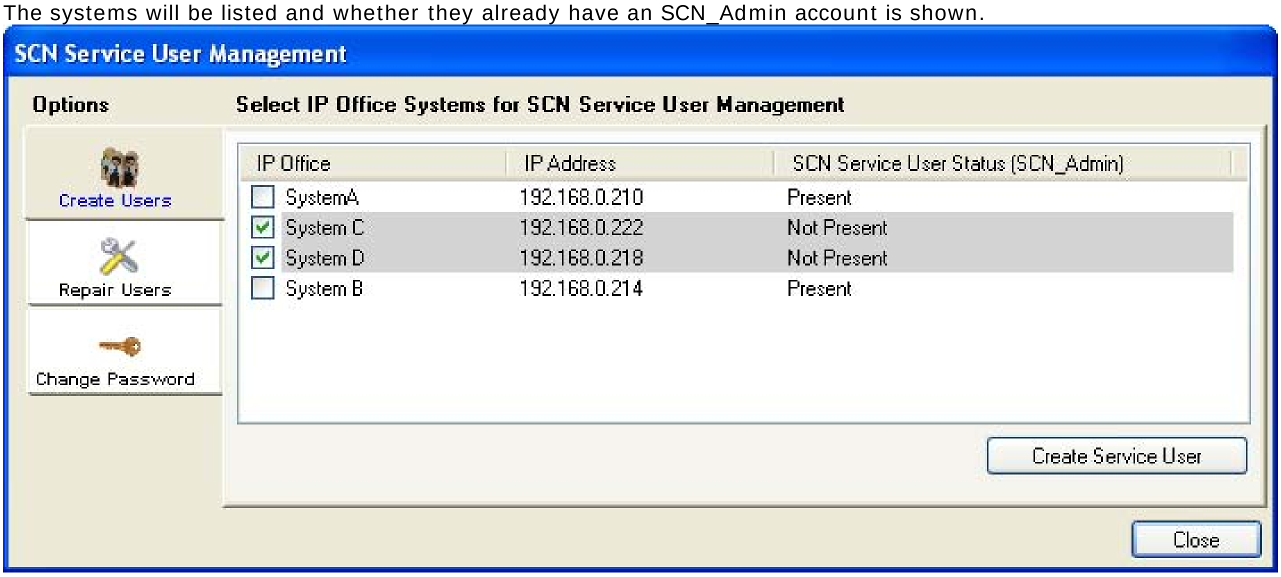
1. With the configuration of the first system received from that system and displayed in Manager, clicking on or Tools | Connect To
2. A discovery menu is displayed and will list any other systems discovered.
3. Select the system to which an SCN connection is required.
4. Enter the login name and password for configuration access to that system.
5. Manager will switch to SCN management 817 mode, displaying the configuration of both systems.
6. Click to save the new configuration back to each system.

4.1.3.4 SCN Service User Management

When managing multiple systems, it may be useful to create a common user name and password on all the systems for configuration access. This tool can be used to create a new service user account, SCN_Admin, for configuration access.

This process requires you to have a user name and password for security configuration access to each of the systems.

1. Select Tools | SCN Service User Management.
    - The option is not shown if a IP Office Essential Edition - PARTNER® Version or IP Office Essential Edition - Norstar Version system configuration is loaded. If no configuration is load, and the option is not shown, select View | Advanced View.
2. The Select IP Office menu will display the list of discoverable systems.
3. Select the systems for which you want to create a common configuration account. Click OK.
4. A user name and password for security configuration access to each system is request. Enter the values and click OK. If the same values can be used for all systems enter those values, select Use above credentials for all remaining, selected IPOs. If each system requires a different security user names and password, deselect Use above credentials for all remaining, selected IPOs.
5. The systems will be listed and whether they already have an SCN_Admin account is shown.



6. To create the SCN_Admin account on each system and set the password for those account click on Create Service User.

7. Enter the common password and click OK.
8. The password can be changed in future using the Change Password option.
9. Click Close.

4.1.3.5 Busy on Held Validation

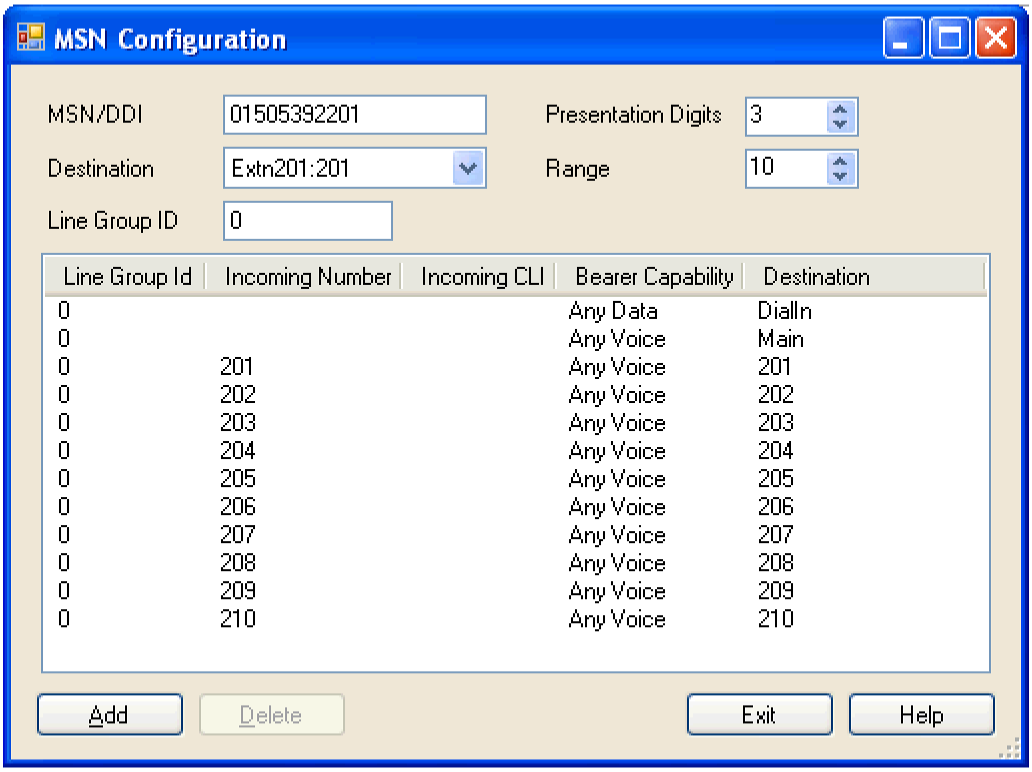
Busy on Held is a user feature where, when the user has a call on hold, the system indicate the user as being busy to any further calls.

The use of Busy on Held in conjunction with multiple call appearance buttons is deprecated. This command can be used to identify those users who have multiple call appearance buttons and for whom Busy on Held is currently set.

When run, it shows a list of the users affected and if selected their Busy on Held setting will be switched off.

4.1.3.6 MSN/DID Configuration

This menu can be used to populate the Incoming Call Route table with a range of MSN or DID numbers.



In the example above, the customer has ten DID numbers starting at 01505392201 with the central office exchange passing through 3 digits for each. Having selected the number of presentation digits (3), set the range (10) and selected the first destination (201); clicking Add created the ten incoming call routes (201 to 210).

- **MSN/DID**
  The first number in the set of MSN numbers for which you have subscribed. Note: If you require to find an exact match between the MSN numbers and the destination numbers, enter a minus (-) sign before the first MSN number.

- **Destination**
  Where incoming calls with matching digits should be routed. The drop-down list contains the extensions and groups on the system.

- **Line Group ID**
  Specifies the incoming line group ID of the trunks to which the DID routing is applied.

- **Presentation Digits**
  Set to match the number of digits from the MSN/DID number that the central office exchange will actually present to the system.

- **Range**
  How many MSN or DID number routes to create in sequence using the selected MSN/DID and Destination as start points. Only routing to user extensions is supported when creating a range of entries.

- **Add**
  Adds the appropriate entries to the Incoming Call Route table using the value entered above.

- **Delete**
  Removes a specific entry.

4.1.3.7 Print Button Labels

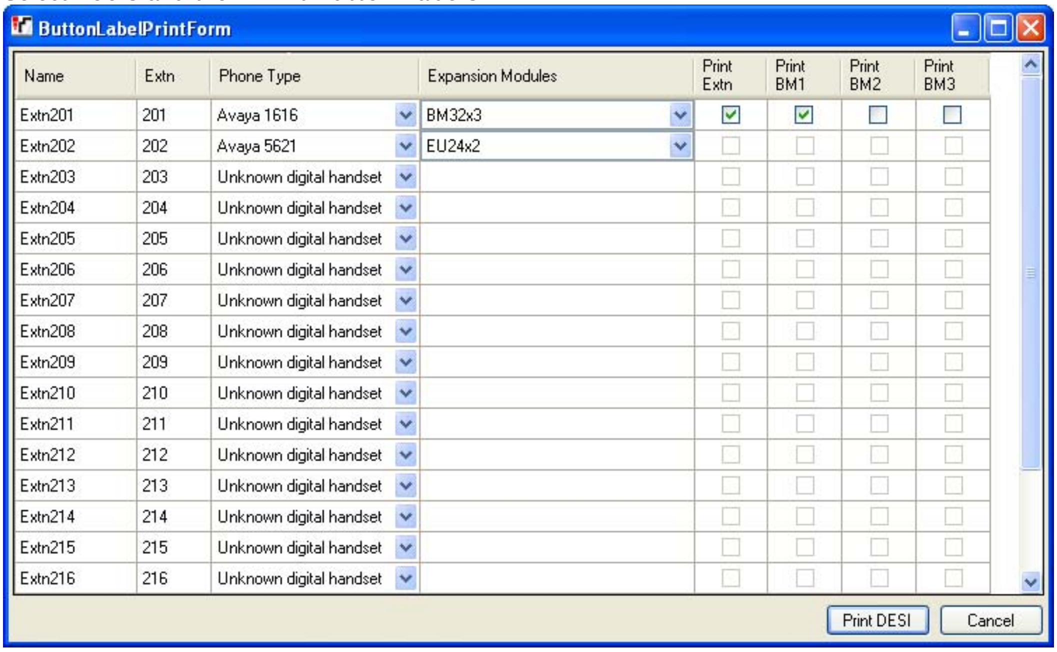
This option is only enabled if a version of DESI software is also installed on the same PC as Manager. It can then be used when a system configuration is loaded in Manager.

DESI software can be obtained from the Avaya support web site (http://support.avaya.com) or from DESI (http//www.desi.com). Currently, though all users are shown, only ETR, 1400 and 1600 phones are supported by DESI templates.

The text used on the labels:

- If a text label has been added in the user's Button Programming settings, that text label is passed to the DESI application.
- Note that the DESI application cannot import non-ASCII characters and may render them incorrectly.
- Manager will display a warning if it estimates that the user's current text for some buttons may exceed the label space of the phone type.
- If no text label has been set, the default label for the action currently assigned to the button is passed to the DESI application.
- Once the labels are shown in the DESI application, the label text can be changed.

1. Load the configuration of the system for which you want to print button labels.
2. Select **Tools** and then **Print Button Labels**.



- **Name / Extn**
  These are the user name and extension number details of the users in the system configuration currently loaded in Manager.
- **Phone Type**
  This field shows the type of phone, if known, that the user is currently associated with. The drop down can be used to change the selection if required.
- **Expansion Modules**
  If the phone type supports additional button modules, this drop down can be used to select the type and number of button modules.
- **Print Extn**
  This check box is used to select whether the phone button details should be included in the output passed to the DESI software. These button will only be selectable if the user's current **Phone Type** is set to an ETR, 1400 and 1600 Series phone.
- **Print BM1 / Print BM2 / Print BM3**
  These check boxes are used to select whether button module button details should be included in the output passed to the DESI software. These button will only be selectable if the user's **Expansion Modules** is set to the number of button modules.

3. Click **Print DESI** to transfer the information to the DESI application. Within DESI, edit the labels as required and then print the labels.

4.2 Security Mode

These commands are available when the Manager is in security configuration mode.

4.2.1 Open Security Settings

This command displays the Select IP Office menu to select and load a system's security settings. This requires entry of a user name and password with rights to access security settings of the selected system.

This behavior changes when configuration settings have already be received from a system using a Service User name and password that also has security access rights for that system. In that case, the system's security settings are automatically loaded without requiring name and password entry.

4.2.2 Close Security Settings

Close the currently open set of security settings received from a system without saving those settings.

4.2.3 Save Security Settings

Send edited security settings back to the system. Requires re-entry of a Service User name and password with access rights for security settings.

4.2.4 Reset Security Settings

IP Office 4.1+: Reset the security settings of the selected syste, to defaults. Requires entry of a Service User name and password with access rights for resetting the security settings. This option is not useable while a set of security configuration settings is loaded.

The command File | Advanced | Erase Security Settings (Default) 119 performs the same action from Manager configuration mode.

4.2.5 Preferences

See Preferences 106 in the Menu Bar Commands | Configuration Mode | File Menu section.

4.2.6 Exit

This command closes Manager.

4.2.7 Configuration

This command returns the Manager application to configuration mode.

4.3 Embedded File Management

These commands are available when the Manager is in embedded file management mode.

4.3.1 Open File Settings

Select a system and display the contents of its memory cards if any are present and in use.

4.3.2 Close File Settings

Close the current memory card contents listing without exiting embedded file management mode.

4.3.3 Refresh File Settings

This command can be used to request a file update from the system.

4.3.4 Upload File

This command can be used to select and upload a file to the memory card in the system.

4.3.5 Upload System Files

This command is available with IP500v2 systems. When this command is selected, Manager will upload the software files for operation to the System SD card. It includes all Manager software, phone software and embedded voicemail prompts not already present on the System SD card.

- ! WARNING
  After this command is completed, the system is rebooted. This will end all calls and services in progress.

4.3.6 Backup System Files

This command is available with IP500v2 systems. When selected, Manager copies the folders and files from the System SD card's */primary* folder to its */backup* folder. Any matching files and folders already present are overwritten. This action can be included as part of the system's automatic daily backup process (System | System | Automatic Backup Command 143).

4.3.7 Restore System Files

This command is available with IP500v2 systems.

When selected, Manager copies the folders and files from the System SD card's */backup* folder to its */primary* folder. Any matching files and folders already present are overwritten.

- ! WARNING
  After this command is completed, the system is rebooted. This will end all calls and services in progress.

4.3.8 Upgrade Binaries

This command is available for IP500v2 systems that have a system SD card and Option SD card installed.

When this command is selected, all files except *config.cfg* and *keys.txt* files in the Optional SD card's *\primary* folder are copied to the System SD card.

- ! WARNING
  After this command is completed, the system is rebooted. This will end all calls and services in progress.

4.3.9 Upgrade Configuration

This command is available for IP500v2 systems that have a system SD card and Option SD card installed.

When this command is selected, any *config.cfg* and *keys.txt* files in the Optional SD card's *\primary* folder are copied to the System SD card.

- ! WARNING
  After this command is completed, the system is rebooted. This will end all calls and services in progress.

4.3.10 Upload Voicemail Files

This command is useable with IP500 and IP406 V2 control units. When this command is selected, Manager copies the prompts necessary for embedded voicemail to the memory card.

For IP500v2 control units, use Upload System Files 135.

4.3.11 Upload Phone Files

This command is available for IP500 and IP406 V2 control units. When this command is selected, Manager copies the software files relating to phone firmware to the memory card.

For IP500v2 control units, use Upload System Files 135.

4.3.12 Copy System Card

This command is available for systems that have a system SD card and Option SD card. When this command is selected, the system will copy the folders and files on its System SD card to the Optional SD card. Any matching files and folders already present on the Optional SD card are overwritten.

This process takes at least 90 minutes and can take longer.

4.3.13 Configuration

This command will exit Embedded File Management and return Manager to configuration editing mode.

Chapter 5.
Configuration Settings

5. Configuration Settings

This following sections detail the various configuration settings provided for different entry types within the system configuration. Depending on the type and locale of the system some settings and tabs may be hidden as they are not applicable. Other settings may be grayed out. This indicates that the setting is either for information only or that another setting needs to be enabled first.

The different entry types are:

- BOOTP 140
  Manager settings for the support of systems requesting software.
- Operator 141
  Manager settings for the editing of configuration settings from pre-IP Office 3.2 systems.
- System 142
  Overall settings for the data and telephony operation of the system.
- Line 185
  Settings for trunks and trunk channels within the system including VoIP trunks.
- Control Unit 250
  Information summary of the control and expansion units in the system.
- Extension 251
  Settings for extension ports including IP extensions.
- User 262
  Settings for system users. They may or may not be associated with an extension.
- Hunt Group 304
  Collections of users to which calls can be directed for answer by any one of those users.
- Short Code 330
  These are numbers which when dialed trigger specific features or are translated for external dialing.
- Service 331
  Configuration settings such as user names and passwords needed for connections to data services such as the Internet.
- RAS 340
  Remote Access Service settings for connecting incoming data calls.
- Incoming Call Route 344
  Entries here are used to match incoming call details to destinations on the system.
- WAN Port 351
  Configuration settings for the WAN ports provided on some units.
- Directory 358
  External names and numbers. Used for matching names to incoming calls and for dialing from user applications.
- Time Profile 360
  Used to control when various functions are active.
- Firewall Profile 363
  Use to control the types of data traffic that can cross into or out of the system.
- IP Route 368
  These entries are used to determine where data traffic on the system LAN and WAN interfaces should be routed.

- Least Cost Routing 370
  On pre-4.0 systems LCR is used to rerouted or block outgoing external calls. For IP Office 4.0+ this has been replace by ARS.

- Account Code 375
  Used for call logging and to control the dialing of certain numbers.

- License 378
  License keys are used to enable system features and applications.

- Tunnel 380
  Used to created IPSec and L2TP data tunnels.

- Logical LAN 385
  Used to allow two subnets to be run and routed on the same physical LAN.

- Wireless 386
  Used to provide wireless services on the Small Office Edition.

- User Rights 389
  Provide templates to control which settings users can change.

- Auto Attendant 400
  Used when an Avaya memory card is installed in the control unit (Small Office Edition, IP406 V2, IP500 or IP500v2 system).

- ARS 407
  Automatic Route Selection is used by to control outgoing external calls.

- Authorization Codes 404
  Authorization codes are similar to account codes. However, unlike account codes which are useable by any user, each authorization code is only useable by a specific user or users associated with a specific set of user rights.

- E911 System 421
  Available of US systems to support E911 services.

5.1 BOOTP

BOOTP is protocol used by devices to request software when restarting. It is used when upgrading the control unit within a system or when the core software within the control unit has been erased. When running, Manager can respond to BOOTP requests and, if it finds a matching BOOTP entry for the system, provide the software file indicated by that entry.

BOOTP entries are not part of a system's configuration settings; instead they are saved on the Manager PC. Normally Manager automatically creates a BOOTP entry for each system with which it has communicated, up to a maximum of 50 entries. However BOOTP entries can be added and edited manually when necessary.

- **File Location**
  The location from which Manager provides files in response to BOOTP is its binaries directory. This can be changed using File | Change Working Directory [105] or File | Preferences | Directories [107]. This directory is also the directory used by Manager when providing files by TFTP.

- **Disabling BOOTP**
  Manager can be disabled from providing BOOTP support for any systems. Select File | Preferences | Preferences | Enable BOOTP and TFTP Server [106].

Settings

- **Enabled**: *Default = Enabled*
  If unticked, BOOTP support for the matching system from this Manager PC is disabled.

- **System Name**
  This field is not changeable. It shows the system name.

- **MAC Address**
  The MAC address of the system's control unit. The address can be obtained and or verified in a number of ways:

  - When a system's configuration settings are loaded into Manager, it is shown as the Serial Number on the Unit [250] form. On defaulted systems, it is also used as the system name.

  - If the system is requesting software, the MAC address is shown as part of the request in the status bar at the base of the Manager screen.

  - If the system can be pinged, it may be possible to obtain its MAC address using the command *arp -a <ip address>*.

- **IP Address**
  The IP address of the system's LAN1.

- **Filename**
  The name of the .bin software file used by that type of control unit. For full details refer to the Manager Installation Manual. To be transferred to the system this file must exist in the Manager applications Working Directory.

- **Time Offset**: *Default = 0.*
  In addition to performing BOOTP support for systems the Manager application can also act as a time server (RFC868). This field sets the offset between the time on the PC running Manager and the time sent to the system in response to its time requests. The field is not used if a specific Time Server IP Address is set through the System form in the system's configuration settings.

  - Manager can be disabled from acting as an Internet Time (RFC868) server. Select File | Preferences | Preferences [106] and untick Enable time server.

5.2 Operator

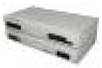
Operator entries are not part of a system's configuration settings. They are used when a pre-3.2 IP Office configuration is loaded to control what parts of a configuration can be edited.

The table below lists the settings for the default operators provided.

Operator	View	Edit	New	Delete	Configuration Entry Types
Administrator	✓	✓	✓	✓	All configuration entries.
Manager	✓	✓	✓	✓	View all. Other actions Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, Directory, Time Profile, Firewall Profile, IP Route, Least Cost Route, Account Code, ARS.
Operator	✓	✓	✗	✗	View all configuration entries. Edit all except System, Line, Control Unit and Authorization Codes.

If when receiving a configuration from a pre-3.2 system an invalid operator is specified, the settings will be loaded using the Guest operator. This additional operator allows a read-only view.

5.3 System



There is only one System entry for each system. When working in SCN Management 817 mode, clicking on the System icon for a particular system displays a system inventory 825 page for that system.

The following tabs are part of the System form:

- System 143
  General settings for the system.
- LAN1 148
  Network settings for the RJ45 Ethernet LAN port on the control unit. Includes DHCP and RIP settings.
- LAN2 155
  Network settings for the RJ45 Ethernet WAN port on the control unit. LAN2 is not supported by all control units.
  - On the Small Office Edition, IP500 and IP500v2 control units the LAN2 settings are used for the RJ45 Ethernet port labeled WAN.
  - On the IP412 control units LAN2 settings are used for the RJ45 ethernet port labeled LAN2.
  - For IP406 V2 control units running IP Office 4.1+, the RJ45 LAN port 8 can be selected to act be LAN2 if required.
- DNS 156
  Specify the Domain Name Server addresses to use for address resolution.
- Voicemail 157
  Details the type and location of the voicemail server.
- Telephony 160
  System-wide telephony settings.
- H.323 Gatekeeper
  Settings used for VoIP end points registering with the system and for DiffServ QoS settings applied to VoIP traffic. This tab is only shown on pre-4.0 systems. For IP Office 4.0+, separate LAN1 and LAN2 VoIP settings are shown on the LAN1 and LAN2 tabs respectively.
- LDAP 169
  Settings to allow the system to include Lightweight Directory Access Protocol database records in its directory.
- System Events 172
  Simple Network Management Protocol (SNMP), email (SMTP), and Syslog settings for the sending of system events.
- Twinning 177
  System wide controls for the use of Mobile Twinning.
- SMDR 179
  Settings for the sending of call records to a specified IP address.
- VCM 181
  This form allows adjustment of the echo control applied by VCM cards.
- CCR 184
  This form is used for settings specific to the Customer Call Reporter (CCR) application.

5.3.1 System

System | System	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓ Changes to Locale, License Server IP Address and Favor RIP Routes over Static Routes require a reboot..

- Name: *Default: = System MAC Address.*
  A name to identify this system. This is typically used to identify the configuration by the location or customer's company name. Some features such as Gatekeeper require the system to have a name. This field is case sensitive and within any network of systems must be unique. Do not use punctuation characters such as #, ?, /, -,. and ,.

- Contact Information: *Default = Blank. Software level = 4.1+.*
  This field is only be edited by service user with administrator rights. If Contact Information is entered, it will set the system under 'special control'.

  - If the contact information is set using a standalone version of Manager, warnings that "This configuration is under special control" are given when the configuration is opened again. This can be used to warn other users of Manager that the system is being monitored for some specific reason and provide them with contact details of the person doing that monitoring. See Loading a Configuration 55.

- Locale
  This setting sets default telephony and language settings based on the selection. It also sets various external line settings and so must be set correctly to ensure correct operation of the system. See Supported Country and Locale Settings 844. For individual users the system settings can be overridden through their own locale setting (User | User | Locale).

  - When the system routes a call to the voicemail server it indicates the locale for which matching prompts should be provided if available. The locale sent to the voicemail server by the system is determined as follows:

Locale Source	Usage
Short Code	The short code locale, if set, is used if the call is routed to voicemail using the short code.
System	If no user or incoming call route locale is set system locale is used unless overridden by a short code locale.
Incoming Call Route	The incoming call route locale, if set, is used if caller is external.
User	The user locale, if set, is used if the caller is internal.

  - Customize Locale Settings: *Software level = IP Office 4.0 Q2 2007 Maintenance Release +*
    The Customize locale matches the Saudi Arabia 885 locale but with the following additional controls:

    - Tone Plan: *Default = Tone Plan 1*
      The tone plan control tones and ringing patterns. The options are:

      - *Tone Plan 1:* United States 895.
      - *Tone Plan 2:* United Kingdom 894.
      - *Tone Plan 3:* France 863.
      - *Tone Plan 4:* Germany 865.
      - *Tone Plan 5:* Spain 888.

    - CLI Type: *Default = FSK V23*
      This should match the method CLI signalling being received by the system.

    - Busy Tone Detection: *Default = Off.*
      Enables or disables the use of busy tone detection for call clearing. This is a system wide setting.

- Password: *Default = password. Software level = 2.1 to 3.1.*
  A password for controlling access to the operation of the Control Unit. This is required to upgrade and reboot and to send or receive configurations from the Control Unit. This is a required option and a prompt is given if left blank. For IP Office 3.2+ systems this setting has become part of the security settings.

- Monitor Password: *Default = blank. Software level = 2.1 to 3.1.*
  This password is used by the Monitor and Call Status applications to allow communication with the main unit. If left blank these applications will use the System Password above. For IP Office 3.2+ systems, this setting has become part of the security settings.

* Time Offset: *Default = 00:00. Software level = 1.0 to 6.0.*
This setting can be used if the system is in a different time zone from its time server. For example, if the system is 5 hours behind the time server, this field should be configured with -5:00 to make the adjustment. The time offset can be adjusted in 15 minute increments. Note: If the time server is a Manager PC, the adjustment can also be done through the Manager BOOTP entry for the system.

* TFTP Server IP Address: *Default = 0.0.0.0 (Broadcast).*
For Avaya IP phones using DHCP, the address set here is used for their TFTP requests for software and settings files.

  * While running, Manager can act as a TFTP server and provides files from its configured binaries directory 107. Manager can be disabled from doing TFTP through the Enable BootP and TFTP Servers 106 command.

  * On systems with an Avaya memory card (Small Office Edition, IP406 V2, IP500 and IP500v2), the LAN1 IP Address can be entered to specify that memory card as the TFTP file source. The files required for download must be transferred onto the card using either TFTP from the PC command line (see File Writer IP Address below) or through Embedded File Management 119.

  * 1100, 1200, 1600 and 9600 Series IP phones do not support TFTP and require an HTTP Server IP Address (see below) to be specified.

  * IP Office Release 5.0+: This address is used in DHCP responses if the Phone File Server Type is set to *Custom*.

* HTTP Server IP Address: *Default = 0.0.0.0 (Disabled). Software level = 4.2+.*
If an address is entered here, Avaya IP phones using DHCP will use that address to request software and settings files. The files for download should be placed in the HTTP server's root directory.

  * IP Office 4.2: Using the Embedded Voicemail memory card is also supported for HTTP file. This is done by setting the TFTP Server IP Address and HTTP Server IP Address to match the control unit IP address. This is supported for up to 50 IP phones total.

  * IP Office 4.2 Q4 2008 maintenance release: HTTP-TFTP Relay is support using Manager as the TFTP server. This is done by setting the TFTP Server IP Address to the address of the Manager PC and the HTTP Server IP Address to the control unit IP address. This method is supported for up to 5 IP phones total.

  * IP Office Release 5.0+: This address is used in DHCP responses if the Phone File Server Type is set to *Custom*.

* Phone File Server Type: *Default = Custom. Software level = 5.0+.*
For IP (H323 and SIP) phones using the system as their DHCP server, the DHCP response can include the address a file server from which the phone should request files. The setting of this field control which address in used in the DHCP response.

Phone File Server Type	DHCP Response uses...	
	HTTP Source	TFTP Source
Custom	*HTTP Server IP Address*	*TFTP Server IP Address*
Memory Card	*LAN IP Address*	*LAN IP Address*
Manager	*LAN IP Address*	*Manager PC IP Address*

  * Custom
The system will respond to all HTTP file requests from H323 phones. The H323 phone DHCP response contains the TFTP Address from the TFTP Server IP Address above and the HTTP Address from the HTTP Server IP Address above.

  * Memory Card
The system will respond to all HTTP file requests from H323 phones using files from the systems embedded memory card. The DHCP response for H323 phones contains the LAN address of the system for both TFTP and HTTP Address.

  * Manager
The system will forward any H323 phone file request to the configured Manager PC IP address below. HTTP-TFTP relay is used for HTTP requests. The DHCP response for H323 phones contains the LAN address of the system for the HTTP Address.

* Manager PC IP Address: *Default = 0.0.0.0 (Broadcast). Software level = 5.0+.*
This address is used when the Phone File Server Type is set to *Manager*.

* Avaya HTTP Clients Only: *Default = On. Software level = 5.0+.*
If selected, the system will only respond to HTTP requests from sources it identifies as another system, an Avaya phone or application.

* Enable SoftPhone HTTP Provisioning: *Default = Off. Software level = 6.0+.*
This option must be enabled if the Avaya IP Office SoftPhone is being supported.

* Automatic Backup Command: *Default = On. Software level = 6.1+.*
This command is available with IP500v2 systems. When selected, as part of its daily backup process, the system automatically copies the folders and files from the System SD card's */primary* folder to its */backup* folder. Any matching files and folders already present in the */backup* folder are overwritten.

- **Branch Prefix**: *Default = Blank, Range = 0 to 999999999. Software level = 4.1+.*
  Used to identify the system within an Avaya SIP for Branch network linked via an SES server. The branch prefixes of each systems within the network must unique and must not overlap. For example 85, 861 and 862 are okay, but 86 and 861 overlap.
- **Local Number Length**: *Default = Blank (Off), Range = Blank (Off) or 3 to 9. Software level = 4.1+.*
  Set the default length for extension numbers for extensions, users and hunt groups. Entry of an extension number of a different length will cause an error warning within Manager.This field is intended for systems being used in an Avaya SIP for Branch network linked via an SES server, where fixed extension lengths must be used. Note that the **Branch Prefix** and the **Local Number Length** must not exceed 15 digits.
- **Time Server IP Address**: *Default = 0.0.0.0 (Default), Software level = 1.0 to 6.0.*
  For pre-IP Office Release 6.1 systems, the time is either set manually (see Date and Time 727) or obtained using Time protocol (RFC868) requests. Both the Voicemail Pro server and the Manager application can respond to Time requests. Entering 0.0.0.1 disables requesting time server updates. The system makes a request to the specified address following a reboot and every 8 hours afterwards.
  - This setting is replaced for IP Office Release 6.1 systems by the **Time Setting Config Source** and **Time Settings** detailed below.
  - 0.0.0.0 means default operation. In this mode, following a reboot the control unit will send out a time request on its LAN interfaces. It first makes the request to the Voicemail Server IP address in its configuration if set and, if it receives no reply, it then makes a broadcast request.
  - If you are running Manager when the voicemail server starts, voicemail does not start as a time server. It is therefore recommended that you have no copy of Manager running when you start or restart the voicemail server. Manager can be disabled from acting as a RFC868 time server by deselecting the **Enable Time Server** option (File | Preferences | Edit | Preferences 106).
- **Time Setting Config Source**: *Default = Voicemail Pro/Manager*
  The time is either set manually (see Date and Time 727), obtained using Time protocol (RFC868) requests or obtained using Network Time Protocol (RFC958) request. This field is used to select which method is used and to apply ancillary settings based on the selected method.
  - ***None***
    Set the system to not make any time requests. The system time and date needs to be set using a phone with **System Phone Rights** (User | User 264). The system can then automatically apply daylight saving settings to the manually set time.
    - **Automatic DST**: *Default = Off*
      If enabled, the system will apply daylight saving time (DST) changes to the displayed local time. Manager will initially apply default values based on the system locale. However these should be checked to ensure that they match the expected values and any recent changes to local DST requirements.
      - **DST Offset**
        Set the amount by by which the time should be changed when DST is applied of removed.
      - **Local Time to Go Forward**
        Set the time at which the DST Offset should be added to the local time when DST is started.
      - **Local Time to Go Back**
        Set the time at which the DST offset should be removed from the local time when DST is removed.
      - **Clock Forward/Back Settings**
        This drop down can be used to set the paired dates for the start (go forward) and end (go back) of the daylight saving. To add a pair of dates, select *<Add New Entry>* in the drop down and click **Edit**.
        - Select **Clock Forward** and double clicking on the required date. Then select **Clock Backward** and double click on the required date.
        - If necessary multiple pairs can be created to cover a number of years.
  - ***Voicemail Pro/Manager***
    Both the Voicemail Pro service and the Manager program can act as RFC868 Time servers for the system. Use of other RFC868 server sources is not supported. They provide both the UTC time value and the local time as set on the PC. The system makes a request to the specified address following a reboot and every 8 hours afterwards.
      - **IP Address**: *Default = 0.0.0.0 (Broadcast)*
        The address to which the RFC868 request is sent. 0.0.0.0 means default operation. In this mode, following a reboot the control unit will send out a time request on its LAN interfaces. It first makes the request to the Voicemail Server IP address in its configuration if set and, if it receives no reply, it then makes a broadcast request.
      - **Time Offset**: *Default = 00:00*
        This value is not normally set as any time changes, including daylight saving changes, that occur on the PC will be matched by the system.
    - If you are running Manager when the voicemail server starts, voicemail does not start as a time server. It is therefore recommended that you have no copy of Manager running when you start or restart the voicemail server. Manager can be disabled from acting as a RFC868 time server by deselecting the **Enable Time Server** option (File | Preferences | Edit | Preferences 106).

* *SNTP (Release 6.1+)*
  Use a list of SNTP servers to obtain the UTC time. The entries in the list are used one at a time in order until there is a response. The system makes a request to the specified addresses following a reboot and every hour afterwards.

  * Time Server Address: *Default = Blank*
    Enter a list of IP addresses or names for the SNTP servers. Separate each entry with a space. The use of broadcast addresses is not supported. The list is used in order of the entries until a response is received.

  * Time Zone: *Default = Match to system Locale*
    The value is defaulted to match the system locale.

  * Local Time Offset from UTC: *Default = 00:00*
    This setting is used to set the local time difference from the UTC time value provided by an SNTP server. For example, if the system is 5 hours behind UTC, this field should be configured with -5:00 to make the adjustment. The time offset can be adjusted in 15 minute increments. If also using the daylight time saving settings below, use this offset to set the non-DST local time.

  * Automatic DST: *Default = Off*
    If enabled, the system will apply daylight saving time (DST) changes to the displayed local time. Manager will initially apply default values based on the system locale. However these should be checked to ensure that they match the expected values and any recent changes to local DST requirements.

    * DST Offset
      Set the amount by by which the time should be changed when DST is applied of removed.

    * Local Time to Go Forward
      Set the time at which the DST Offset should be added to the local time when DST is started.

    * Local Time to Go Back
      Set the time at which the DST offset should be removed from the local time when DST is removed.

    * Clock Forward/Back Settings
      This drop down can be used to set the paired dates for the start (go forward) and end (go back) of the daylight saving. To add a pair of dates, select *<Add New Entry>* in the drop down and click Edit.

      * Select Clock Forward and double clicking on the required date. Then select Clock Backward and double click on the required date.

      * If necessary multiple pairs can be created to cover a number of years.

* File Writer IP Address: *Default = 0.0.0.0 (Disabled)*
  On systems with an Avaya memory card (Small Office Edition, IP406 V2, IP500 and IP500v2), this field sets the address of the PC allowed to send files to the memory card in their PCMCIA or CF TII slot.

* License Server IP Address: *Default = 255.255.255.255*
  This is the IP address of the server providing license key validation for the system. The serial number of the Feature Key dongle at that address must match the serial number used to generate the licenses in the system's configuration. See License 378 for more details.

  * For parallel and USB Feature Key dongles this address should be the IP address of a PC running the Feature Key server software. Note that separate systems cannot use the same Feature Key server for license validation, nor can a system validate its licenses against more than one address.

  * For a serial port Feature Key dongle, the address should be set to 0.0.0.0.

  * For IP500 and IP500v2 systems, this field is hidden as those systems must use a Feature Key dongle inserted into the control unit.

* Dongle Serial Number
  This field is for information only. It shows the serial number of the feature key dongle against which the system last validated its licenses. *Local* is shown for a serial port or Smart Card feature key plugged directly into the control unit. *Remote* is shown for a parallel or USB feature key connected to a feature Key Server PC. For IP400 and IP500 systems, the serial number is also printed on the feature key dongle and prefixed with SN. For IP500v2 systems, the serial number is printed on the System SD card and prefixed with FK.

* AVPP IP Address: Default = 0.0.0.0 (Disabled)
  Where Avaya 3600 Series SpectraLink wireless handsets are being used with the system, this field is used to specify the IP address of the Avaya Voice Priority Processor (AVPP).

* Conferencing Center URL: *Default = Blank (Disabled).*
  This is the root URL of the web server being used to support Conferencing Center, for example http://server/. This address is then used by the Phone Manager and SoftConsole applications to launch Conferencing Center functions. For Phone Manager, setting this value enables use of the join conference controls.

* DSS Status:
  This setting has been moved to the System | Telephony | Telephony 160 sub-tab.

* Beep on Listen: *Default = On (USA)/On (Rest of World)*
  This setting has been moved to the System | Telephony | Tones & Music 163 sub-tab.

- Hide auto record: *Default = On (USA)/Off (Rest of World)*
  During call recording by Voicemail Pro, some Avaya phones display REC or similar to show that the call is being recorded. When on, Hide auto record suppresses this recording indication.

- Favour RIP Routes over Static Routes: *Default = Off*
  RIP 369 can be enabled on the system LAN1 148 and LAN2 155 interfaces, and on specific Services 331. When this setting is on, the RIP route to a destination overrides any static route to the same destination in the system's IP Routes 368, regardless of the RIP route's metric. The only exception is RIP routes with a metric of 16 which are always ignored. Note: If a previously learnt RIP route fails, the system applies a metric of 16 five minutes after the failure. When off, any RIP route to a destination for which a static route has been configured is ignored.

5.3.2 LAN1

This tab is used to configure the behavior of the services provided by the system LAN's. Depending in the control unit, up to 2 LAN's (LAN1 and LAN2) can be configured.

Depending on the type of control unit, the relationship between the physical RJ45 Ethernet ports and LAN1 and LAN2 within the system configuration varies as follows:

- **IP500/IP500v2**
  The control unit has 2 RJ45 Ethernet ports, marked as LAN and WAN. These form a full-duplex managed layer-3 switch. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.
- **IP460 V2**
  The control unit has 8 RJ45 Ethernet ports marked as LAN 1 to 8. These form a full-duplex unmanaged layer-2 LAN switch. Ports are auto-MDI/MDIX. Within the system configuration the physical LAN ports are LAN1.
  - For IP Office 4.1+, port 8 can be configured to act as LAN2 using the Use Port 8 as LAN2 option on the LAN1 LAN Settings tab.
- **IP412**
  The control unit has 2 RJ45 Ethernet ports marked as LAN 1 to 2. These form a half-duplex managed layer-3 switch. Both ports are fixed MDI crossover ports. Within the system configuration, physical port 1 is LAN1, physical port 2 is LAN2.
- **Small Office Edition**
  The control unit has 4 RJ45 Ethernet ports marked LAN 1 to 4. These form a full-duplex unmanaged layer-2 switch. An addition RJ45 Ethernet socket marked as WAN exists. With the LAN ports this acts as a managed layer-3 switch. Within the system configuration, the physical LAN ports are LAN1, the physical WAN port is LAN2.

5.3.2.1 LAN Settings

System | LAN1 | LAN Settings	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- IP Address: *Default = 192.168.42.1 or DHCP client. See Default Settings 65.*
  This is the IP address of the Control Unit on LAN1. If the control unit is also acting as a DHCP server on LAN1 then this address is the starting address of the DHCP address range.

- IP Mask: *Default = 255.255.255.0 or DHCP client. See Default Settings 65.*
  This is the IP subnet mask used with the IP address.

- Use Port 8 as LAN2: *Default = Off. Software level = 4.1+.*
  This option is provided for IP406 V2 control units with IP Office 4.1 or higher software only. When selected, LAN port 8 on the control unit acts as LAN2 155 for the control unit. Note that this setting is retained by the Manager even if the system is defaulted. To change the setting, the value in the configuration should be changed and the configuration sent back to the system for an immediate reboot.

- Primary Trans. IP Address: *Default = 0.0.0.0 (Disabled)*
  This setting is only available on control units that support a LAN2. Any incoming IP packets without a service or session are translated to this address if set.

- RIP Mode: *Default = None*
  Routing Information Protocol (RIP) 369 is a method by which network routers can exchange information about device locations and routes. Routes learnt using RIP are known as 'dynamic routes'. The system also supports 'static routes' though its IP Route 367 entries.

  - *None*
    The LAN does not listen to or send RIP messages.

  - *Listen Only (Passive)*
    Listen to RIP-1 and RIP-2 messages in order to learn RIP routes on the network.

  - *RIP1*
    Listen to RIP-1 and RIP-2 messages and send RIP-1 responses as a sub-network broadcast.

  - *RIP2 Broadcast (RIP1 Compatibility)*
    Listen to RIP-1 and RIP-2 messages and send RIP-2 responses as a sub-network broadcast.

  - *RIP2 Multicast*
    Listen to RIP-1 and RIP-2 messages and send RIP-2 responses to the RIP-2 multicast address.

- Enable NAT: *Default = Off*
  This setting controls whether NAT 798 should be used for IP traffic from LAN1 to LAN2.It is only available on Small Office Edition, IP412, IP500 and IP500v2 systems. This setting should not be used on the same LAN interface as a connected WAN3 expansion module.

- Number of DHCP IP Addresses: *Default = 200 or DHCP client. See Default Settings 65, Range = 1 to 999.*
  This defines the number of sequential IP addresses available for DHCP clients.

- DHCP Mode: *Default = DHCP Client. See Default Settings 65.*
  This controls the control unit's DHCP mode for LAN1. When doing DHCP; LAN devices are allocated addresses from the bottom of the available address range upwards, Dial In users are allocated addresses from the top of the available range downwards. If the control unit is acting as a DHCP server on LAN1 and LAN2, Dial in users are allocated their address from the LAN1 pool of addresses first.

  - *Server*
    When this option is selected, the system will act as a DHCP Server on LAN1, allocating address to other devices on the network and to PPP Dial in users.

  - *Disabled*
    When this option is selected, the system will not use DHCP. It will not act as a DHCP server and it will not request an IP address from a DHCP server on this LAN.

  - *Dial In*
    When this option is selected, the system will allocate DHCP addresses to PPP Dial In users only. On IP500/IP500v2 systems using DHCP pools, only addresses from a pool on the same subnet as the system's own LAN address will be used.

  - *Client*
    When this option is selected, the system will request its IP Address and IP Mask from a DHCP server on the LAN.

  - Advanced: *Software level = 4.2+ (IP500/IP500v2 only).*
    IP500/IP500v2 control units can be configured with a number of DHCP pools from which the system can then issue addresses. See DHCP Pools 154.

5.3.2.2 VoIP

These settings are only shown on IP Office 4.0+ systems. On pre-4.0 systems the H.323 settings were located on a separate System | H.323 Gatekeeper tab for the whole system.

System | LAN1 | VoIP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✗.

- H323 Gatekeeper Enable: *Default = On*
  This settings enables gatekeeper operation.

- SIP Trunks Enable: *Default = On*
  This settings enables support of SIP trunks. It also requires entry of ***SIP Trunk Channels*** licenses.

- SIP Registrar Enable: *Default = On. Software level = 5.0+.*
  This setting enables support for SIP extensions.When selected, the SIP Endpoints 155 tab if visible. These are also require ***IP End Point*** license.

- H323 Auto-create Extn: *Default = On*
  When this option is on, an extension entry is automatically created for H.323 phones registering themselves with the system as their gatekeeper. SIP Extensions use a separate setting on the SIP Endpoints 155 tab.

- H323 Auto-create User: *Default = Off*
  When this option is on and H323 Auto-create Extn is also on, when a new H.323 extension is created a matching user record is also created.

- Enable RTCP Monitor On Port 5005: *Default = On. Software level = 5.0+.*
  For 1600, 4600, 5600 and 9600 Series H323 phones, the system can collect VoIP QoS (Quality of Service) data from the phones. For other phones, including non-IP phones, it can collect QoS data for calls if they use a VCM channel. The QoS data collected by the system is displayed by the System Status Application.

  - This setting is mergeable. However it only affects H323 phones when the register with the system. therefore any change to this setting requires H323 phones that have already been registered to be rebooted. Avaya H323 phones can be remotely rebooted using the System Status Application.

  - The QoS data collected includes: RTP IP Address, Codec, Connection Type, Round Trip Delay, Receive Jitter, Receive Packet Loss.

  - This setting is not the same as the RTCPMON option within Avaya H323 phone settings. The system does not support the RTCPMON option.

- RTP Port Number Range: *Software level = 3.0+.*
  For each VoIP call, a receive port for incoming Real Time Protocol (RTP) traffic is selected from a defined range of possible ports, using the even numbers in that range. The Real Time Control Protocol (RTCP) traffic for the same call uses the RTP port number plus 1, that is the odd numbers. For control units and Avaya H.323 IP phones, the default port range used is 49152 to 53246. On some installations, it may be a requirement to change or restrict the port range used. It is recommended that only port numbers between 49152 and 65535 are used, that being the range defined by the Internet Assigned Numbers Authority (IANA) for dynamic usage.

  - Port Range (minimum): *Default = 49152. Range = 1024 to 64510.*
    This sets the lower limit for the RTP port numbers used by the system.

  - Port Range (maximum): *Default = 53246. Range = 2048 to 65534.*
    This sets the upper limit for the RTP port numbers used by the system. The gap between the minimum and the maximum must be at least 1024.

- DiffServ Settings
  When transporting voice over low speed links it is possible for normal data packets (1500 byte packets) to prevent or delay voice packets (typically 67 or 31 bytes) from getting across the link. This can cause unacceptable speech quality. Therefore it is important that all traffic routers and switches in a network to have some form of Quality of Service mechanism (QoS). QoS routers are essential to ensure low speech latency and to maintain sufficient audible quality.

    - The system supports the DiffServ (RFC2474) QoS mechanism. This uses a Type of Service (ToS) field in the IP packet header. The system uses this field to prioritize voice and voice signaling packets on its WAN interfaces. Note that the system does not perform QoS for its Ethernet ports including the WAN Ethernet port on the Small Office Edition.

    - The hex and decimal entry fields for the following values are linked, the hex value being equal to the decimal multiplied by 4.

  - DSCP (Hex): *Default = B8 (Hex)/46 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
    The DiffServ Code Point (DSCP) setting applied to VoIP calls. For correct operation, especially over WAN links, the same value should be set at both ends.

  - DSCP Mask (Hex): *Default = FC (Hex)/63 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
    Allows a mask to be applied to packets for the DSCP value.

  - SIG DSCP (Hex): *Default = 88 (Hex)/34 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
    This setting is used to prioritize VoIP call signaling.

- DHCP Settings

  - Primary Site Specific Option Number (SSON): *Default = 176, Range = 128 to 254.*
    A site specific option number (SSON) is used as part of DHCP to request additional information. For system it is used by Avaya IP Phones being supported by DHCP. 176 is the default SSON used by 4600 Series and 5600 Series IP phones.

  - Secondary Site Specific Option Number (SSON): *Default = 242, Range = 128 to 254.*
    Similar to the primary SSON. 242 is the default SSON used by 1600 and 9600 Series IP phones requesting installation settings via DHCP.

  - VLAN: *Default = Not present. Software level = Release 5+.*
    This option is applied to H323 phones using the system for DHCP support. If set to *Disabled*, the L2Q value indicated to phones in the DHCP response is 2 (disabled). If set to *Not Present*, no L2Q value is included in the DHCP response.

  - 1100 Voice VLAN Site Specific Option Number (SSON): *Default = 232. Software level = 6.1*
    This is the SSON used for responses to 1100/1200 Series phones using the system for DHCP.

  - 1100 Voice VLAN IDs: *Default = blank. Software level = 6.1*
    For 1100/1200 phone being supported by DHCP, this field sets the VLAN ID that should be provided if necessary. Multiple IDs (up to 10) can be added, each separated by a + sign.

- RTP keepalives: *Software level = 6.1+*
  These settings can be used with SIP trunks associated with the LAN through their Use Network Topology Info setting (Line | Transport 236). For some scenarios, with frequent call forwarding on the same SIP trunk, speech path may be lost during the connection. The use of periodic keepalive packets may prevent the issue.

  - Scope: *Default = Disabled*
    Select whether the sending of keepalive packets should be disabled or sent for RTP or for both RTP and RTCP.

  - Periodic timeout: *Default = 0 (Off), Range = 0 to 180 seconds.*
    Sets how long the system will wait before sending a keepalive if no other packets of the select SCOPE are seen.

  - Initial keepalives: *Default = Disabled.*
    If enabled, keepalives can also been sent during the initial connection setup.

5.3.2.3 Network Topology

These settings are used for SIP trunk connections from the LAN. Use of SIP requires entry of SIP Trunk Channels licenses. For further details of system SIP operation refer to the SIP Line 230 section.

STUN (Simple Traversal of UDP through NAT) is a mechanism used with UDP SIP to overcome the effect of NAT firewalls. Test SIP packets are sent to a STUN server. The STUN server replies and includes copies of the packets it received in the reply. By comparing the packet sent to the STUN server and the copies of the packets it received, it is possible to determine the type of NAT firewall and to then modify future SIP packets to overcome negative effects of NAT.

The use of STUN is unnecessary if the SIP ITSP uses a Session Border Controller (SBC).

System | LAN1 | Network Topology Settings	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✗.

The following fields can be completed either manually or the system can attempt to automatically discover the appropriate values. To complete the fields automatically, only the STUN Server IP Address is required. STUN operation is then tested by clicking Run STUN. If successful the remaining fields are filled with the results.

- STUN Server IP Address: *Default = Blank*
  This is the IP address of the SIP ITSP's STUN server. The system will send basic SIP messages to this destination and from data inserted into the replies can try to determine the type NAT changes being applied by any firewall between it and the ITSP.

- STUN Port: *Default = 3478. Software level = 4.1+.*
  Defines the port to which STUN requests are sent if STUN is used.

- Firewall/NAT Type: *Default = Unknown*
  The settings here reflect different types of network firewalls. Options include *Blocking Firewall*, *Symmetric Firewall*, *Open Internet*, *Symmetric NAT*, *Full Cone NAT*, *Restricted Cone NAT*, *Port Restricted Cone NAT*, *Static Port Block* and *Unknown*.

  - *Open Internet*
    No action required. If this mode is selected, settings obtained by STUN lookups are ignored. The IP address used is that of the system LAN interface.

  - *Symmetric Firewall*
    SIP packets are unchanged but ports need to be opened and kept open with keep-alives. If this type of NAT is detected or manually selected, a warning 'Communication is not possible unless the STUN server is supported on same IP address as the ITSP' will be displayed as part of the manager validation.

  - *Full Cone NAT*
    A full cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Furthermore, any external host can send a packet to the internal host, by sending a packet to the mapped external address. SIP packets need to be mapped to NAT address and Port; any Host in the internet can call in on the open port, that is the local info in the SDP will apply to multiple ITSP Hosts. No warning will be displayed for this type of NAT because the system has sufficient information to make the connection).

  - *Symmetric NAT*
    A symmetric NAT is one where all requests from the same internal IP address and port, to a specific destination IP address and port, are mapped to the same external IP address and port. If the same host sends a packet with the same source address and port, but to a different destination, a different mapping is used. Furthermore, only the external host that receives a packet can send a UDP packet back to the internal host. SIP Packets need to be mapped but STUN will not provide the correct information unless the IP address on the STUN server is the same as the ITSP Host. If this type of NAT/Firewall is detected or manually selected, a warning 'Communication is not possible unless the STUN server is supported on same IP address as the ITSP' will be displayed as part of the manager validation.

  - *Restricted Cone NAT*
    A restricted cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Unlike a full cone NAT, an external host (with IP address X) can send a packet to the internal host only if the internal host had previously sent a packet to IP address X. SIP packets needs to be mapped. Responses from hosts are restricted to those that a packet has been sent to. So if multiple ITSP hosts are to be supported, a keep alive will need to be sent to each host. If this type of NAT/Firewall is detected or manually selected, no warning will be displayed for this type of NAT.

- *Port Restricted Cone NAT*
  A port restricted cone NAT is like a restricted cone NAT, but the restriction includes port numbers. Specifically, an external host can send a packet, with source IP address X and source port P, to the internal host only if the internal host had previously sent a packet to IP address X and port P. SIP packets needs to be mapped. Keep-alives must be sent to all ports that will be the source of a packet for each ITSP host IP address. If this type of NAT/Firewall is detected or manually selected, no warning will be displayed for this type of NAT. However, some Port Restricted NAT's have been found to be more symmetric in behavior, creating a separate binding for each opened Port, if this is the case the manager will display a warning 'Communication is not possible unless the STUN server is supported on same IP address as the ITSP' as part of the manager validation.

- Static Port Block: *Software level = 4.1+.*
  Use the RTP Port Number Range specified on the VoIP 150 tab without STUN translation. Those ports must be fixed as open on any NAT firewall involved.

- Binding Refresh Time (seconds): *Default = 0 (Never), Range = 0 to 3600 seconds.*
  Having established which TCP/UDP port number to use, through either automatic or manual configuration, the system can send recurring 'SIP Options requests' to the remote proxy terminating the trunk. Those requests will keep the port open through the firewall. Requests are sent every x seconds as configured by this field.

  - Note: If a binding refresh time has not been set you may experience problems receiving inbound SIP calls as they are unable to get through the Firewall. In these circumstances make sure that this value has been configured.

- Public IP Address: *Default = 0.0.0.0*
  This value is either entered manually or discovered by the Run STUN process. If no address is set, the system LAN1 address is used.

- Public Port: *Default = 0*
  This value is either entered manually or discovered by the Run STUN process.

- Run STUN
  This button tests STUN operation between the system LAN and the STUN Server IP Address set above. If successful the results are used to automatically fill the remaining fields with appropriate values discovered by the system. Before using Run STUN the SIP trunk must be configured.

  - When this option is used, a information icon is shown against the fields to indicate that the values were automatically discovered rather than manually entered.

- Run STUN on startup: *Default = Off*
  This option is used in conjunction with values automatically discovered using Run STUN. When selected, the system will rerun STUN discovery whenever the system is rebooted or connection failure to the SIP server occurs.

5.3.2.4 DHCP Pools

DHCP pools allows the configuration of up to 8 ranges of IP addresses for allocation by the Manager when acting as a DHCP server. By default the DHCP settings (IP Address, IP Mask and Number of DHCP IP Addresses) set on the LAN Settings 149 tab are reflected by the first pool here.

- For support of PPP Dial In address requests, at least one of the pools must be on the same subnet as the system's LAN. Only addresses from a pool on the same subnet as the system's own LAN address will be used for PPP Dial In.

System | LAN1 | DHCP Pools	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✗, IP412 ✗, IP500 ✓, IP500v2 ✓.
Software Level	4.2+.
Mergeable	✗.

- Apply to Avaya IP Phones Only: *Default = Off.*
  If this option is selected, the DHCP addresses are only used for requests from Avaya IP phones. Other devices connected to the system LAN will have to use static addresses or obtain their address from another DHCP server.
  - In addition to the above control, Avaya IP phones will only complete DHCP against a DHCP server configured to supports a Site Specific Option Number (SSON) that matches that set on the phone. The SSON numbers supported by the system DHCP are set on the VoIP 150 sub-tab.
- DHCP Pool
  Up to 8 pools can be added. The first pool matches the IP Address, IP Mask and Number of DHCP IP Addresses on the LAN Settings 149 sub-tab. When adding or editing pools, Manager will attempt to warn about overlaps and conflicts between pools.
  - Start Address
    Sets the first address in the pool.
  - Subnet Mask: *Default = 255.255255.0*
    Sets the subnet mask for addresses issued from the pool.
  - Default Router: *Default = 0.0.0.0*
    For pools issuing IP addresses on the same subnet as the system LAN's, *0.0.0.0* instructs the system to determined the actual default router address to issue by matching the IP address/subnet mask being issued in the IP Routing 368 table. This matches the default behaviour used by systems without multiple pools. For pools issuing addresses not on the same subnet as the system LAN's, the default router should be set to the correct value for devices on that subnet.
  - Pool Size: *Default = 0*
    Set the number of DHCP client addresses available in the pool.

5.3.2.5 SIP Registrar

This tab is used to set the system parameters for the system acting as a SIP Registrar to which SIP endpoint devices can register. Separate SIP registrars can be configured on LAN1 and LAN2.

Registration of a SIP endpoint requires an available *IP Endpoints* license. SIP endpoints are also still subject to the extension capacity limits of the system.

System | LAN1 | SIP Endpoints	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✗.

- Domain Name: *Default = Blank*
  This is the local SIP registrar domain name that will be needed by SIP endpoints in order to register with the system. If this field is left blank, registration is against the LAN IP address.
- Layer 4 Protocol: *Default = Both TCP & UDP.*
  Both TCP and UDP SIP endpoints are supported. This field can be used to restrict the system to just TCP or UDP if required.
- TCP Port: *Default = 5060.*
  The port to use for TCP support.
- UDP Port: *Default = 5060.*
  The port to use for UDP support.
- Challenge Expiry Time (secs): *Default = 10.*
  The challenge expiry time is used during SIP extension registration. When a device registers, the system SIP Registrar will send a challenge back to the device and waits for an appropriate response. If the response is not received within this timeout the registration is failed.
- Auto-create Extn/User: *Default = On.*
  If on, a new extension and user entries are created by the registration of a new SIP endpoint. If off, SIP endpoint can only register against existing configuration entries.

5.3.3 LAN2

LAN2 is not supported by all control units.

- On the Small Office Edition and IP500/IP500v2 control units the LAN2 settings are used for the RJ45 Ethernet port labeled WAN.
- On the IP412 control units LAN2 settings are used for the RJ45 ethernet port labeled LAN2.
- For IP406 V2 control units running IP Office 4.1+, the RJ45 LAN port 8 can be selected to act be LAN2 if required. This is done using the Use Port 8 as LAN2 option on the LAN1 | LAN Settings 149 tab.

System | LAN2	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓*, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

*Optional on IP Office 4.1+.

The fields available for LAN2 are the same as for LAN1 148 except for the following additional field:

- Firewall: *Default = <None> (No firewall)*
  Allows the selection of a system firewall to be applied to traffic routed from LAN2 to LAN1.

5.3.4 DNS

DNS is a mechanism through which the URL's requested by users, such as www.avaya.com, are resolved into IP addresses. These requests are sent to a Domain Name Server (DNS) server, which converts the URL to an IP address. Typically the internet service provider (ISP) will specify the address of the DNS server their customers should use.

WINS (Windows Internet Name Service) is a similar mechanism used within a Windows network to convert PC and server names to IP addresses via a WINS server.

If the system is acting as a DHCP server, in addition to providing clients with their own IP address settings for LAN1 148 or LAN2 155 it can also provide them with their DNS and WINS settings if requested by the client

System | DNS	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- DNS Service IP Address: *Default = 0.0.0.0 (Do not provide DNS/Use DNS forwarding)*
  This is the IP address of an DNS Server. Your Internet service provider or network administrator provides this information. If this field is left blank, the system uses its own address as the DNS server for DHCP client and forward DNS requests to the service provider when Request DNS is selected in the service being used (Service | IP 335).
  - Backup DNS Server IP Address: *Default = 0.0.0.0 (No backup)*
- DNS Domain: *Default = Blank (No domain)*
  This is the domain name for your IP address. Your Internet service provider or network administrator provides this. Typically this field is left blank.
- WINS Server IP Address: *Default = 0.0.0.0 (Do not provide WINS)*
  This is the IP address of your local WINS server. This is only used by Windows PCs, and normally points to an NT server nominated by your network administrator as your WINS server. Setting a value will result in also sending a mode of "hybrid".
  - Backup WINS Server IP Address: *Default = 0.0.0.0 (No backup)*
- WINS Scope: *Default = Blank (no scope)*
  This is provided by your network administrator or left blank.

5.3.5 Voicemail

The following settings are used to set the system's voicemail server type and location. The fields are enabled or grayed out as appropriate to the selected voicemail type. Refer to the appropriate voicemail system installation manual for full details.

System | Voicemail	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✓*Changes to Voicemail Type require a reboot.

- Voicemail Type: *Default = Embedded Voicemail*
  Sets the type of voicemail system being used.

  - *None*
    No voicemail operation.

  - *Centralized Voicemail*
    Select this option when using a Voicemail Pro system installed and licensed on another system in a Small Community Network. The outgoing line group for connection to the system with the Voicemail Pro should be entered as the Voicemail Destination.

  - *Distributed Voicemail: Software level = 6.0+.*
    This option can be used when additional Voicemail Pro voicemail servers are installed in a Small Community Network and configured to exchange messages with the central voicemail server using email. This option is used if this system should use one of the additional servers for its voicemail services rather than the central sever. When selected, the Voicemail Destination field is used for the outgoing SCN line to the central system and the Voicemail IP Address is used for the IP address of the distributed voicemail server the system should use.

  - *Embedded Voicemail*
    On systems with an Avaya memory card (Small Office Edition, IP406 V2, IP500 and IP500v2), select this option to run embedded voicemail which stores messages and prompts on the memory card. It also supports internal Auto Attendant 400 configuration through the IP Office system configuration. The IP500/IP500v2 supports 2 simultaneous embedded voicemail calls by default but can be licensed for up to 6. The licensed limit applies to total number of callers leaving messages, collecting messages and or using an auto attendant.

  - *Group Voicemail*
    This option is used to support third-party voicemail systems attached by extension ports in the group specified as the Voicemail Destination.

  - *Remote Audix Voicemail*
    Select this option if using a remote Avaya Intuity Audix or MultiMessage voicemail system. Requires entry of an *Audix Voicemail* license in Licenses.

  - *Voicemail Lite/Pro*
    Select this option when using Voicemail Lite or Voicemail Pro. The IP address of the PC being used should be set as the Voicemail IP Address. Use of Voicemail Pro requires licenses for the number of simultaneous calls to be supported. Note that Voicemail Lite is not supported with IP Office Release 5+.

- Voicemail Destination: *Default = blank*
  This setting is used when the Voicemail Type is set to *Remote Audix Voicemail*, *Centralized Voicemail* or *Distributed Voicemail*. It is used to enter the outgoing line group of the lines configured for connection to the phone system hosting the central voicemail server. It is also used for *Group Voicemail* to specify the group whose user extensions are connected to the 3rd party voicemail system.

- Voicemail IP Address: *Default = 255.255.255.255*
  This setting is used when the Voicemail Type is set to *Voicemail Pro* or *Distributed Voicemail*. It is the IP address of the PC running the voicemail server that the system should use for its voicemail services. If set as 255.255.255.255, the control unit broadcasts on the LAN for a response from a voicemail server. If set to a specific IP address, the system connects only to the voicemail server running at that address.

- Backup Voicemail IP Address : *Default = 0.0.0.0 (Off). Software level = 6.0+.*
  This option is supported with Voicemail Pro. An additional voicemail server can be setup but left unused. If contact to the voicemail server specified by the Voicemail IP Address is lost, responsibility for voicemail services is temporarily transferred to this backup server address.

- Messages Button Goes to Visual Voice: *Default = On. Software level = 4.2+*
  Visual Voice allows phone users to check their voicemail mailboxes and perform action such as play, delete and forward messages through menus displayed on their phone. By default, on phones with a MESSAGES button, the navigation is via spoken prompts. This option allows that to be replaced by Visual Voice on phones that support Visual Voice menus. For further details see Visual Voice 668.

- Voicemail Password : *Default = blank. Software level = 2.1 to 3.1.*
  This password is used by the main unit to confirm connection has been made to the correct Voicemail Pro Server. The password entered must correspond to the password set on the Voicemail Pro server. This entry should be left blank when using the any other voicemail server application. For Manager 3.2 and higher this value is set through the Manager security settings.

- Audix UDP
  Available if the voicemail type Remote Audix Voicemail is selected. Needs to be completed with a four digit number from the Universal Dial Plan of the Avaya Communication Manager system.

- Maximum Record Time: *Default = 120 seconds, Range = 30 to 180 seconds. Software level = 3.0+.*
  This field is only available when Embedded Voicemail is selected as the Voicemail Type. The value sets the maximum record time for messages and prompts.

- Voicemail Channel Reservation: *Software level = 4.0+*
  These settings allow the channels between the system and Voicemail Pro to be reserved for particular functions. Unreserved channels can be used for any function but reserved channels cannot be used for any function other than that indicated. These settings are not available unless the configuration includes validated licenses for the total number of voicemail channels.

    - Note that the voicemail server also restricts the maximum number of channels that can be used for some services that would be taken from the Unreserved Channels pool. Alarms and callbacks are each limited to up to 2 channels at any time. Outcalling and conference invites are each limited to up to 5 channels at any time.

  - Unreserved Channels
    This setting cannot be changed and by default will show the total number of licensed voicemail channels. This number will decrease as channels are reserved for the following functions.

  - Mailbox Access: *Default = 0*
    This setting sets the number of channels reserved for users accessing mailboxes to collect messages.

  - Auto-Attendant: *Default = 0*
    This setting sets the number of channels reserved for users directed to Voicemail Pro short code and module start points.

  - Voice Recording: *Default = 0*
    This setting sets the number of channels reserved for voice recording other than mandatory voice recording (see below). If no channels are available recording does not occur though recording progress may be indicated.

  - Mandatory Voice Recording: *Default = 0*
    This setting sets the number of channels reserved for mandatory voice recording. When no channels are available for a call set to mandatory recording, the call is barred and the caller hears busy tone.

  - Announcements: *Default = 0*
    This setting sets the number of channels reserved for announcements. When no channels are available calls continue without announcements.

- DTMF Breakout: *Software level = 5.0+*
  Allows system defaults to be set. These are then applied to all user mailboxes unless the users own settings differ.

  - Reception / Breakout (DTMF 0)
    The number to which a caller is transferred if they press *0* while listening to the mailbox greeting rather than leaving a message (**0* on embedded voicemail).

    - For systems set to Intuity emulation mode, the mailbox user can also access this option when collecting their messages by dialing **0*.

    - If the mailbox has been reached through a call flow containing a Leave Mail action, the option provided when *0* is pressed are:

      - For IP Office mode, the call follows the Leave Mail action's *Failure* or *Success* results connections depending on whether the caller pressed *0* before or after the record tone.

      - For Intuity mode, pressing *0* always follows the Reception / Breakout (DTMF 0) setting.

  - Breakout (DTMF 2)
    The number to which a caller is transferred if they press 2 while listening to the mailbox greeting rather than leaving a message (**2* on embedded voicemail). Pre-IP Office 5 this option is not support for Voicemail Pro running in IP Office mailbox mode.

  - Breakout (DTMF 3)
    The number to which a caller is transferred if they press *3* while listening to the mailbox greeting rather than leaving a message (**3* on embedded voicemail). Pre-IP Office 5 this option is not support for Voicemail Pro running in IP Office mailbox mode.

- SIP Settings
  For Voicemail Pro, for calls made or received on a SIP line where any of the line's SIP URI fields are set to *Use Internal Data*, that data is taken from these settings. These options are hidden if there are no system SCN lines in the configuration or no SIP lines with a URI set to *Use Internal Data*.

- SIP Name: *Default = User name.*
  The value from this field is used when the From field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- SIP Display Name (Alias): *Default = User name.*
  The value from this field is used when the Display Name field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- Contact: *Default = User name.*
  The value from this field is used when the Contact field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- Anonymous: *Default = Off.*
  If the From field in the SIP URI is set to *Use Internal Data*, selecting this option inserts *Anonymous* into that field rather than the SIP Name set above.

5.3.6 Telephony

This tab is used to set the default telephony operation of the system. Some settings shown here can be overridden for individual users through their User | Telephony 274 tab. The settings are split into a number of sub-tabs.

5.3.6.1 Telephony

System | Telephony | Telephony	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✓*

*Changes to Companding LAW and Automatic Codec Preference require a reboot.

- Analog Extensions
  These settings apply only to analog extension ports provided by the system.

  - Default Outside Call Sequence: *Default = Normal*
    This setting is only used with analog extensions. It sets the ringing pattern used for incoming external calls. For details of the ring types see Ring Tones 762 in the Telephone Features section. This setting can be overridden by a user's User | Telephony | Call Settings | Outside Call Sequence 274 setting. Note that changing the pattern may cause fax and modem device extensions to not recognize and answer calls.

  - Default Inside Call Sequence: *Default = Ring Type 1*
    This setting is only used with analog extensions. It sets the ringing pattern used for incoming internal calls. For details of the ring types see Ring Tones 762 in the Telephone Features section. This setting can be overridden by a user's User | Telephony | Call Settings | Inside Call Sequence 274 setting.

  - Default Ring Back Sequence: *Default = Ring Type 2*
    This setting is only used with analog extensions. It sets the ringing pattern used for ringback calls such as hold return, park return, voicemail ringback, and Ring Back when Free. For details of the ring types see Ring Tones 762 in the Telephone Features section. This setting can be overridden by a user's User | Telephony | Call Settings | Ringback Call Sequence 274 setting.

  - Restrict Analog Extension Ringer Voltage: *Default = Off. Software level = 6.1+*
    Supported on IP500v2 systems only. If selected, the ring voltage on analogue extension ports on the system is limited to a maximum of 40V Peak-Peak. Also when selected, the message waiting indication (MWI) settings for analog extension 254 are limited to *Line Reversal A*, *Line Reversal B* or *None*. Any analog extension already set to another MWI setting is forced to *Line Reversal A*.

- Dial Delay Time (secs): *Default = 4 (USA/Japan) or 1 (ROW), Range = 1 to 99 seconds.*
  This setting sets the time the system waits following a dialed digit before it starts looking for a short code match. In situations where there are potential short codes matches but not exact match, it also sets the delay following the dialing of a digit before dialing complete is assumed. See the Short Codes 426 section.

- Dial Delay Count: *Default = 0 digits (USA/Japan) or 4 digits (ROW), Range = 0 to 30 digits.*
  This setting sets the number of digits dialed after which the system starts looking for a short code match regardless of the Dial Delay Time.

- Default No Answer Time (secs): *Default = 15 seconds, Range = 6 to 99999 seconds.*
  This setting controls the amount of time before an alerting call is considered as unanswered. How the call is treated when this time expires depends on the call type.

  - For calls to a user, the call follows the user's Forward on No Answer settings if enabled. If no forward is set, the call will go to voicemail if available or else continues to ring. This timer is also used to control the duration of call forwarding if the forward destination does not answer. It also controls the duration of ringback call alerting. This setting is overridden by the User | Telephony | Call Settings | No Answer Time 274 setting for a particular user if different.

  - For calls to hunt groups, this setting controls the time before the call is presented to the next available hunt group member. This setting is overridden by the Hunt Group | Hunt Group | No Answer Time 306 setting for a particular hunt group if different.

- Hold Timeout (secs): *Default = 120 (US) or 15 (Rest of World), Range = 0 (Off) to 99999 seconds.*
  This setting controls how long calls remain on hold before recalling to the user who held the call. Note that the recall only occurs if the user has no other connected call. Recalled calls will continue ringing and do not follow forwards or go to voicemail.

- Park Timeout (secs): *Default = 300 seconds, Range 0 (Off) to 99999 seconds.*
  This setting controls how long calls remain parked before recalling to the user who parked the call. Note that the recall only occurs if the user has no other connected call. Recalled calls will continue ringing and do not follow forwards or go to voicemail.

- Ring Delay: *Default = 5 seconds, Range = 0 to 98 seconds. Software level = 3.2+.*
  This setting is used when any of the user's programmed appearance buttons is set to Delayed ringing. Calls received on that button will initially only alert visually. Audible alerting will only occur after the ring delay has expired. This setting can be overridden by a ring delay set for an individual user (User | Telephony | Multi-line Options | Ring Delay 278).

- Call Priority Promotion Time (secs): *Default = Disabled, Range = Disabled, 10 to 999 seconds. Software level = 4.2+.*
  When calls are queued for a hunt group, higher priority calls are placed ahead of lower priority calls, with calls of the same priority sort by time in queue. External calls are assigned a priority (*1-Low*, *2-Medium* or *3-High*) by the Incoming Call Route that routed the call. Internal calls are assigned a priority of *1-Low*. This option can be used to increase the priority of a call each time it has remained queued for longer than this value. The calls priority is increased by 1 each time until it reaches 3-High.

  - In situations where calls are queued, high priority calls are placed before calls of a lower priority. This has a number of effects:

    - Mixing calls of different priority is not recommended for destinations where Voicemail Pro is being used to provided queue ETA and queue position messages to callers since those values will no longer be accurate when a higher priority call is placed into the queue. Note also that Voicemail Pro will not allow a value already announced to an existing caller to increase.

    - If the addition of a higher priority call causes the queue length to exceed the hunt group's Queue Length Limit 316, the limit is temporarily raised by 1. This means that calls already queued are not rerouted by the addition of a higher priority call into the queue.

- Companding LAW
  These settings should not normally be changed for their defaults. They should only be used where 4400 Series phones (U-Law) are installed on systems which have A-Law digital trunks. Note that U-Law is also called Mu-Law or µ-Law.

  - A-Law or U-Law
    PCM (Pulse Code Modulation) is a method for encoding voice as data. In telephony, two methods of PCM encoding are widely used, A-Law and U-Law (also called Mu-Law or µ-Law). Typically U-Law is used in North America and a few other locations while A-Law is used elsewhere. As well as setting the correct PCM encoding for the region, the A-Law or U-Law setting of a system when it is first started affects a wide range of regional defaults relating to line settings and other values.

    - For IP400 systems, each control units was manufactured as either an A-Law variant or a U-Law variant.

    - For IP500 and IP500v2 systems, the encoding default is set by the type of Feature Key installed when the system is first started. The cards are either specifically A-Law or U-Law. PARTNER Version cards are U-Law. Norstar Version cards are A-Law.

- Default Currency: *Default = Locale specific 844. Software level = 4.0+.*
  This setting is used with ISDN Advice of Charge (AOC) services. Note that changing the currency clears all call costs stored by the system except those already logged through SMDR 898. The currency is displayed in Phone Manager Pro and included in the system SMDR output.

- DSS Status: *Default = Off*
  This setting affects Avaya display phones with programmable buttons. It controls whether pressing a DSS key set to another user who has a call ringing will display details of the caller. When off, no caller information is displayed.

- Auto Hold: *Default = On. Software level = 3.0+.*
  Used for users with multiple appearance buttons. When on, if a user presses another appearance button during a call, their current call is placed on hold. When off, if a users presses another appearance button during a call, their current call is disconnected.

- Dial By Name: *Default = On*
  When on, allows the directory features on various phones to match the dialing of full names. When off, the directory features use first character match only. See User Directory Access 729. For IP Office Release 5+ this option is fixed as *On* and is not adjustable.

- Show Account Code: *Default = On*
  This setting controls the display and listing of system account codes:
  - When on:

    - When entering account codes through Phone Manager, users can select from a drop-down list of available account codes.

    - When entering account codes through a phone, the account code digits are shown while being dialed.

  - When off:

    - Within Phone Manager the drop-down list of available account codes is not useable, instead account codes must be entered using the Phone Managers PIN Code features.

    - When entering account codes through a phone, the account code digits are replaced by s characters on the display.

- **Allow Outgoing Transfer**: *Default = Off. Software level = 3.0 to 3.2.*
  When not enabled, users are only able to transfer or forward off-switch incoming external calls. When enabled, users can forward both incoming and outgoing external calls. For pre-3.0 system the default behaviour is to bar outgoing transfers. For IP Office 4.0 and higher the default behaviour is to always allow outgoing transfers.

- **Inhibit Off-Switch Forward/Transfer**: *Default = Off (Italy = On)*
  When enabled, this setting stops any user from transferring or forwarding calls externally. See Off-Switch Transfer Restrictions 782.

- **Restrict Network Interconnect**: *Default = Off. Software Level = 4.2+.*
  When this option is enabled, each trunk is provided with a **Network Type** option that can be configured as either *Public* or *Private*. The system will not allow calls on a public trunk to be connected to a private trunk and vice versa, returning number unobtainable indication instead.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.

- **WAN Mode Override**: *Default = Off. Software level = 3.2/4.0 Q2 2007 Maintenance Release.*
  Alters the configuration of the WAN interface from the default to that required for BT X25 link inter-working. Used with IP406 V2 systems only.

- **Automatic Codec Preference**: *Default = G.729(a) 8K CS-ACELP. Software level = 5.0+.*
  This setting is used by all IP lines and extensions where the **Compression Mode** has been set to *Automatic Select*. The codec selected here will be the first codec the line or extension uses codec negotiation. Note that the G.711 codecs are treated as a pair, so if one is selected as the first preference, the other will be second in the list.

Setting	Preference Order
G.729	*G729(a) 8K CS-ACELP / G711 U-Law 64K / G711 A-Law 64K / G723.1 6K3 MP-MLQ.*
G.723	*G723.1 6K3 MP-MLQ / G729(a) 8K CS-ACELP / G711 U-Law 64K / G711 A-Law 64K.*
G.711 U-Law	*G711 U-Law 64K / G711 A-Law 64K / G729(a) 8K CS-ACELP / G723.1 6K3 MP-MLQ.*
G.711 A-Law	*G711 A-Law 64K / G711 U-Law 64K / G729(a) 8K CS-ACELP / G723.1 6K3 MP-MLQ.*

- **Drop External Only Impromptu Conference**: *Default = Off. Software level = 5.0+.*
  If selected, when the last remaining internal user in a conference exits the conference, the conference is ended, regardless of whether it contains any external callers. The **Inhibit Off-Switch Forward/Transfer** option above is no longer applied to conference calls. If not selected, the conference is automatically end when the last internal party or trunk that supports reliable disconnect exits the conference.

- **Visually Differentiate External Call**: *Default = Off. Software level = 5.0+.*
  This setting is applied to the lamp flashing rate used for bridged appearance and call coverage appearance buttons on 1400, 1600 and 9600 Series phones and on SMB24, BM32 and DBM32 button modules. When selected, external calls alerting on those buttons will use a slow flash (200ms on/50ms off). If not selected or if the call is internal, normal flashing (500ms on/500ms off) is used.

5.3.6.2 Tones and Music

System | Telephony | Tones and Music	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.1+.
Mergeable	Pre-3.2 ✗, 3.2+ ✓*

*Changes to Busy Tone Detection and the hold music System Source requires a reboot. Deleting any of the hold music Alternate Sources also requires a reboot.

- Conferencing Tone: *Default = Entry & Exit Tones.*
  This settings controls how conference tones are used. It can be set to either *Entry & Exit Tones* or *Repeating Tone*. With *Entry & Exit Tones* a single tone is heard when a new party joins a conference and double-tone is heard when a party leaves the conference. With *Repeating Tone* a conference tone is heard every 10 seconds by all conference parties.

- Disconnect Tone: *Default = Default (Use locale setting 846). Software level = 4.0+.*
  For digital and IP phones, when the system detects that the far end of a call has disconnected, it can make the near end either go idle or play disconnect tone. By default the chosen behaviour depends on the system locale 844. This field can be used to override the locale's default action and force either disconnect tone or go idle.

  - Default: Use the system locale specific action for disconnected calls.
  - On: Play disconnect tone when far end disconnection is detected.
  - Off: Go idle when far end disconnection is detected.

- Local Dial Tone: *Default = On*
  For all normal operation this setting should be left enabled as it allows the system to provide dial tone to users (essential for MSN working).

- Local Busy Tone: *Default = Off*
  This setting should only be used when the local exchange gives a busy signal via Q.931 but does not provide busy tone.

- Beep on Listen: *Default = On (USA)/On (Rest of World)*
  This setting controls whether call parties hear a repeating tone when their call is monitored by another party using the Call Listen 443 feature.

- GSM Silence Suppression: *Default = Off. Software level = 3.0+.*
  This setting should only be selected if voice quality problems are experienced with calls to voicemail or while recording calls. When on, the system signals silence by generating silence data packets in periods when the voicemail system is not playing prompts. Note that use of this option may cause some timeout routing options in voicemail to no longer work.

- Busy Tone Detection: *Default = System Frequency (Tone defined by system locale)*
  Allows configuration of the system's busy tone detection settings. These are on lines that do not provide reliable disconnect signalling. In that case, the system will uses tone disconnect clearing to disconnect such lines after 6 seconds of continuous tone. The default tone (frequency and on/off cadence) detection used is defined by the system locale 844. The settings should not be adjusted unless advised by Avaya Technical Support. Changes to this setting require a reboot rather than a merge when the new configuration is sent to the system.

- Hold Music
  This section is used to define the source for the system's music on hold source. You must ensure that any MOH source you use complies with copyright, performing rights and other local and national legal requirements. For full details of music on hold configuration, see Music on Hold 763.

  - System Source: *Default = WAV File.*
    Selects the default hold music source for most uses of music on hold. Changes to this setting require a reboot rather than a merge when the new configuration is sent to the system.

    - *WAV File:*
      Use the WAV file *holdmusic.wav*. This file is loaded via TFTP.
    - *External: Software level = 3.1+.*
      Use the audio source connected to the back of the control unit. This causes the system to not load the *holdmusic.wav* file via TFTP when rebooted. Previous the presence of a *holdmusic.wav* file loaded onto the system automatically overrode the use of the control units external source.
    - *Tone: Software level = 4.2+.*
      The use of a double beep tone (425Hz, 02./0.2/0.2/3.4 seconds on/off) can be selected as the system source.

      - IP Office 4.0+: The hold music tone is automatically used if the system source is set to WAV File but the *holdmusic.wav* file has not been successfully downloaded.

* **Alternate Sources:** *Software level = 4.2+ (Not Small Office Edition).*
  Up to three additional hold music source can be specified. These are wav files loaded and stored in the same way as *holdmusic.wav*. Note that adding and changing a source can be done using a merge but deleting a source requires a reboot.

  * **Number**
    Assigned automatically by the system. The alternate sources are numbered 2, 3 and 4.

  * **Name:** *Up to 31 characters*
    This field is used to associate a name with the alternate source. That name is then used in the **Hold Music Source** field on Incoming Call Routes and Hunt Groups.

  * **Source:** *Up to 31 characters*
    Defines the source for the music on hold.

    * For a wav file, enter WAV: followed by the file name. For example, for the file *holdmusic2.wav*, enter *WAV:holdmusic2.wav*. The system will automatically attempt to load that file via TFTP following a reboot.

    * IP Office Release 6.1+: Any analog extension with its Equipment Classification 254 set as *MOH Source* can be entered as the alternate source. Enter XTN: followed by the extension's Base Extension number. For example *XTN:224*.

5.3.6.3 Call Log

IP Office 5.0+: The system can store a centralized call log 745 for users. Each users' centralized call log can contain up to 30 call records for user calls (10 on IP412 and IP406 V2 systems). When this limit is reached, new call records replace the oldest record.

On 1400, 1600 and 9600 Series phones with a Call Log or History button, that button can be used to display the user's centralized call log. The user can use the call log to make calls or to store as a personal speed dial. They can also edit the call log to remove records. The same call log is also if the user logs into one-X Portal for IP Office application.

The centralized call log moves with the user if they log on and off from different phones. This includes if they hot desk within a Small Community Network.

System | Telephony | Call Log	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✓

- Default Centralized Call Log On: *Default = On.*
  When selected, each user is defaulted to have the system store a call log of their calls. This call log will be displayed on the phone when the user is using a 1400, 1600 or 9600 Series phone. The use of centralized call logging can be enabled/disabled on a per user basis using the Centralized Call Log 280 user setting (User | Telephony | Call Log 280).

- Logged Missed Calls Answered at Coverage: *Default = Off.*
  This setting controls how calls to a user, that are answered by a covering user should be logged in the centralized call log. This option applies for calls answered elsewhere (covered) by pickup, call coverage (call coverage 686 buttons or coverage group 748), bridged appearance button, user BLF, voicemail, etc.

Setting	Targeted User	Covering User
Off (Default)	Nothing	Answered Call
On	Missed Call	Answered Call

- Log Missed Hunt Group Calls: *Default = Off.*
  By default, hunt group calls are not included in any user's centralized call log unless answered by the user. If this option is selected, a separate call log is kept for each hunt group of calls that are not answered by anyone. It includes hunt group calls that go to voicemail.

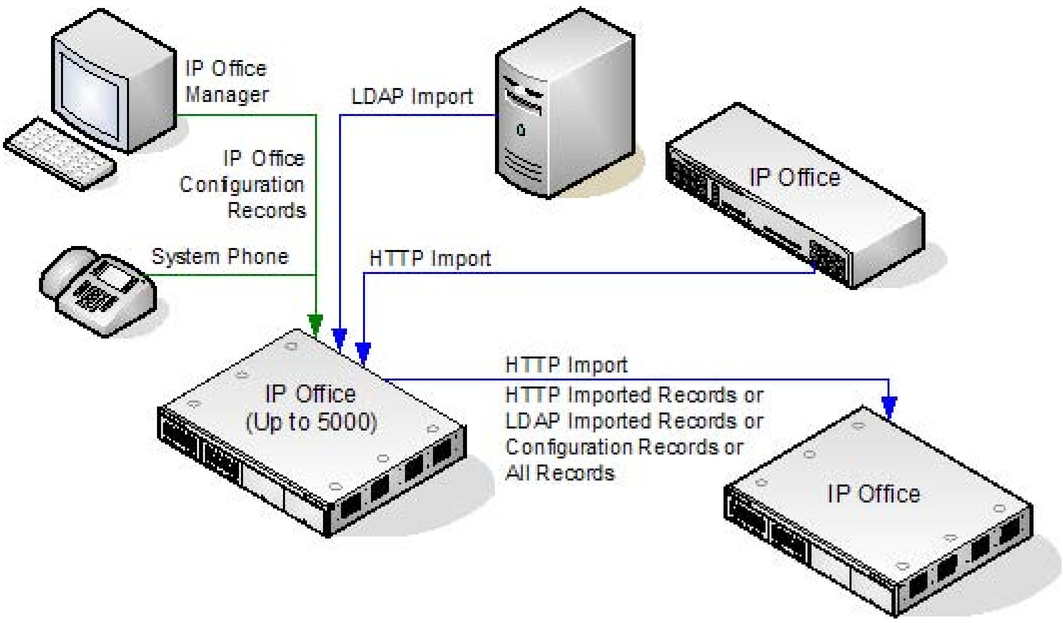
  - If missed hunt group calls are also being logged, the system stores up to 10 call records for each hunt group. When this limit is reached, new call records replace the oldest record.
  - Within the user call log setting (User | Telephony | Call Log 280) the list of hunt groups allows selection of which hunt groups' missed call records should be displayed as part of the centralized call log when a user is using a 1608 or 1616 phone.

5.3.7 Directory Services

Directory services can be used to import directory entries (names and numbers) from external sources. These sets of entries are regularly re-imported.

For systems, the directory records can come from the following sources:

- LDAP Import 169
  The system can import up to 5000 LDAP records for use within directories shown by user phones and applications. LDAP import is configured through the System | Directory Services | LDAP 169 form. For pre-IP Office Release 5.0+ system, LDAP was restricted to 500 records and only displayed within user applications.
- HTTP Import 171 (IP Office Release 5.0+)
  Systems are able to import the directory entries from another system using HTTP. HTTP import is configured through the System | Directory Services | HTTP 171 form by specifying an IP address (or SCN connection). The records imported can be any or all of the following record types held by the system from which the records are being imported: LDAP imported entries, HTTP imported entries, configuration entries.
- System Directory Records 358 *(Configuration entries)*
  Up to 2500 records can be entered directly into the system configuration through the Directory 358 menu. System directory records override matching LDAP/HTTP imported records. For pre-IP Office Release 5.0+ system, the number of system directory records was 1000.
  - IP Office Release 5+: 1400, 1600 and 9600 Series phones with a CONTACTS button and System Phone 765 privileges, can add, delete and edit the system directory records of the system at which they are logged in. They cannot edit LDAP or HTTP imported entries.



System	Number of Directory Records			Total Number of Directory Records
	Configuration	LDAP Import	HTTP Import	
IP500 / IP500v2	2500	5000	5000	5000
IP412	2500	2500	2500	2500
IP406 V2	2500	2500	2500	2500

Directory entries are used for two types of function:

- **Directory Dialing**
  Directory numbers are displayed by user applications such as Phone Manager and SoftConsole. Directory numbers are viewable through the Dir 615 function on many Avaya phones (**Contacts** or **History**). They allow the user to select the number to dial by name. The directory will also contain the names and numbers of users and hunt groups on the system.
  - The Dir function groups directory entries shown to the phone user into the following categories. Depending on the phone, the user may be able to select the category currently displayed. In some scenarios, the categories displayed may be limited to those supported for the function being performed by the user:
    - **External**
      Directory entries from the system configuration. **IP Office 5.0+:** This includes HTTP and LDAP imported entries.
    - **Groups**
      Groups on the system. If the system is in a Small Community Network it will also include groups on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - **Users or Index**
      Users on the system. If the system is in a Small Community Network it will also include users on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - **Personal**
      Available for T3 phones, T3 IP phones, 1400, 1600 and 9600 Series phones. These are the user's personal directory entries stored within the system configuration.
- **Name Matching**
  Directory entries are also used to associate a name with the dialled number on outgoing calls or the received CLI on incoming calls. When name matching is being done, a match in the users personal directory overrides any match in the system directory. Note that some user applications also have their own user directory.
  - The IP Office Phone Manager and SoftConsole applications have their own user directories which are also used by the applications name matching. Matches in the application directory may lead to the application displaying a different name from that shown on the phone.
  - Name matching is not performed when a name is supplied with the incoming call, for example QSIG trunks.
  - Directory name matching is not supported for DECT handsets.

The following characters *(IP Office Release 5.0+*) are supported in directory entries. They are supported in both system configuration entries and in HTTP/LDAP imported entries.

- **? = Any Digit**
  Directory entries containing a ? are only used for name matching against the dialed or received digits on outgoing or incoming. They are not included in the directory of numbers to dial available to users through their phones or applications. The wildcard can be used in any position but typically would be used at the end of the number.
  - In the following example, any calls where the dialed or received number is 10 digits long and starts 732555 will have the display name Homdel associated with them.
    - **Name:** Holmdel
    - **Number:** 9732555????
- **(and) brackets = Optional Digits**
  These brackets are frequently used to enclose an optional portion of a number, typically the area code. Only one pair of brackets are supported in a number. Entries containing digits inside ( ) brackets are used for both name matching or user dialling. When used for name matching, the dialed or received digits are compared to the directory number with and without the ( ) enclosed digits. When used for dialling from a phone or application directory, the full string is dialed with the ( ) brackets removed.
  - The following example is a local number. When dialed by users they are likely to dial just the local number. However on incoming calls, for the CLI the telephony provider includes the full area code. Using the ( ) to enclose the area code digits, it is possible for the single directory entry to be used for both incoming and outgoing calls.
    - **Name:** Raj Garden
    - **Number:** 9(01707)373386
- **Space and - Characters**
  Directory entries can also contain spaces and - characters. These will be ignored during name matching and dialing from the directory.

Imported Records

- Imported directory records are temporary until the next import refresh. They are not added to the system's configuration.
- They cannot be viewed or edited using Manager or edited by a System Phone user.
- The temporary records are lost if the system is restarted. However the system will request a new set of imported directory records after a system restart.
- The temporary records are lost if a configuration containing Directory changes is merged. The system will then import a new set of temporary records without waiting for the *Resync Interval*.
- If an configuration record is edited by a System Phone user to match the name or number of a temporary record, the matching temporary record is discarded.

Importation Rules

When a set of directory records is imported by HTTP or LDAP, the following rules are applied to the new records:

- Imported records with a blank name or number are discarded.
- Imported records that match the name or number of any existing record are discarded.
- When the total number of directory records has reached the system limit, any further imported records are discarded.

System	Number of Directory Records			Total Number of Directory Records
	Configuration	LDAP Import	HTTP Import	
IP500 / IP500v2	2500	5000	5000	5000
IP412	2500	2500	2500	2500
IP406 V2	2500	2500	2500	2500

5.3.7.1 LDAP

LDAP (Lightweight Directory Access Protocol) is a software protocol for enabling anyone to locate organizations, individuals, and other resources such as files and devices in a network, whether on the Internet or on a corporate intranet. LDAP is a "lightweight" (smaller amount of code) version of DAP (Directory Access Protocol), which is part of X.500, a standard for directory services in a network. LDAP is lighter because in its initial version, it did not include security features.

- IP Office 5.0+: The system supports the import of directory records from one system to another using HTTP. That includes using HTTP to import records that another system has learnt using LDAP. HTTP import 171, which is simpler to configure, can be used to relay LDAP records with LDAP configured on just one system.
- LDAP records can contain several telephone numbers. Each will be treated as a separate directory entry when imported into the system directory.
- The NoUser source number setting ExtendLDAPDirectLimit usable with IP Office 4.1-4.2 systems is no longer supported for IP Office 5+.

In a network, a directory tells you where in the network something is located. On TCP/IP networks, including the Internet, the Domain Name System (DNS) is the directory system used to relate the domain name to a specific network address. However, you may not know the domain name. LDAP allows you to search for an individual without knowing where they're located (although additional information will help with the search).

An LDAP directory is organized in a simple "tree" hierarchy consisting of the following levels:

- The "root" directory (the starting place or the source of the tree), which branches out to
- Countries, each of which branches out to
- Organizations, which branch out to
- Organizational units (divisions, departments, and so forth), which branches out to (includes an entry for)
- Individuals (which includes people, files, and shared resources such as printers)

An LDAP directory can be distributed among many servers. Each server can have a replicated version of the total directory that is synchronized periodically. An LDAP server is called a Directory System Agent (DSA). An LDAP server that receives a request from a user takes responsibility for the request, passing it to other DSA's as necessary, but ensuring a single coordinated response for the user.

LDAP Directory Synchronization allows the telephone number Directory held in the Control Unit to be synchronized with the information on an LDAP server.The feature can be configured to interoperate with any server that supports LDAP version 2 or higher.

System | Directory Services | LDAP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- LDAP Enabled: *Default = Off*
  This option turns LDAP support on or off.
- User Name: *Default = blank*
  Enter the user name to authenticate connection with the LDAP database. To determine the domain-name of a particular Windows 2000 user look on the "Account" tab of the user's properties under "Active Directory Users and Computers". Note that this means that the user name required is not necessarily the same as the name of the Active Directory entry. There should be a built-in account in Active Directory for anonymous Internet access, with prefix "IUSR_" and suffix server_name (whatever was chosen at the Windows 2000 installation). Thus, for example, the user name entered is this field might be: IUSR_CORPSERV@example.com
- Password: *Default = blank*
  Enter the password to be used to authenticate connection with the LDAP database. Enter the password that has been configured under Active Directory for the above user. Alternatively an Active Directory object may be made available for anonymous read access. This is configured on the server as follows:
  - In "Active Directory Users and Computers" enable "Advanced Features" under the "View" menu. Open the properties of the object to be published and select the "Security" tab. Click "Add" and select "ANONYMOUS LOGON", click "Add", click "OK", click "Advanced" and select "ANONYMOUS LOGON", click "View/Edit", change "Apply onto" to "This object and all child objects", click "OK", "OK", "OK".

Once this has been done on the server, any entry can be made in the User Name field in the System configuration form (however this field cannot be left blank) and the Password field left blank. Other non-Active Directory LDAP servers may allow totally anonymous access, in which case neither User Name nor Password need be configured.

- Server IP Address: *Default = blank*
  Enter the IP address of the server storing the database.

- Server Port: *Default = 389*
  This setting is used to indicate the listening port on the LDAP server.

- Authentication Method: *Default = Simple*
  Select the authentication method to be used.

  - *Simple:* clear text authentication

  - *Kerberos:* Kerberos 4 LDAP and Kerberos 4 DSA encrypted authentication (for future use).

- Resync Interval (secs): *Default = 3600 seconds, Range = 1 to 99999 seconds.*
  The frequency at which the system should resynchronize the directory with the server. This value also affects some aspects of the internal operation.

  - The LDAP search inquiry contains a field specifying a time limit for the search operation and this is set to 1/16th of the resync interval. So by default a server should terminate a search request if it has not completed within 225 seconds (3600/16).

  - The client end will terminate the LDAP operation if the TCP connection has been up for more than 1/8th of the resync interval (default 450 seconds). This time is also the interval at which a change in state of the "LDAP Enabled" configuration item is checked.

- Search Base / Search Filter: *Default = blank*
  These 2 fields are used together to refine the extraction of directory entries. Basically the Base specifies the point in the tree to start searching and the Filter specifies which objects under the base are of interest. The search base is a distinguished name in string form (as defined in RFC1779).

  The Filter deals with the attributes of the objects found under the Base and has its format defined in RFC2254 (except that extensible matching is not supported).

  If the Search Filter field is left blank the filter defaults to "(objectClass=*)", this will match all objects under the Search Base.

  The following are some examples applicable to an Active Directory database:

  - To get all the user phone numbers in a domain:

    ```
    Search Base: cn=users,dc=acme,dc=com
    Search Filter: (telephonenumber=*)
    ```

  - To restrict the search to a particular Organizational Unit (eg office) and get cell phone numbers also:

    ```
    Search Base: ou=holmdel,ou=nj,DC=acme,DC=com
    Search Filter: (|(telephonenumber=*)(mobile=*))
    ```

  - To get the members of distribution list "group1":

    ```
    Search Base: cn=users,dc=acme,dc=com
    Search Filter: (&(memberof=cn=group1,cn=users,dc=acme,dc=com)(telephonenumber=*))
    ```

- Number Attributes: *Default = see below*
  Enter the number attributes the server should return for each entry that matches the Search Base and Search Filter. Other entries could be ipPhone, otherIpPhone, facsimileTelephoneNumber, otherfacsimileTelephone Number, pager or otherPager. The attribute names are not case sensitive. Other LDAP servers may use different attributes.

  - By default the entry is "telephoneNumber,otherTelephone,homePhone=H,otherHomePhone=H,mobile=M, otherMobile=M", as used by Windows 2000 Server Active Directory for Contacts.

  - The optional "=string" sub-fields define how that type of number is tagged in the directory. Thus, for example, a cell phone number would appear in the directory as: John Birbeck M 7325551234

5.3.7.2 HTTP

IP Office 5+: The system can use HTTP to import the directory records held by another system.

Note that support for HTTP can be disabled. The Avaya HTTP Clients Only 143 setting (System | System 143) can restrict a system from responding to HTTP requests. The system's Unsecured Interface 90 security settings also included controls for HTTP access (HTTP Directory Read and HTTP Directory Write).

System | Directory Services | HTTP	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✗.

- Directory Type: *Default = None (No HTTP import)*
  Set whether HTTP import should be used and the method of importation.
  - *None*
    Do not use HTTP import.
  - *IP Office*
    Import from the system at the IP address set in the Source field.
  - *IP Office SCN*
    Import from a system in a Small Community network. The Source field is used to select the Outgoing Line ID that matches the H323 SCN line to the remote system.
- Source: *Default = Blank*
  The form of this field changes according to the Directory Type selection above.
- List: *Default = All*
  This field sets what types of directory record should be imported.
  - *All*
    Import the full set of directory records from the remote system.
  - *Config Only*
    Import just directory records that are part of the remote system's configuration. Note that these will be treated as imported records and will not be added to the local systems own configuration records.
  - *LDAP Only*
    Import just directory records that the remote system has obtained as the result of its own LDAP import. This allows LDAP directory records to be relayed from one system to another.
  - *HTTP Only*
    Import just directory records that the remote system has obtained as the result of its own HTTP import. This allows HTTP directory records to be relayed from one system to another.
- URI: *Default = /system/dir/complete_dir_list*
  This field is for information only and cannot be adjusted. The path shown changes to match the List setting above.
- Resync Interval (secs): *Default = 3600 seconds.*
  Set how often the system should request an updated import. When a new import is received, all previously imported records are discarded and the newly imported records are processed.

5.3.8 System Events

The system supports a number of methods by which events occurring on the system can be reported. These are in addition to the real-time and historical reports available through the System Status Application (SSA).

- SNMP Reporting *(IP Office 2.1+)*
  Simple Network Management Protocol (SNMP) allows SNMP clients and servers to exchange information. SNMP clients are built into devices such as network routers, server PC's, etc. SNMP servers are typically PC application which receive and/or request SNMP information. The system SNMP client allows the system to respond to SNMP polling and to send alarm information to SNMP servers.
  - In order for an SNMP server application to interact with a system, the IP Office MIB files, provided on the IP Office Admin Suite DVD, must be compiled into the SNMP server's applications database. For full details, refer to the Manager Installation Manual.
- SMTP Email Reporting *(IP Office 3.2+)*
  The system can send alarms to an SMTP email server. Using SMTP requires details of a valid SMTP email account user name and password and server address. If SMTP email alarms are configured but for some reason the system cannot connect with the SMTP server, only the last 10 alarms are stored for sending when connection is successful.
  - Use of SMTP alarms requires the SMTP server details to be entered in the SMTP 177 tab.
- Syslog Reporting *(IP Office 4.1+)*
  The system can also send alarms to a Syslog server (RFC 3164) without needing to configure an SNMP server. In addition Syslog output can include audit trail events.

Multiple event destinations can be created, each specifying which events and alarms to include, the method of reporting to use (SNMP, Syslog or Email) and where to send the events. Up to 2 alarm destinations can be configured for SNMP, 2 for Syslog and 3 for SMTP email.

Enabling SNMP Alarms

1. Select System.
2. Select the System Events tab.
3. Select SNMP Enabled.
4. Complete the information in the SNMP Info section by entering the SNMP port and community details to match those expected by your SNMP server. Details of installing the MIB files required for SNMP are included in the Manager Installation manual.
5. Click OK.

Editing Alarm Destinations
The Alarms section of the System Events tab displays the currently created alarm traps. It shows the event destinations and the types of alarms that will trigger the send of event reports. Up to 2 alarm destinations can be configured for SNMP, 2 for Syslog and 3 for SMTP email.

1. Select the Alarms sub-tab.
2. Use the Add, Remove and Edit controls to alter the traps.
3. Click Add or select the alarm to alter and then click Edit.
4. For a new alarm, set the Destination to either *Trap (SNMP)* or *Syslog* or *Email (SMTP)*. Note that once a destination has been saved by clicking OK it cannot be changed to another sending mode.
5. The remaining details will indicate the required destination information and allow selection of the alarm events to include.
6. When completed, click OK.
7. Click OK again.

5.3.8.1 Configuration

System | System Events | Configuration	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- SNMP Enabled: *Default = Off*
  Enables support for SNMP. This option is not required if using SMTP or Syslog.
  - Community (Read-only): *Default = public*
    The SNMP community name to which the system belongs.
  - SNMP Port: *Default = 161, Range = 0 to 65534.*
    The port on which the system listens for SNMP polling.
  - Device ID
    This is a text field used to add additional information to alarms.
  - Contact
    This is a text field used to add additional information to alarms.
  - Location
    This is a text field used to add additional information to alarms.

- QoS Parameters: *Software level = 5.0+.*
  These parameters are used if Enable RTCP Monitor on Port 5005 150 is selected (Systems | LAN1 | VoIP 150). They are used as alarm thresholds for the QoS data collected by the system for calls made by Avaya H323 phones and for phones using VCM channels. If a monitored call exceeds any of the threshold an alarm is sent to the System Status application. Quality of Service alarms can also be sent from the system using Alarms 174.
    - The alarm occurs at the end of a call. If a call is held or parked and then retrieved, an alarm can occur for each segment of the call that exceeded a threshold.
    - Where a call is between two extensions on the system, it is possible that both extensions will generate an alarm for the call.
    - An alarm will not be triggered for the QoS parameters recorded during the first 5 seconds of a call.
  - Round Trip Delay (msec): *Default = 350.*
    Less than 160ms is high quality. Less than 350ms is good quality. Any higher delay will be noticeable by those involved in the call. Note that, depending on the compression codec being used, some delay stems from the signal processing and cannot be removed: G711 = 40ms, G723a = 160ms, G729 = 80ms.
  - Jitter (msec): *Default =20.*
    Jitter is a measure of the variance in the time for different voice packets in the same call to reach the destination. Excessive jitter will become audible as echo.
  - Packet Loss (%): *Default = 3.0.*
    Excessive packet loss will be audible as clipped words and may also cause call setup delays.

Round Trip Delay	Good Quality	High Quality
Round Trip Delay	< 350ms	< 160ms
Jitter	< 20ms	< 20ms
Packet Loss	< 3%	< 1%

5.3.8.2 Alarms

This form is used to configure what can cause alarms to be sent using the different alarm methods.

- Up to 2 alarm traps can be configured for use with the SNMP settings on the System | System Events | Configuration 173 tab.
- Up to 3 email alarms can be configured for the sending using the systems System | SMTP 177 settings. The email destination is set as part of the alarm configuration below.
- Up to 2 alarms can be configured for sending to a Syslog destination that is included in the alarm settings.

System | System Events | Alarms	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- Events
  This section is used to show and edit the alarm.
  - Destination
    The options are *Trap (SNMP)*, *Syslog* or *Email (SMTP)*. To use SNMP or Email the appropriate settings must be configured on the Configuration sub-tab. Note that the Destination type will also be grayed out if the maximum number of configurable alarms destinations of that type has been reached. Up to 2 alarm destinations can be configured for SNMP, 2 for Syslog and 3 for SMTP email.
    - Trap:
      If selected, the details required in addition to the selected Events are:
      - IP Address: *Default = 0.0.0.0*
        The IP address of the SNMP server to which trap information is sent.
      - Port: *Default = 162, Range = 0 to 65534.*
        The SNMP transmit port
      - Community: *Default = Blank*
        The SNMP community for the transmitted traps. Must be matched by the receiving SNMP server.
    - Email: *Software level = 3.2+.*
      If selected, the details required in addition to the selected Events are:
      - Email:
        The destination email address.
    - Syslog: *Software level = 4.1+.*
      If selected, the details required in addition to the selected Events are:
      - IP Address: *Default = 0.0.0.0*
        The IP address of the Syslog server to which trap information is sent.
      - Port: *Default = 516, Range = 0 to 65534.*
        The Syslog destination port.

- Events: *Default = None*
  Sets which types of system events should be collected and sent. The table below lists the alarms associated with each type of event. Text in italics in the messages is replaced with the appropriate data. Items in [] brackets are included in the message if appropriate. The subject line of SMTP email alarms takes the form "System name: IP address - System Alarm".

Type	Events	Event State	Message
Entity	Application	Voicemail operation	The Voicemail server is now operational.
		Voicemail Failure	The Voicemail server is down.
		Voicemail Event - storage OK	The Voicemail server storage is OK.
		Voicemail Event - storage nearly full	The Voicemail server storage is nearly full.
		Voicemail Event - storage full	The Voicemail server storage is full.
	Compact Flash Card	Change	The PC card in *name* has changed.
	Expansion Module	Operational	Expansion module *name* link is up.
		Failure	Expansion module *name* link is down.
		Error	Expansion module *name* link has a link error.
		Change	Expansion module *name* link has changed.
	Trunk	Operational	Trunk number (name) [on expansion module number] is now operational.
		Failure	Trunk number (name) [on expansion module number] is down.
	VCM	Operational	VCM module *name* is now operational.
		Failure	VCM module *name* has failed.
Memory Card	Invalid Card		
	Free Capacity		
Generic	Generic	Non-primary location boot alarm	System running backup software.
		Invalid SD Card	Incompatible or Invalid *(System or Optional)* SD Card fitted.
		Network link failure	Network Interface *name (ip address)* has been disconnected.
		Network link operational	Network Interface *name (ip address)* has been connected.
		System warm start	System has been restarted (warm start).
		System cold start	System has restarted from power fail (cold start).
		SNMP Invalid community	Invalid community specified in SNMP request.
Licence	Licence Server	Server operational	The license server is now operational.
		Server failure	The license server is no longer operational.
	Licence Key Failure	Licence Key Failure	
Loopback	Loopback	Near end line loopback	Trunk *number (name)* [on expansion module *number*] is in near end loopback.
		Near end payload loopback	Trunk *number (name)* [on expansion module *number*] is in near end loopback with payload.
		Loopback off	Trunk *number (name)* [on expansion module *number*] has no loopback.
Phone Change	Phone Change	Phone has been unplugged	The phone with id *n* has been removed from extension *extension* (*unit*, port *number*).
		Phone has been plugged in	The phone with type *type* (id *number*) has been plugged in for extension *extension* (unit, port *number*).
Quality of Service	QoS Monitoring	If Enable RTCP Monitor on Port 5005 150 is selected, any monitored calls that exceeds the set QoS Parameters 173 will cause an alarm.	

Type	Events	Event State	Message
Syslog	Basic Audit	Events as written to the system Audit Trail. Available on Syslog output only.	
	System Shutdown		
	Running Backup		

- Voicemail Pro Storage Alarms
  The alarm threshold is adjustable through the Voicemail Pro client.
- Embedded Voicemail Storage Alarms
  A disk full alarm is generated when the embedded voicemail memory card reaches 90% full. In addition a critical space alarm is generated at 99% full (98% for the Small Office) and an OK alarm is generated when the disk space returns to below 90% full.
- Loopback
  This type of alarm is only available for systems with a United States locale.

5.3.9 SMTP

- IP Office 3.2+: SMTP can be used as the method of sending system alarms. The email destination is set as part of the email alarms configured in System | System Events | Alarms 174.
- IP Office 4.2+: SMTP can be used with Embedded Voicemail for Voicemail Email 267. The voicemail destination is set by the user's Voicemail Email 267 address.

System | System Events | SMTP Configuration	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✗

- IP Address: *Default = 0.0.0.0*
  This field sets the IP address of the SMTP server being used to forward SNMP alarms sent by email.
- Port: *Default = 25, Range = 0 to 65534.*
  This field set the destination port on the SMTP server.
- Email From Address: *Default = Blank*
  This field set the sender address to be used with mailed alarms. Depending of the authentication requirements of the SMTP server this may need to be a valid email address hosted by that server. Otherwise the SMTP email server may need to be configured to support SMTP relay.
- Server Requires Authentication: *Default = On*
  This field should be selected if the SMTP server being used requires authentication to allow the sending of emails. When selected, the User Name and Password fields become available.
  - User Name: *Default = Blank*
    This field sets the user name to be used for SMTP server authentication.
  - Password: *Default = Blank*
    This field sets the password to be used for SMTP server authentication.
  - Use Challenge Response Authentication (CRAM-MD5): *Default = Off.*
    This field should be selected if the SMTP uses CRAM-MD5.

5.3.10 Twinning

These settings are used with Mobile Twinning, see the User | Mobility 293 tab for further details. The use of mobile twinning requires entry of a Mobility Features license.

- Outgoing CLI Warning
  Changing the outgoing CLI for calls requires the line provider to support that function. You must consult with your line provider before attempting to change the outgoing CLI, failure to do so may result in loss of service. If changing the outgoing CLI is allowed, most line providers required that the outgoing CLI used matches a number valid for return calls on the same trunks. Use of any other number may cause calls to be dropped or the outgoing CLI to be replaced with a valid number.
- On mobile twinned calls, if the original party information is used or a specific calling party information CLI is set, that number overrides setting the outgoing CLI using short codes.

System | Twinning	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- Send Original Party Information for Mobile Twinning: *Default = On*
  When on, the system will attempt to send the ICLID information provided with the incoming call to the twinning destination.
  - IP Office 5.0+: The SIP line Send Caller ID setting takes priority.
  - IP Office 6.0+: The values on the System | Twinning 177 tab override the SIP lines Send Caller ID setting.

* Calling Party Information for Mobile Twinning: *Default = Blank (Disabled), Range = Up to 32 digits.*
  This field is useable when Send Original Party Information for Mobile Twinning is off. Note that the number entered here for use as the CLI must be a valid number for return calls to the same site. Some line providers may reject calls that use a number that is not valid for return calls to the same site. In addition depending on the line type and line provider settings the maximum number of digits may be limited.

5.3.11 SMDR

Using a specified IP address, the system can send a call record for each completed call.

IP Office 4.2+: The system can be configured to send SMDR call log records rather than CDR records. This removes the need to have a the Manager Delta Server application installed on a PC.

IP Office 5+: Support for CDR format records has been dropped.

System | CDR	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.1 to 4.2.
Mergeable	Pre-3.2 ✗, 3.2+ ✓.

- Output: *Default = No Output.*
  Select the type of call record that the Manager should output via IP.
  - *No Output*
  - *CDR Only*: *Software level = 3.2 to 4.2 only.*
    Send call records using the CDR setting below.
  - *SMDR Only*: *Software level = 4.2+.*
    Send call records using the SMDR settings below.

SMDR Settings

These settings are shown if SMDR Only is selected as the Output. For further details refer to Appendix: SMDR 898.

- IP Address: *Default = 0.0.0.0 (Listen).*
  The destination IP address for SMDR records.
  - The address 0.0.0.0 puts the control unit in listen mode on the specified TCP port. When a connection is made on that port, all SMDR records in the buffer are provided.
- TCP Port: *Default = 0.*
  The destination IP port for SMDR records.
- Records to Buffer: *Default = 500, Range = 10 to 3000.*
  The system can cache up to 3000 SMDR records if it detects a communications failure with destination address. If the cache is full, the system will begin discarding the oldest records for each new record. Pre-IP Office 4.2+ the maximum number of records that can be buffered was 1500.
- Call Splitting for Diverts: *Default = Off.*
  When enabled, for calls forwarded off-switch using an external trunk, the SMDR produces separate initial call and forwarded call records. This applies for calls forwarded by forward unconditional, forward on no answer, forward on busy, DND or mobile twinning. It also applies to calls forwarded off-switch by an incoming call route. The two sets of records will have the same Call ID. The call time fields of the forward call record are reset from the moment of forwarding on the external trunk. For full details see SMDR Call Records 898.

CDR Settings

These settings are shown if *CDR Only* is selected as the Output. CDR is not supported for IP Office 5 and higher.

A number of different CDR formats can be selected to match the requirements of the call logging/accounting software being used at the destination address. For further details refer to Appendix: CDR Records 910.

- Enable intra-switch CDRs: *Default = Off.*
  When on, includes CDR records for internal calls.
- Formatting Options
  These fields are used to select the format and type of CDR records required. They must match the records expected by the call logging application receiving the CDR records.
  - Record Format: *Default = Unformatted.*
    Allows selection from a number of common CDR record formats 910.
  - Record Options: *Default = Enhanced.*
    Sets the options to include in the CDR record.
  - Date Format: *Default = Day\Month.*
    Sets the date format used in the CDR records.
- Call Detail Recorder Communications
  This section sets the destination and method for sending of the CDR data.

- IP Address: *Default = 0.0.0.0.*
  The destination IP address for CDR records.

- IP Port: *Default = 0.*
  The destination IP port for CDR records.

- Max CDRs: *Default = 500. Range = 0 to 1500.*
  The system can cache up to 1500 CDR records if it detects a communications failure with destination address. If the cache is full, the system will begin discarding the oldest records for each new record. For system 4.2+ the maximum number of records that can be buffered has been increased to 3000 for all systems other than the Small Office Edition.

- Use UDP: *Default = Off (Use TCP)*
  When selected, this field switches the sending of CDR record packets to use UDP instead of TCP.

  - If *off*, TCP is used. In this mode the system will resend missed or corrupted records using the standard TCP protocol. Records are buffered until successfully sent.

  - If *on*, UDP is used. In this mode the system will not resend missed or corrupt records. Also when using UDP, the system is less likely to detect a communications failure which would triggered record caching.

5.3.12 VCM

This form allows adjustment of the operation of Voice Compression Modules (VCM's) installed in the control unit.

- The options for IP400 VCM controls are supported on IP400 systems from the IP Office 3.2/4.0 Q2 2007 Maintenance Releases onwards.
  - The same controls are supported for IP400 VCM cards in IP500 systems from IP Office 4.1 Q1 2008 Maintenance Release onwards.
- The options for IP500 VCM controls are supported from IP Office 4.2 onwards.

Calls to and from IP devices can require conversion to the audio codec format being used by the IP device. For systems this conversion is done by voice compression channels. These support the common IP audio codecs G711, G723 and G729a. For details of how to add voice compression resources to a system, refer to the Manager Installation Manual.

When are Voice Compression Channels Used

The voice compression channels are used as follows.

- **IP Device to Non-IP Device**
  These calls require a voice compression channel for the duration of the call. If no channel is available, busy indication is returned to the caller.
- **IP Device to IP Device**
  - Call progress tones (for example dial tone, secondary dial tone, etc) do not require voice compression channels with the following exceptions:
    - Short code confirmation, ARS camp on and account code entry tones require a voice compression channel.
    - Devices using G723 require a voice compression channel for all tones except call waiting.
  - When a call is connected:
    - If the IP devices use the same audio codec no voice compression channel is used.
    - If the devices use differing audio codecs, a voice compression channel is required for each.
- **Non-IP Device to Non-IP Device**
  No voice compression channels are required except for Small Office Edition Embedded Voicemail access.
- **Music on Hold**
  This is provided from the system's TDM bus and therefore requires a voice compression channel when played to an IP device.
- **Conference Resources and IP Devices**
  Conferencing resources are managed by the conference chip which is on the system's TDM bus. Therefore, a voice compression channel is required for each IP device involved in a conference. This includes services that use conference resources such as call listen, intrusion, call recording and silent monitoring.
- **Page Calls to IP Device**
  Page calls require 1 voice compression channel per audio codec being used by any IP devices involved. The system only uses G729a for page calls, therefore only requiring one channel but also only supporting pages to G729a capable devices.
- **Voicemail Services and IP Devices**
  Calls to the system voicemail servers are treated as data calls from the TDM bus. Therefore calls from an IP device to voicemail require a voice compression channel.
  - On the Small Office Edition, embedded voicemail uses voice compression channels for audio conversion. Therefore all calls to Small Office Edition embedded voicemail require a voice compression channel and calls from IP devices require two voice compression channels.
- **Fax Calls**
  These are voice calls but with a slightly wider frequency range than spoken voice calls. The system only supports fax across IP between systems with the Fax Transport option selected.
- **SIP Calls**
  - **SIP Line Call to/from Non-IP Devices**
    Voice compression channel required.
  - **Outgoing SIP Line Call from IP Device**
    No voice compression channel required.
  - **Incoming SIP Line Call to IP Device**
    Voice compression channel reserved until call connected.
- **T38 Fax Calls** *(IP Office 5.0+)*
  The system supports T38 fax on SIP trunks and SIP extensions. Each T38 fax call uses a VCM channel.

* Within a Small Community Network, an T38 fax call can be converted to a call across across an H323 SCN lines using the Fax Transport Support protocol. This conversion uses 2 VCM channels.
* In order use T38 Fax connection, the Equipment Classification of an analog extension connected to a fax machine can be set *Fax Machine*. Additionally, a new short code feature Dial Fax is available.

Note: T3 IP devices must be configured to 20ms packet size for the above conditions to apply. If left configured for 10ms packet size, a voice compression channel is needed for all tones and for non-direct media calls.

Measuring Channel Usage

The System Status Application can be used to display voice compression channel usage. Within the Resources section it displays the number of channel in use. It also displays how often there have been insufficient channels available and the last time such an event occurred.

These settings should only be adjusted under the guidance of Avaya support.

System | VCM	
Control Unit	SOE ✗, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+ (Manager 3.2/4.0 Q2 2007 Maintenance Releases).
Mergeable	✓.

Echo
Echoes are typically generated by impedance mismatches when a signal is converted from one circuit type to another, most notably from analog to IP. To resolve this issue, an estimated echo signal can be created from one output and then subtracted from the input to hopefully remove any echo of the output.

- Echo Return Loss (dB): *Default = 6dB. IP400 and IP500 VCM's.*
  This control allows adjustment of expected echo loss that should be used for the echo cancellation process. The options are *0dB*, *3dB*, *6dB* and *9dB*.

Comfort Noise/NLP
A low level of comfort noise is required on digital lines during periods where there would normally be just silence. This is necessary to reassure users that the call is still connected. These controls allow adjustment of the comfort noise generated by the nonlinear processor (NLP) component of the VCM.

- Nonlinear Processor Mode: *Default = Adaptive. IP400 and IP500 VCM's.*
  Allows selection of one of the following options:
  - *Adaptive*
    Adaptive means the comfort noise generated by the NLP will try to match background noise.
  - *Silence*
    Silence means the NLP will not generate comfort noise at all
  - *Disabled*
    Nonlinear processing is not applied, in which case some residual echo may be heard.
- NLP Comfort Noise Attenuation: *Default = -9dB. IP500 VCM's only.*
  Options are *-3dB*, *-6dB* and *-9dB*.
- NLP Comfort Noise Ceiling: *Default =-30dB. IP500 VCM's only.*
  Options are *-30dB* and *-55dB*.

Modem: *Software level = 5.0+. IP500 VCM only.*
For Fax relay, these settings allow adjustment of the TDM side operation applied to fax calls using VCM channels.

- Tx Level (dB): *Default = -9dB, Range = 0 to -13dB.*
- CD Threshold: *Default = -43dB, Options = -26dB, -31dB or -43dB.*
- No Activity Timeout (secs): *Default = 30 seconds, Range = 10 to 600 seconds.*

5.3.13 CCR

Customer Call Reporter (CCR) is an application that collects and displays information on the current status of hunt groups and users that have been configured for Customer Call Reporter operation.

System | CCR	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.2+.
Mergeable	✓.

- Busy Not Available Reason Codes
  Agents who indicate that they are in a 'busy not available' state can be prompt to also indicate the reason for being in that state. This menu allows descriptions for the possible reasons to be entered. The descriptions are then used in menus from which the Agent's make selections when setting themselves into busy not available state and in reports on Agent status.
  - Code/Reason
    Rows 1 to 8 can be used to contain descriptions of up to 31 characters each. Rows 0 and 9 are fixed as *Unsupported* and *Busy Not Available*.
  - For Customer Call Reporter 6.1, the reason codes are used to categorize calls in the Agent Time Card report. Reason 1 is used to define lunch. All other reason codes are reported as breaks.
- Default After Call Work Time (seconds): *Default = 10, Range = 10 to 999 seconds.*
  If an agent goes into the After Call Work (ACW) state, either automatically or manually, this field sets the duration of that state after which it is automatically cleared. This duration can be overridden by the Agent's own setting (User | Telephony | Supervisor Settings | After Call Work Time 276). During ACW state, hunt group calls are not presented to the user.

5.4 Line Settings

The line settings shown in the system configuration will change according to the types of trunk cards installed in the control unit or added using external expansion modules.

- WARNING: Changing Trunk Cards
  Changing the trunk card installed in an control unit will result in line settings for both the previous trunk card and the currently installed trunk card. In order to change the trunk card type in a particular card slot, the configuration must be defaulted. This does not apply if replacing an existing card with one of a higher capacity or fitting a trunk card into a previously unused slot.

Trunk Incoming Call Routing

Each trunk type can be categorized as either an external trunk or internal trunk. The trunk type affects how the system routes calls received on that trunk and the routing of calls to the trunk.

	External Trunks	Internal Trunks
Trunk Types	• Analog trunks • T1 Robbed Bit • E1R2 • ISDN BRI (excluding So) • ISDN PRI T1 • ISDN PRI E1 • SIP	• QSIG (T1, E1 or H323) • BRI So • H323 • SCN • SES
Incoming Calls Routed by	All incoming calls are routed by comparison of call details for matches within the system Incoming Call Routes 342. Line short codes are not used.	Incoming calls are routed by looking for a match to the incoming digits in the following order: • Extension number (including remote numbers in a Small Community Network). • Trunk short codes (excluding ? short code). • System short codes (excluding ? short code). • Trunk ? short code. • System ? short code.

Line Groups

Each system trunk (or in some cases individual trunk channels) can be configured with an Incoming Group ID and an Outgoing Group ID. These group IDs are used as follows:

- Incoming Call Routes 342
  For incoming calls on external trunks, the Incoming Group ID of the trunk is one of the factors used to match the call to one of the configured incoming call routes.

- Short Codes - Routing Outgoing Calls 330
  For dialing which matches a short code set to a Dial feature, the short codes Line Group ID can indicate either an ARS form or to use a trunk from set to the same Outgoing Group ID. If the call is routed to an ARS form, the short codes in the ARS form will specify the trunks to use by matching Outgoing Group ID.

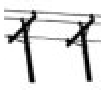
Removing Unused Trunks

In cases where a trunk is not connected, it is important to ensure that the trunk is set as being *Out of Service* within the configuration. This is especially important with analog trunks.

Failure to do this may cause the system to attempt to present outgoing calls to that trunk. Similarly, where the number of channels subscribed is less than those supportable by the trunk type, the unsubscribed channels should be disabled.

- Clock Quality
  Calls between systems using digital trunks (for example E1, E1R2, T1 PRI and BRI) require an common clock signal. The system will try to obtain this clock signal from an exchange through one of its digital trunks. This is done by setting the Clock Quality setting of that Line to Network. If there are multiple trunks to public exchanges, another trunk can be set as Fallback should the primary clock signal fail. Other trunks should be set as Unsuitable.

5.4.1 Analog Line



Analog trunks can be provided within the systems in the following ways. In all cases the physical ports are labeled as Analog. For full details of installation refer to the Manager Installation manual.

- Using ICLID
  The system can route incoming calls using the ICLID received with the call. However ICLID is not sent instantaneously. On analog trunks set to Loop Start ICLID, there will be a short delay while the system waits for any ICLID digits before it can determine where to present the call.
- Line Status
  Analog line do not indicate call status other than whether the line is free or in use. Some system features, for example retrieving unanswered forwards and making twinned calls make use of the call status indicated by digital lines. This is not possible with analog lines. Once an analog line has been seized, the system has to assume that the call is connected and treats it as having been answered.
- Dialing Complete
  The majority of North-American telephony services use en-bloc dialing. Therefore the use of a ; is recommended at the end of all dialing short codes that use an N. This is also recommended for all dialing where secondary dial tone short codes are being used.
- Ground Start
  This type of analog trunk is only supported through the Analog Trunk external expansion module.

5.4.1.1 Line

This tab covers general settings for an analog line.

Line | Line (Analog)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Number
  This parameter is not configurable, it is allocated by the system.

- Card/Module: *Software level = 4.1+.*
  Indicates the card slot or expansion module being used for the trunk device providing the line.

  - For IP400 control units: SLOT A on the control unit is shown as 1, SLOT B is shown as 2. Expansion modules are numbered from 4 upwards, for example trunks on the module in Expansion Port 1 are shown as 4.

  - For IP500 and IP500v2 control units: 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example trunks on the module in Expansion Port 1 are shown as 5.

- Port: *Software level = 4.1+.*
  Indicates the port on the Card/Module above to which the configuration settings relate.

- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.

  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.

- Telephone Number
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.

- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.

- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.

- Outgoing Channels: *Default = 1 (not changeable)*

- Voice Channels: *Default = 1 (not changeable)*

- Prefix: *Default = Blank*
  Enter the number to prefix to all incoming numbers for callback. This is useful if all users must dial a prefix to access an outside line. The prefix is automatically placed in front of all incoming numbers so that users can dial the number back.

  - For outgoing calls: Pre-IP Office 4.0: When a outgoing call is presented to the line with a leading digit to dial that matches the Prefix, that digit is stripped from the number. IP Office 4.0+: The system does not strip the prefix, therefore any prefixes not suitable for external line presentation should be stripped using short codes.

- National Prefix: *Default = 0 (not changeable)*

- Line Appearance ID: *Default = Auto-assigned, Range = 2 to 9 digits. Software level = 3.0+.*
  Allows a number to be assigned to the line to identify it. On phone's that support call appearance buttons, a Line Appearance button with the same number will show the status of the line and can be used to answer calls on the line. The line appearance ID must be unique and not match any extension number.

- Admin: *Default = In Service. Software level = 6.1.*
  This field allows a trunk to be taken out of service if required for maintenance or if the trunk is not connected.

5.4.1.2 Analog Options

This tab covers analog line specific settings.

Line | Analog Options	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- **Channel:**
  Set by the system. Shown for information only.

- **Trunk Type:** *Default = Loop Start*
  Sets the analog line type (*Ground Start*, *Loop Start*, *Loop Start ICLID*).

  - **Ground Start**
    Ground Start is only supported on trunks provided by the Analog Trunk 16 expansion module. It requires that the module and the control unit are grounded. Refer to the Manager installation manual.

  - **Delay Waiting for Caller ID Information.**
    As the system can use ICLID to route incoming calls, on analog Loop Start ICLID trunks there is a few seconds delay while ICLID is received before the call routing can be determined.

- **Signaling Type:** *Default = DTMF Dialing*
  Sets the signaling method used on the line (*DTMF Dialing* or *Pulse Dialing*).

- **Direction:** *Default = Both Directions*
  Sets the allowed direction of operation of the line (*Incoming*, *Outgoing* or *Both Directions*).

- **Flash Pulse Width:** *Default = 500ms, Range = 0 to 2550ms.*
  Set the time interval for the flash pulse width.

- **Await Dial Tone:** *Default = 3000ms, Range = 0 to 25500ms.*
  Sets how long the system should wait before dialing out.

- **Echo Cancellation:** *Default = 16ms.*
  Allows settings of *Off*, *8*, *16*, *32*, *64* and *128* milliseconds. The echo cancellation should only be adjusted as high as required to remove echo problems. Setting it to a higher value than necessary can cause other distortions. Not used with external expansion module trunks.

- **Mains Hum Filter:** *Default = Off. Software level = IP500 4.2+.*
  If mains hum interference on the lines is detected or suspected, this settings can be used to attempt to remove that interference. Useable with ATM16 trunks and IP500 ATM4U trunks. Options are *Off*, *50Hz* or *60Hz*.

- **Impedance**
  Set the impedance used for the line. This field is only available for certain system locales. The range of values available will depend on the system locale. The value used for *Default* is set by the system **Locale**.

  - **Brazil:** *Default = 900R*
    Adjustable between 600R and 900R as required by the line provider.

  - **Korea:** *Default = Default. Software level = 3.2 and 4.0 Q2 2007+.*
    In addition to the default impedance settings, an alternate set of impedance values can be selected.

  - **United States:** *Default = Default. Software level = 3.2 and 4.0 Q2 2007+.*
    In addition to the default impedance setting, the following alternate sets of impedance values *Alternate1*, *Alternate2* and *Alternate3* can be selected.

  - The following values used for **Automatic Impedance Matching**: 600+2150nF, 600, 900+2150nF, 900, 220+820||115nF, 370+620||310nF, 270+750||150nF, 320+1050||230nF, 350+1000||210nF, 800+100||210nF.

- **Quiet Line:** *IP500/IP500v2 only. Default = Off. Software level = 4.2+.*
  This field is only available for certain system locales (see below). The setting may be required to compensate for signal loss on long lines.

- **Automatic Balance Impedance Match:** *IP500/IP500v2. Software level = 4.2+.*
  These controls can be used to test the impedance of a line and to then display the best match resulting from the test. Testing should be performed with the line connected but the system otherwise idle. To start testing click **Start**. The systemwill then send a series of signals to the line and monitor the response, repeating this at each possible impedance setting. Testing can be stopped at any time by clicking **Stop**. When testing is complete, Manager will display the best match and ask whether that match should be used for the line. If *Yes* is selected, Manager will also ask whether the match should be applied to all other analog lines provided by the same analog trunk card or module.

  - IP Office 4.2+: **Automatic Balance Impedance Matching** and **Quiet Line** are only available for North American locales.

- IP Office 6.0+: Automatic Balance Impedance Matching and Quiet Line are available for North American locales and for the Bahrain 851, Egypt 861, Kuwait 872, Morocco 874, Oman 878, Pakistan 879, Qatar 883, Saudi Arabia 885, South Africa 887, Turkey 892, United Arab Emirates 893 and Customize 859 locales.

- Allow Analog Trunk to Trunk Connect: *Default = Not selected (Off).*
  When not enabled, users cannot transfer or forward external calls back off-switch using an analog trunk if the calls was originally made or received on another analog trunk. This prevents transfers to trunks that do not support disconnect clear.

- BCC: *Default = Not selected [Brazil locale only]*
  A collect call is a call at the receiver's expense and by his permission. If supported by the line provider, BCC (Block Collect Call) can be used to bar collect calls.

- Secondary Dial Tone: *Default = Off*
  Configures the use of secondary dial tone on analog lines. This is a different mechanism from secondary dial tone using short codes. This method is used mainly within the Russian locale. When selected, the following additional settings are accessible:

  - Await time: *Default = 3000ms, Range = 0 to 25500ms.*
    Used when secondary dial tone (above) is selected. Sets the delay.

  - After n Digits: *Default = 1, Range = 0 to 10.*
    Sets where in the dialing string, the delay for secondary dial tone, should occur.

  - Matching Digit: *Default =8, Range = 0 to 9.*
    The digit which, when first matched in the dialing string, will cause secondary dial tone delay.

- Long CLI Line: *Default = Off*
  The CLI signal on some analog lines can become degraded and is not then correctly detected. If you are sure that CLI is being provided but not detected, selecting this option may resolve the problem.

- Modem Enabled: *Default = Off*
  The first analog trunk in a control unit can be set to modem operation (V32 with V42 error correction). This allows the trunk to answer incoming modem calls and be used for system maintenance. When on, the trunk can only be used for analog modem calls. The default system short code *9000* can be used to toggle this setting. For the Small Office Edition control unit, when on, the control unit status LED flashes alternate red/green.

- Pulse Dialing
  These settings are used for pulse dialing.

  - Mark: *Default = 80 (80ms), Range = 0 to 255.*
    Interval when DTMF signal is kept active during transmission of DTMF signals.

  - Space: *Default = 80 (80ms), Range = 0 to 255.*
    Interval of silence between DTMF signal transmissions.

  - Inter-Digit Pause: *Default = 500ms, Range = 0 to 2550ms.*
    Sets the pause between digits transmitted to the line.

- Ring Detection

  - Ring Persistency: *Default = Set according to system locale, Range = 0 to 2550ms.*
    The minimum duration of signal required to be recognized.

  - Ring Off Maximum: *Default = Set according to system locale, Range = 0 to 25500ms.*
    The time required before signaling is regarded as ended.

- Disconnect Clear
  Disconnect clear (also known as Line Break or Reliable Disconnect) is a method used to signal from the line provider that the call has cleared. The system also uses Tone Disconnect, which clears an analog call after 6 seconds of continuous busy or NU tone, configured through the Busy Tone Detection 163 (System | Telephony | Tones & Music 163) settings.

  - Enable: *Default = On*
    Enables the use of disconnect clear.

  - Units: *Default = 500ms, Range = 0 to 2550ms.*
    This time must be less than the actual disconnect time period used by the line provider by at least 150ms.

- DTMF
  These settings are used for DTMF dialing.

  - On: *Default = 40ms, Range = 0 to 255ms.*
    The width of the on pulses generated during DTMF dialing.

  - Off: *Default = 60ms, Range = 0 to 255ms.*
    The width of the off pulses generated during DTMF dialing.

- BCC Flash Pulse Width: *[Brazil locale only] Default = 100 (1000ms), Range = 0 to 255.*
  Sets the BCC (Block collect call) flash pulse width.

- Gains:
  These settings are used to adjust the perceived volume on all calls.

- A -> D: *Default = 0dB*
  Sets the analog to digital gain applied to the signal received from the trunk by the system. Range -4.0dB to +3.5dB in 0.5dB steps. To conform with the Receive Objective Loudness Rating at distances greater than 2.7km from the central office, on analog trunks a receive gain of 1.5dB must be set.
- D -> A: *Default = 0dB*
  Sets the digital to analog gain applied to the signal from the system to the trunk. Range -4.0dB to +3.5dB in 0.5dB steps.
- Voice Recording: *Default = Low*
  Used to adjust the volume level of calls recorded by voicemail. Options are *Low*, *Medium* and *High*.

5.4.2 BRI Line

BRI trunks are provided by the installation of a BRI trunk card into the control unit. The cards are available in different variants with either 2 or 4 physical ports. Each port supports 2 B-channels for calls. For full details of installation refer to the Manager Installation manual.

Point-to-Point or Multipoint

BRI lines can be used in either Point-to-Point or Point-to-Multipoint mode. Point-to-Point lines are used when only one device terminates a line in a customer's office. Point-to-Multipoint lines are used when more than one device may be used on the line at the customer's premises. There are major benefits in using Point-to-Point lines:-

1. The exchange knows when the line/terminal equipment is down/dead, thus it will not offer calls down that line. If the lines are Point-to-Multipoint, calls are always offered down the line and fail if there is no response from the terminal equipment. So if you have two Point-to-Multipoint lines and one is faulty 50% of incoming calls fail.
2. You get a green LED on the Control Unit when the line is connected. With Point-to-Multipoint lines some exchanges will drop layer 1/2 signals when the line is idle for a period.
3. The timing clock is locked to the exchange. If layer 1/2 signals disappear on a line then the Control Unit will switch to another line, however this may result in some audible click when the switchover occurs.

The system's default Terminal Equipment Identifier (TEI) will normally allow it to work on Point-to-Point or Point-to-Multipoint lines. However if you intend to connect multiple devices simultaneously to an BRI line, then the TEI should be set to 127. With a TEI of 127, the control unit will ask the exchange to allocate a TEI for operation.

Note: When connected to some manufactures equipment, which provides an S0 interface (BRI), a defaulted Control Unit will not bring up the ISDN line. Configuring the Control Unit to a TEI of 127 for that line will usually resolve this.

5.4.2.1 BRI Line

Settings

Line | BRI Line	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Card/Module: *Software level = 4.1+.*
  Indicates the card slot or expansion module being used for the trunk device providing the line.
  - For IP400 control units: SLOT A on the control unit is shown as 1, SLOT B is shown as 2. Expansion modules are numbered from 4 upwards, for example trunks on the module in Expansion Port 1 are shown as 4.
  - For IP500 and IP500v2 control units: 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example trunks on the module in Expansion Port 1 are shown as 5.
- Port: *Software level = 4.1+.*
  Indicates the port on the Card/Module above to which the configuration settings relate.
- Line Number
  This parameter is not configurable; it is allocated by the system.
- Line Sub Type: *Default = ETSI*
  Select to match the particular line type provided by the line provider.
  - *S-Bus: Software level = 4.2+ (IP500 BRI daughter card only).*
    IP500 BRI daughter cards can be configured for So (S-Bus) operation for connection to ISDN terminal devices. Note that this requires the addition of terminating resistors at both the system and remote ends, and the use of a suitable cross-over cable. For full details refer to the Manager Installation manual.
- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.
- Telephone Number:
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.
- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.
- Prefix: *Default = Blank.*
  The prefix is used in the following ways:
  - For incoming calls
    The ISDN messaging tags indicates the call type (National, International or Unknown). If the call type is unknown, then the number in the Prefix field is added to the ICLID.
  - For outgoing calls
    Pre-IP Office 4.0: When a outgoing call is presented to the line with a leading digit to dial that matches the Prefix, that digit is stripped from the number. IP Office 4.0+: The prefix is not stripped, therefore any prefixes not suitable for external line presentation should be stripped using short codes.
- National Prefix: *Default = 0*
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 1923000000 is converted to 01923000000.
- International Prefix: *Default = 00*
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 00441923000000.
- TEI: *Default = 0*
  The Terminal Equipment Identifier. Used to identify each device connected to a particular ISDN line. For Point-to-Point lines this is 0. It can also be 0 on a Point to Multipoint line, however if multiple devices are sharing a Point-to-Multipoint line it should be set to 127 which results in the exchange allocating the TEI's to be used.

* Number of Channels: *Default = 2. Range = 0 to 2.*
  Defines the number of operational channels that are available on this line.

* Outgoing Channels: *Default = 2. Range = 0 to 2.*
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.

* Voice Channels: *Default = 2. Range = 0 to 2.*
  The number of channels available for voice use.

* Data Channels: *Default = 2. Range = 0 to 2.*
  The number of channels available for data use. If left blank, the value is 0.

* Clock Quality: *Default = Network*
  Refer to the Manager Installation Manual for full details. This option sets whether the system should try to take its clock source for call synchronization and signalling from this line. Preference should always be given to using the clock source from a central office exchange if available by setting at least one exchange line to *Network*.

  * If multiple lines are set as *Network*, the order in which those lines are used is described in the Manager Installation Manual. If additional lines are available, *Fallback* can be used to specify a clock source to use should the *Network* source not be available.

  * Lines from which the clock source should not be taken should be set as *Unsuitable*.

  * If no clock source is available, the system uses its own internal 8KHz clock source.

  * In scenarios where several systems are network via digital trunk lines, care must be taken to ensure that all the systems use the same clock source. The current source being used by a system is reported within the System Status Application.

* Add 'Not-end-to-end ISDN' Information Element: *Default = Never*. Software level = 4.2+.*
  Sets whether the optional 'Not end-to-end ISDN' information element should be added to outgoing calls on the line. The options are *Never*, *Always* or *POTS* (only if the call was originated by an analog extension). *The default is *Never* except for the following locales; for Italy the default is *POTS*, for New Zealand the default is *Always*.

* Supports Partial Rerouting: *Default = Off. Software level = 4.0+.*
  Partial rerouting (PR) is an ISDN feature. It is supported on external (non-network and QSIG) ISDN exchange calls. When an external call is transferred to another external number, the transfer is performed by the ISDN exchange and the channels to the system are freed. Use of this service may need to be requested from the line provider and may incur a charge.

  * Force Number Plan to ISDN: *Default = Off. Software level = 4.2+.*
    This option is only configurable when Support Partial Rerouting is also enabled. When selected, the plan/type parameter for Partial Rerouting is changed from *Unknown/Unknown* to *ISDN/Unknown*. For IP Office 4.0 and 4.1 the plan/type is fixed as *Unknown/Unknown*. The use of this setting will depend on line provider requirements for partial rerouting.

* Send Redirecting Number: *Default = Off. Software level = 6.0+, IP500/IP500v2 only.*
  This option can be used on ISDN trunks where the redirecting service is supported by the trunk provider. Where supported, on twinned calls the caller ID of the original call is passed through to the twinning destination. This option is only used for twinned calls.

* Support Call Tracing: *Default = Off. Software level = 4.0+.*
  The system supports the triggering of malicious caller ID (MCID) tracing at the ISDN exchange. Use of this feature requires liaison with the ISDN service provider and the appropriate legal authorities to whom the call trace will be passed. The user will also need to be enabled for call tracing and be provider with either a short code or programmable button to activate MCID call trace. Refer to Malicious Call Tracing 753 in the Telephone Features section for full details.

* Active CCBS Support: *Default = Off. Software level = 4.0+.*
  Call completion to a busy subscriber (CCBS). It allows automatic callback to be used on outgoing ISDN calls when the destination is busy. This feature can only be used on point-to-point trunks. Use of this service may need to be requested from the line provider and may incur a charge.

* Passive CCBS: *Default = Off. Software level = 4.0+.*

* Cost Per Charging Unit: *Software level = 4.0+.*
  Advice of charge (AOC) information can be display on T3/T3IP phones and output in SMDR 898. The information is provided in the form of charge units. This setting is used to enter the call cost per charging unit set by the line provider. The values are 1/10,000th of a currency unit. For example if the call cost per unit is £1.07, a value of 10700 should be set on the line. Refer to Advice of Charge 740 in the Telephone Features section.

5.4.2.2 Channels

This tab allows settings for individual channels within the trunk to be adjusted. To edit a channel either double-click on it or click the channel and then select Edit.

To edit multiple channels at the same time, select the required channels using Ctrl or Shift and then click Edit. When editing multiple channels, fields that must be unique such as Line Appearance ID are not shown.

Line | Channels (BRI)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Appearance ID: *Default = Auto-assigned, Range = 2 to 9 digits. Software level = 3.0+.*
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number. Line appearance is not supported for trunks set to QSIG operation and is not recommended for trunks be used for DID.

5.4.3 E1 Line

PRI trunks are provided by the installation of a PRI trunk card into the control unit. IP400 PRI cards are available in E1, E1R2 and T1/US PRI variants. The IP500 PRI-U trunk card can be configured (see below) to one of those line types. The cards are also available with either 1 or 2 physical ports. The number of B-channels supported by each physical port depends on the line type of the card.

- E1: 30 B-channels and 1 D-channel per port.
- T1: 24 B-channels per port.
- US PRI: 23 B-channels and 1 D-channel per port.
- E1-R2: 30 B-channels and 1 D-channel per port.

The Small Office Edition control unit only supports a 1 single port IP400 T1 PRI trunk card. For an IP406 V2 control unit, dual port PRI trunk cards are only supported in Slot A. For full details of installation refer to the Manager Installation manual.

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

IP500 PRI-U Trunk Card Line Type

The IP500 PRI-U card can be configured to support either E1, T1 or E1-R2 PRI line types. To select the line type required, right-click on the line in the group or navigation pane and select **Change Universal PRI Card Line Type**.

The IP500/IP500v2 supports 8 B-channels on any IP500 PRI-U card fitted. Additional B-channels up to the full capacity of IP500 PRI-U ports installed require licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, license instances are consumed by the number of calls in progress on B-channels.
- For T1, E1R2 and ETSI CHI trunks, licenses instances are consumed by the channels set as in service.

5.4.3.1 PRI Line

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

Line | PRI Line (E1)	
Control Unit	SOE ✗, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Number
  This parameter is not configurable; it is allocated by the system.
- Line Sub Type:
  Select to match the particular line type provided by the line provider. E1 PRI trunks support *ETSI*, *ETSI CHI*, *QSIG A* or *QSIG B*.
  - *ETSI CHI* is used to send the channel allocation ID (CHI) in the call setup signalling. This is a request to use a particular B-channel rather than use any B-channel allocated by the central office exchange.
  - QSIG trunks and H323 IP trunks are not supported on IP500 and IP500v2 systems without IP500 Voice Networking 833 licenses.
- Card/Module: *Software level = 4.1+.*
  Indicates the card slot or expansion module being used for the trunk device providing the line.
  - For IP400 control units: SLOT A on the control unit is shown as 1, SLOT B is shown as 2. Expansion modules are numbered from 4 upwards, for example trunks on the module in Expansion Port 1 are shown as 4.
  - For IP500 and IP500v2 control units: 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example trunks on the module in Expansion Port 1 are shown as 5.
- Port: *Software level = 4.1+.*
  Indicates the port on the Card/Module above to which the configuration settings relate.
- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.
- Telephone Number:
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.
- Channel Allocation: *Default = 30->1. Software level = 4.2+.*
  For lines set to *ETSI CHI*, this option allows the system to select the default order in which channels should be used for outgoing calls. Typically this is set as the opposite of the default order in which the central office exchange uses channels for incoming calls.
- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.
  - IP Office 4.2+: For lines set to the Line Sub Type of *ETSI CHI*, the Incoming Group ID is set as part of the individual channel 201 settings.
- Prefix: *Default = Blank.*
  The prefix is used in the following ways:
  - For incoming calls
    The ISDN messaging tags indicates the call type (National, International or Unknown). If the call type is unknown, then the number in the Prefix field is added to the ICLID.
  - For outgoing calls
    Pre-IP Office 4.0: When a outgoing call is presented to the line with a leading digit to dial that matches the Prefix, that digit is stripped from the number. IP Office 4.0+: The prefix is not stripped, therefore any prefixes not suitable for external line presentation should be stripped using short codes.

- National Prefix: *Default = 0*
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 1923000000 is converted to 01923000000.

- International Prefix: *Default = 00*
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 00441923000000.

- TEI: *Default = 0*
  The Terminal Equipment Identifier. Used to identify each Control Unit connected to a particular ISDN line. For Point to Point lines this is typically (always) 0. It can also be 0 on a Point to Multi-Point line, however if multiple devices are sharing a Point to Multi-Point line it should be set to 127 which results in the exchange deciding on the TEI's to be used.

- Number of Channels
  Defines the number of operational channels that are available on this line. Up to 30 for E1 PRI, 23 for T1 PRI.

- Outgoing Channels
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls. Only available when the Line Sub Type is set to *ETSI*.

- Voice Channels
  The number of channels available for voice use. Only available when the Line Sub Type is set to *ETSI*.

- Data Channels
  The number of channels available for data use. Only available when the Line Sub Type is set to *ETSI*.

- CRC Checking: *Default = On*
  Switches CRC on or off.

- Line Signalling: *Default = CPE*
  This option is not used for lines where the Line SubType is set to *QSIG*. Select either *CPE* (customer premises equipment) or *CO* (central office). The *CO* feature is intended to be used primarily as a testing aid. It allows PRI lines to be tested in a back-to-back configuration, using crossover cables.

  - The CO feature operates on this line type by modifying the way in which incoming calls are disconnected for system configuration in Brazil and Argentina. In these locales, the CO setting uses Forced-Release instead of Clear-Back to disconnect incoming calls. The Brazilian Double-Seizure mechanism, used to police Collect calls, is also disabled in CO mode.

- Clock Quality: *Default = Network*
  Refer to the Manager Installation Manual for full details. This option sets whether the system should try to take its clock source for call synchronization and signalling from this line. Preference should always be given to using the clock source from a central office exchange if available by setting at least one exchange line to *Network*.

  - If multiple lines are set as *Network*, the order in which those lines are used is described in the Manager Installation Manual. If additional lines are available, *Fallback* can be used to specify a clock source to use should the *Network* source not be available.

  - Lines from which the clock source should not be taken should be set as *Unsuitable*.

  - If no clock source is available, the system uses its own internal 8KHz clock source.

  - In scenarios where several systems are network via digital trunk lines, care must be taken to ensure that all the systems use the same clock source. The current source being used by a system is reported within the System Status Application.

- Add 'Not-end-to-end ISDN' Information Element: *Default = Never*. Software level = 4.2+.*
  Sets whether the optional 'Not end-to-end ISDN' information element should be added to outgoing calls on the line. The options are *Never*, *Always* or *POTS* (only if the call was originated by an analog extension). *The default is *Never* except for the following locales; for Italy the default is *POTS*, for New Zealand the default is *Always*.

- Supports Partial Rerouting: *Default = Off. Software level = 4.0+.*
  Partial rerouting (PR) is an ISDN feature. It is supported on external (non-network and QSIG) ISDN exchange calls. When an external call is transferred to another external number, the transfer is performed by the ISDN exchange and the channels to the system are freed. Use of this service may need to be requested from the line provider and may incur a charge.

  - Force Number Plan to ISDN: *Default = Off. Software level = 4.2+.*
    This option is only configurable when Support Partial Rerouting is also enabled. When selected, the plan/type parameter for Partial Rerouting is changed from *Unknown/Unknown* to *ISDN/Unknown*. For IP Office 4.0 and 4.1 the plan/type is fixed as *Unknown/Unknown*. The use of this setting will depend on line provider requirements for partial rerouting.

- Send Redirecting Number: *Default = Off. Software level = 6.0+, IP500/IP500v2 only.*
  This option can be used on ISDN trunks where the redirecting service is supported by the trunk provider. Where supported, on twinned calls the caller ID of the original call is passed through to the twinning destination. This option is only used for twinned calls.

- Support Call Tracing: *Default = Off. Software level = 4.0+.*
  The system supports the triggering of malicious caller ID (MCID) tracing at the ISDN exchange. Use of this feature requires liaison with the ISDN service provider and the appropriate legal authorities to whom the call trace will be passed. The user will also need to be enabled for call tracing and be provider with either a short code or programmable button to activate MCID call trace. Refer to Malicious Call Tracing 753 in the Telephone Features section for full details.

- Active CCBS Support: *Default = Off. Software level = 4.0+.*
  Call completion to a busy subscriber (CCBS). It allows automatic callback to be used on outgoing ISDN calls when the destination is busy. This feature can only be used on point-to-point trunks. Use of this service may need to be requested from the line provider and may incur a charge.

- Passive CCBS: *Default = Off. Software level = 4.0+.*

- Cost Per Charging Unit: *Software level = 4.0+.*
  Advice of charge (AOC) information can be display on T3/T3IP phones and output in SMDR 898. The information is provided in the form of charge units. This setting is used to enter the call cost per charging unit set by the line provider. The values are 1/10,000th of a currency unit. For example if the call cost per unit is £1.07, a value of 10700 should be set on the line. Refer to Advice of Charge 740 in the Telephone Features section.

- Admin: *Default = In Service. Software level = 6.1.*
  This field allows a trunk to be taken out of service if required for maintenance or if the trunk is not connected.

The following fields are shown for a US T1 trunk card set to ETSI or QSIG operation. These cards have the same settings E1 PRI trunk cards set to ETSI or QSIG but only support 23 channels.

- CSU Operation
  Tick this field to enable the T1 line to respond to loop-back requests from the line.

- Haul Length: *Default = 0-115 feet*
  Sets the line length to a specific distance.

- Channel Unit: *Default = Foreign Exchange*
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are Foreign Exchange, Special Access or Normal.

5.4.3.2 Short Codes

Line short codes can be applied to the digits received with incoming call. The stage at which they are applied also varies depending on the trunk type.

The line Short Code tab is shown for internal trunk types: *QSIG* (T1, E1, H323), *BRI S0*, *H323*, *SCN*, *SES*. Incoming calls are routed by looking for a match to the incoming digits in the following order:

- Extension number (including remote numbers in a Small Community Network).
- Line short codes (excluding ? short code).
- System short codes (excluding ? short code).
- Line ? short code.
- System ? short code.

Line | Short Codes	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

5.4.3.3 Channels

This tab allows settings for individual channels within the trunk to be adjusted. To edit a channel either double-click on it or click the channel and then select Edit.

To edit multiple channels at the same time, select the required channels using Ctrl or Shift and then click Edit. When editing multiple channels, fields that must be unique such as Line Appearance ID are not shown.

Line | Channels (E1 PRI)	
Control Unit	SOE ✗, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Appearance ID: *Default = Auto-assigned, Range = 2 to 9 digits. Software level = 3.0+.*
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number. Line appearance is not supported for trunks set to QSIG operation and is not recommended for trunks be used for DID.
  - IP Office 4.2+: If the trunk Line Sub Type is set to *ETSI CHI*, outgoing line appearance calls must use the correspond channel.

IP Office 4.2+: The following additional fields are shown for lines where the Line Sub Type is set to *ETSI CHI*.

- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.
- Direction: *Default = Bothways*
  Controls the direction of calls allowed on the channel. Options are *Bothways*, *Incoming* or *Outgoing*.
- Bearer: *Default = Any.*
  Allows selection of the type of call that can use the channel. Options are *Any*, *Data* or *Voice*.
- Admin: *Default = Out of Service.*
  This field can be used to indicate whether the channel is in use or not. On trunks where only a limited number of channels have been requested from the trunk provider (known as sub-equipped trunks), those channels not provided should be set as *Out of Service*. For channels that are available but are temporarily not being used select *Maintenance*.
- Tx Gain: *Default = 0dB, Range = -10dBb to +5dB.*
  The transmit gain in dB.
- Rx Gain: *Default = 0dB, Range = -10dBb to +5dB.*
  The receive gain in dB.

5.4.4 E1R2 Line

PRI trunks are provided by the installation of a PRI trunk card into the control unit. IP400 PRI cards are available in E1, E1R2 and T1/US PRI variants. The IP500 PRI-U trunk card can be configured (see below) to one of those line types. The cards are also available with either 1 or 2 physical ports. The number of B-channels supported by each physical port depends on the line type of the card.

- E1: 30 B-channels and 1 D-channel per port.
- T1: 24 B-channels per port.
- US PRI: 23 B-channels and 1 D-channel per port.
- E1-R2: 30 B-channels and 1 D-channel per port.

The Small Office Edition control unit only supports a 1 single port IP400 T1 PRI trunk card. For an IP406 V2 control unit, dual port PRI trunk cards are only supported in Slot A. For full details of installation refer to the Manager Installation manual.

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

IP500 PRI-U Trunk Card Line Type

The IP500 PRI-U card can be configured to support either E1, T1 or E1-R2 PRI line types. To select the line type required, right-click on the line in the group or navigation pane and select **Change Universal PRI Card Line Type**.

The IP500/IP500v2 supports 8 B-channels on any IP500 PRI-U card fitted. Additional B-channels up to the full capacity of IP500 PRI-U ports installed require licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, license instances are consumed by the number of calls in progress on B-channels.
- For T1, E1R2 and ETSI CHI trunks, licenses instances are consumed by the channels set as in service.

5.4.4.1 E1-R2 Options (Line)

Line | Line (E1-R2)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.4+.
Mergeable	✗.

- Card/Module: *Software level = 4.1+.*
  Indicates the card slot or expansion module being used for the trunk device providing the line.
  - For IP400 control units: SLOT A on the control unit is shown as 1, SLOT B is shown as 2. Expansion modules are numbered from 4 upwards, for example trunks on the module in Expansion Port 1 are shown as 4.
  - For IP500 and IP500v2 control units: 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example trunks on the module in Expansion Port 1 are shown as 5.
- Port: *Software level = 4.1+.*
  Indicates the port on the Card/Module above to which the configuration settings relate.
- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restricted Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.
- Line Number:
  Allocated by the system.
- Line SubType: *Default = E1-R2*
  Supported options are *E1-R2*, *ETSI*, *QSIGA* or *QSIGB*.
  - QSIG trunks and H323 IP trunks are not supported on IP500 and IP500v2 systems without IP500 Voice Networking 833 licenses.
- Channel Allocation: *Default = 30 -> 1*
  The order, *30 -> 1* or *1 -> 30*, in which channels are used.
- Country (Locale): *Default = Mexico.*
  Select the locale that matches the area of usage. Note that changing the locale will return the MFC Group settings to the defaults for the selected locale. Currently supported locales *Argentina*, *Brazil*, *China*, *India*, *Korea*, *Mexico* and *None*.
- Admin: *Default = In Service. Software level = 6.1.*
  This field allows a trunk to be taken out of service if required for maintenance or if the trunk is not connected.

The table at the base of the form displays the settings for the individual channels provided by the line. For details of the channel settings see Edit Channel (E1-R2) 204.

To edit a channel, either double-click on it or right-click and select Edit. This will display the Edit Channel 204 dialog box. To edit multiple channels at the same time select the channels whilst pressing the Shift or Ctrl key. Then right-click and select Edit.

5.4.4.2 Channels

This tab allows settings for individual channels within the trunk to be adjusted. To edit a channel, select the required channel or channels and click Edit.

Line | Channels (E1-R2)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.4+.
Mergeable	✗.

The channel settings are split into two sub-tabs, E1R2 Edit Channel and Timers.

E1R2 Edit Channel Settings

- Channel
  The channel or channels being edited.
- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.
- Direction: *Default = Both Directions*
  The direction of calls on the channel (*Incoming*, *Outgoing* or *Both Directions*).
- Bearer: *Default = Any*
  The type of traffic carried by the channel (*Voice*, *Data* or *Any*).
- Line Signaling Type: *Default = R2 Loop Start*
  The signaling type used by the channel. Current supported options are: *R2 Loop Start*, *R2 DID*, *R2 DOD*, *R2 DIOD*, *Tie Immediate Start*, *Tie Wink Start*, *Tie Delay Dial*, *Tie Automatic*, *WAN Service* and *Out of Service*.
- Dial Type: *Default = MFC Dialing*
  The type of dialing supported by the channel; *MFC Dialing*, *Pulse Dialing* or *DTMF Dialing*.

Timers Settings
This sub-tab displays the various timers provided for E1-R2 channels. These should only be adjusted when required to match the line provider's settings.

5.4.4.3 MFC Group

These tabs show the parameter assigned to each signal in an MFC group. The defaults are set according to the Country (Locale) on the Line tab. All the values can be returned to default by the Default All button on the Advanced tab.

To change a setting either double-click on it or right-click and select Edit.

Line | MFC Group (E1-R2)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.4+.
Mergeable	✗.

5.4.4.4 Advanced

Line | Advanced (E1-R2)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.4+.
Mergeable	✗.

- Zero Suppression: *Default = HDB3*
  Selects the method of zero suppression used (HDB3 or AMI).
- Clock Quality: *Default = Network*
  Refer to the Manager Installation Manual for full details. This option sets whether the system should try to take its clock source for call synchronization and signalling from this line. Preference should always be given to using the clock source from a central office exchange if available by setting at least one exchange line to *Network*.
  - If multiple lines are set as *Network*, the order in which those lines are used is described in the Manager Installation Manual. If additional lines are available, *Fallback* can be used to specify a clock source to use should the *Network* source not be available.
  - Lines from which the clock source should not be taken should be set as *Unsuitable*.
  - If no clock source is available, the system uses its own internal 8KHz clock source.
  - In scenarios where several systems are network via digital trunk lines, care must be taken to ensure that all the systems use the same clock source. The current source being used by a system is reported within the System Status Application.
- Pulse Metering Bit: *Default = A Bit*
  Sets which bit should be used to indicate the pulse metering signal; *A Bit*, *B Bit* or *C Bit*.
- Line Signaling: *Default = CPE*
  Select either *CPE* or *CO*. The *CO* feature is intended to be used primarily as a testing aid. It allows T1 and E1 lines to be tested in a back-to-back configuration, using crossover (QSIG) cables.
  - The CO feature operates by modifying the way in which incoming calls are disconnected for system configuration in Brazil and Argentina. In these locales, the CO setting uses Forced-Release instead of Clear-Back to disconnect incoming calls. The Brazilian Double-Seizure mechanism used to police Collect calls, is also disabled in CO mode.
- Incoming Routing Digits: *Default = 4*
  Sets the number of incoming digits used for incoming call routing.
- CRC Checking: *Default = Ticked (On)*
  Switches CRC on or off.
- Default All Group Settings
  Default the MFC Group tab settings.
- Line Signaling Timers:
  To edit one of these timers, either double-click on the timer or right-click on a timer and select the action required.

5.4.5 T1 Line

PRI trunks are provided by the installation of a PRI trunk card into the control unit. IP400 PRI cards are available in E1, E1R2 and T1/US PRI variants. The IP500 PRI-U trunk card can be configured (see below) to one of those line types. The cards are also available with either 1 or 2 physical ports. The number of B-channels supported by each physical port depends on the line type of the card.

- E1: 30 B-channels and 1 D-channel per port.
- T1: 24 B-channels per port.
- US PRI: 23 B-channels and 1 D-channel per port.
- E1-R2: 30 B-channels and 1 D-channel per port.

The Small Office Edition control unit only supports a 1 single port IP400 T1 PRI trunk card. For an IP406 V2 control unit, dual port PRI trunk cards are only supported in Slot A. For full details of installation refer to the Manager Installation manual.

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

IP500 PRI-U Trunk Card Line Type

The IP500 PRI-U card can be configured to support either E1, T1 or E1-R2 PRI line types. To select the line type required, right-click on the line in the group or navigation pane and select **Change Universal PRI Card Line Type**.

The IP500/IP500v2 supports 8 B-channels on any IP500 PRI-U card fitted. Additional B-channels up to the full capacity of IP500 PRI-U ports installed require licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, license instances are consumed by the number of calls in progress on B-channels.
- For T1, E1R2 and ETSI CHI trunks, licenses instances are consumed by the channels set as in service.

- **Dialing Complete**
  The majority of North-American telephony services use en-bloc dialing. Therefore the use of a ; is recommended at the end of all dialing short codes that use an N. This is also recommended for all dialing where secondary dial tone short codes are being used.

5.4.5.1 Line

Line | Line (T1)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Number:
  Allocated by the system.
- Card/Module: *Software level = 4.1+.*
  Indicates the card slot or expansion module being used for the trunk device providing the line.
  - For IP400 control units: SLOT A on the control unit is shown as 1, SLOT B is shown as 2. Expansion modules are numbered from 4 upwards, for example trunks on the module in Expansion Port 1 are shown as 4.
  - For IP500 and IP500v2 control units: 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example trunks on the module in Expansion Port 1 are shown as 5.
- Port: *Software level = 4.1+.*
  Indicates the port on the Card/Module above to which the configuration settings relate.
- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.
- Line Sub Type: *Default = T1*
  Set to T1 for a T1 line. For PRI see Line Form (US PRI) 211. If set to ETSI, QSIG A or QSIG B see Line (E1) 197.
- Channel Allocation: *Default = 24 -> 1*
  The order, 24 to 1 or 1 to 24, in which channels are used.
- Prefix: *Default = Blank*
  Enter the number to prefix to all incoming numbers for callback. This is useful if all users must dial a prefix to access an outside line. The prefix is automatically placed in front of all incoming numbers so that users can dial the number back.
- Framing: *Default = ESF*
  Selects the type of signal framing used (*ESF* or *D4*).
- Zero Suppression: *Default = B8ZS*
  Selects the method of zero suppression used (*B8ZS* or *AMI ZCS*).
- Clock Quality: *Default = Network*
  Refer to the Manager Installation Manual for full details. This option sets whether the system should try to take its clock source for call synchronization and signalling from this line. Preference should always be given to using the clock source from a central office exchange if available by setting at least one exchange line to *Network*.
  - If multiple lines are set as *Network*, the order in which those lines are used is described in the Manager Installation Manual. If additional lines are available, *Fallback* can be used to specify a clock source to use should the *Network* source not be available.
  - Lines from which the clock source should not be taken should be set as *Unsuitable*.
  - If no clock source is available, the system uses its own internal 8KHz clock source.
  - In scenarios where several systems are network via digital trunk lines, care must be taken to ensure that all the systems use the same clock source. The current source being used by a system is reported within the System Status Application.
- Haul Length: *Default = 0-115 feet*
  Sets the line length to a specific distance.
- Channel Unit: *Default = Foreign Exchange*
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are *Foreign Exchange*, *Special Access* or *Normal*.
- CRC Checking: *Default = On*
  Turns CRC on or off.

- Line Signaling: *Default = CPE*
  This field affects T1 channels set to Loop-Start or Ground-Start. The field can be set to either CPE (Customer Premises Equipment) or CO (Central Office). This field should normally be left at its default of CPE. The setting CO is normally only used in lab back-to-back testing.

- Incoming Routing Digits: Default=0 (present call immediately)
  Sets the number of routing digits expected on incoming calls. This allows the line to present the call to the system once the expected digits have been received rather than waiting for the digits timeout to expire. This field only affects T1 line channels set to E&M Tie, E&M DID, E&M Switched 56K and Direct Inward Dial.

- CSU Operation:
  Tick this field to enable the T1 line to respond to loop-back requests from the line.

- Enhanced Called Party Number: *Default = Off*
  This option is not supported for systems set to the United States locale. Normally the dialed number length is limited to 15 digits. Selecting this option increases the allowed dialed number length to 30 digits.

- Admin: *Default = In Service. Software level = 6.1.*
  This field allows a trunk to be taken out of service if required for maintenance or if the trunk is not connected.

5.4.5.2 Channels

The settings for each channel can be edited. Users have the option of editing individual channels by double-clicking on the channel or selecting and editing multiple channels at the same time. Note that the Line Appearance ID cannot be updated when editing multiple channels.

When editing a channel or channels, the settings available are displayed on two sub-tabs; T1 Edit Channel and Timers.

Line | Channels (T1)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

T1 Edit Channel Sub-Tab Settings

- Channel
  Allocated by the system.
- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.
- Line Appearance ID: *Default = Auto-assigned, Range = 2 to 9 digits. Software level = 3.0+.*
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number. Line appearance is not supported for trunks set to QSIG operation and is not recommended for trunks be used for DID.
- Direction: *Default = Bothway*
  The direction of calls on the channel (*Incoming*, *Outgoing* or *Bothway*).
- Bearer: *Default = Any*
  The type of traffic carried by the channel.
- Type: *Default = Out of Service*
  The T1 emulates the following connections (*Ground-Start*, *Loop-Start*, *E&M - TIE*, *E&M - DID*, *E&M Switched 56K*, *Direct Inward Dial*, *Clear Channel 64K* or *Out of Service*). Trunks set to *E&M - DID* will only accept incoming calls.
  - If *E&M - TIE* is selected and the Outgoing Trunk Type is set to *Automatic*, no secondary dial tone is provided for outgoing calls on this line/trunk.
- Dial Type: *Default = DTMF Dial*
  Select the dialing method required (*DTMF Dial* or *Pulse Dial*).
- Incoming Trunk Type: *Default = Wink-Start*
  Used for E&M types only. The handshake method for incoming calls (*Automatic*, *Immediate*, *Delay Dial* or *Wink-Start*).
- Outgoing Trunk Type: *Default = Wink-Start*
  Used for E&M types only. The handshake method for outgoing calls (*Automatic*, *Immediate*, *Delay Dial* or *Wink-Start*).
  - If the line Type is set to *E&M-TIE* and the Outgoing Trunk Type is set to *Automatic*, no secondary dial tone is provided for outgoing calls on this line/trunk.
- Tx Gain: *Default = 0dB*
  The transmit gain in dB.
- Rx Gain: *Default = 0dB*
  The receive gain in dB.

Timers Sub-Tab Settings

This sub-tab allows various timers relating to operation of an individual channel to be adjusted. These should only be adjusted to match the requirements of the line provider. The following is a list of the default values. To reset a value, click on the current value and then right click and select from the default, minimize and maximize options displayed.

- Outgoing Seizure: 10.
- Wink Start: 5000.
- Wink Validated: 80.
- Wink End: 350.
- Delay End: 5000.
- Outgoing Dial Guard: 590.
- Outgoing IMM Dial Guard: 1500.
- Outgoing Pulse Dial Break: 60.
- Outgoing Pulse Dial Make: 40.
- Outgoing Pulse Dial Inter Digit: 720.
- Outgoing Pulse Dial Pause: 1500.
- Flash Hook Generation: 500.
- Outgoing End of Dial: 1000.
- Answer Supervision: 300.
- Incoming Confirm: 20.
- Incoming Automatic Delay: 410.
- Incoming Wink Delay: 100.
- Wink Signal: 200.
- Incoming Dial Guard: 50.
- First Incoming Digit: 15000.
- Incoming Inter Digit: 5000.
- Maximum Inter Digit: 300.
- Flash Hook Detect: 240.
- Incoming Disconnect: 300.
- Incoming Disconnect Guard: 800.
- Disconnected Signal Error: 240000.
- Outgoing Disconnect: 300.
- Outgoing Disconnect Guard: 800.
- Ring Verify Duration: 220.
- Ring Abandon: 6300.
- Ping Verify: 600.
- Long Ring Time: 1100.
- Silent Interval: 1100.

5.4.6 T1 PRI Line

PRI trunks are provided by the installation of a PRI trunk card into the control unit. IP400 PRI cards are available in E1, E1R2 and T1/US PRI variants. The IP500 PRI-U trunk card can be configured (see below) to one of those line types. The cards are also available with either 1 or 2 physical ports. The number of B-channels supported by each physical port depends on the line type of the card.

- E1: 30 B-channels and 1 D-channel per port.
- T1: 24 B-channels per port.
- US PRI: 23 B-channels and 1 D-channel per port.
- E1-R2: 30 B-channels and 1 D-channel per port.

The Small Office Edition control unit only supports a 1 single port IP400 T1 PRI trunk card. For an IP406 V2 control unit, dual port PRI trunk cards are only supported in Slot A. For full details of installation refer to the Manager Installation manual.

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

IP500 PRI-U Trunk Card Line Type

The IP500 PRI-U card can be configured to support either E1, T1 or E1-R2 PRI line types. To select the line type required, right-click on the line in the group or navigation pane and select Change Universal PRI Card Line Type.

The IP500/IP500v2 supports 8 B-channels on any IP500 PRI-U card fitted. Additional B-channels up to the full capacity of IP500 PRI-U ports installed require licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, license instances are consumed by the number of calls in progress on B-channels.
- For T1, E1R2 and ETSI CHI trunks, licenses instances are consumed by the channels set as in service.

Each physical trunk port supports up to 24 channels in T1 mode, 23 channels in PRI and QSIG modes.

- Dialing Complete
  The majority of North-American telephony services use en-bloc dialing. Therefore the use of a ; is recommended at the end of all dialing short codes that use an N. This is also recommended for all dialing where secondary dial tone short codes are being used.
- AT&T Provider Settings
  For AT&T operation two information elements, TNS (Transit Network Selector) and NSF (Network Specific Facility), are sent in the call setup to the service provider. The values for TNS, NSF and the actual phone number presented to the line are determined by parsing the number dialed through, in sequence, the TNS, Special and Call by Call tabs. These tabs appear when the Provider setting on the Line tab is set to AT&T. Note also that B-channels within the same line can be brought from different service providers. Additionally some B-channels can be used 'call by call', that is, use a different service provider for each call.

5.4.6.1 Line

Line | Line (T1 PRI)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Number:
  Allocated by the system.
- Card/Module: *Software level = 4.1+.*
  Indicates the card slot or expansion module being used for the trunk device providing the line.
  - For IP400 control units: SLOT A on the control unit is shown as 1, SLOT B is shown as 2. Expansion modules are numbered from 4 upwards, for example trunks on the module in Expansion Port 1 are shown as 4.
  - For IP500 and IP500v2 control units: 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example trunks on the module in Expansion Port 1 are shown as 5.
- Port: *Software level = 4.1+.*
  Indicates the port on the Card/Module above to which the configuration settings relate.
- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.
- Line SubType: *Default = PRI*
  Set to PRI. If set to T1 see Line Form (T1) 206. If set to *ETSI*, *ETSI CHI (IP Office 4.2+)*, *QSIG A* or *QSIG B* see Line (E1) 197.
  - QSIG trunks and H323 IP trunks are not supported on IP500 and IP500v2 systems without IP500 Voice Networking 833 licenses.
- Channel Allocation: *Default = 23 -> 1*
  The order, 23 to 1 or 1 to 23, in which channels are used.
- Switch Type: *Default = NI2*
  Options *4ESS*, *5ESS*, *DMS100* and *NI2*.
- Provider: *Default = Local Telco*
  Select the PSTN service provider (*AT&T*, *Sprint*, *WorldCom* or *Local Telco*).
- Prefix: *Default = Blank*
  Enter the number to prefix to all incoming numbers for callback. This is useful if all users must dial a prefix to access an outside line. The prefix is automatically placed in front of all incoming numbers so that users can dial the number back.
- Add 'Not-end-to-end ISDN' Information Element: *Default = Never*. Software level = 4.2+.*
  Sets whether the optional 'Not end-to-end ISDN' information element should be added to outgoing calls on the line. The options are *Never*, *Always* or *POTS* (only if the call was originated by an analog extension). *The default is *Never* except for the following locales; for Italy the default is *POTS*, for New Zealand the default is *Always*.
- Send Redirecting Number: *Default = Off. Software level = 6.0+, IP500/IP500v2 only.*
  This option can be used on ISDN trunks where the redirecting service is supported by the trunk provider. Where supported, on twinned calls the caller ID of the original call is passed through to the twinning destination. This option is only used for twinned calls.
- Send Names: *Software level = 6.0+*
  This option is available when the Switch Type above is set to *DMS100*. If set, names are sent in the display field. The Z shortcode character 427 can be used to specify the name to be used.
- Names Length: *Software level = 6.0+*
  Set the allowable length for names, up to 15 characters, when Send Names is set above.
- Test Number:
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.
- Framing: *Default = ESF*
  Selects the type of signal framing used (ESF or D4).
- Zero Suppression: *Default = B8ZS*
  Selects the method of zero suppression used (B8ZS or AMI ZCS).

- Clock Quality: *Default = Network*
  Refer to the Manager Installation Manual for full details. This option sets whether the system should try to take its clock source for call synchronization and signalling from this line. Preference should always be given to using the clock source from a central office exchange if available by setting at least one exchange line to *Network*.
  - If multiple lines are set as *Network*, the order in which those lines are used is described in the Manager Installation Manual. If additional lines are available, *Fallback* can be used to specify a clock source to use should the *Network* source not be available.
  - Lines from which the clock source should not be taken should be set as *Unsuitable*.
  - If no clock source is available, the system uses its own internal 8KHz clock source.
  - In scenarios where several systems are network via digital trunk lines, care must be taken to ensure that all the systems use the same clock source. The current source being used by a system is reported within the System Status Application.
- CSU Operation
  Tick this field to enable the T1 line to respond to loop-back requests from the line.
- Haul Length: *Default = 0-115 feet*
  Sets the line length to a specific distance.
- Channel Unit: *Default = Foreign Exchange*
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are *Foreign Exchange*, *Special Access* or *Normal*.
- CRC Checking: *Default = On*
  Turns CRC on or off.
- Line Signaling:
  The field can be set to either *CPE* (Customer Premises Equipment) or *CO* (Central Office). This field should normally be left at its default of *CPE*. The setting *CO* is normally only used in lab back-to-back testing.
- Incoming Routing Digits: *Default=0 (present call immediately)*
  Sets the number of routing digits expected on incoming calls. This allows the line to present the call to the system once the expected digits have been received rather than waiting for the digits timeout to expire. This field only affects T1 line channels set to *E&M Tie*, *E&M DID*, E&M Switched 56K and *Direct Inward Dial*.
- Admin: *Default = In Service. Software level = 6.1.*
  This field allows a trunk to be taken out of service if required for maintenance or if the trunk is not connected.

5.4.6.2 Channels

This tab allows settings for individual channels within the trunk to be adjusted. This tab is not available for trunks sets to ETSI or QSIG mode.

Line | Channels (T1 PRI)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Channel
  Allocated by the system.
- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.
- Line Appearance ID: *Default = Auto-assigned, Range = 2 to 9 digits. Software level = 3.0+.*
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number.
- Direction: *Default = Both Directions*
  The direction of calls on the channel (*Incoming*, *Outgoing* or *Both Directions*).
- Bearer: *Default = Any*
  The type of traffic carried by the channel (*Voice*, *Data* or *Any*).
- Service: *Default = None.*
  If the line provider is set to AT&T, selects the type of service provided by the channel from *Call by Call*, *SDN* (inc GSDN), *MegaCom800*, *MegaComWats*, *Accunet*, *NLDS*, *I800*, *ETN*, *Private Line*, *AT&T Multiquest*. For other providers the service options are *None* or *No Service*.
- Admin: *Default = Out of Service*
  Used to indicate the channel status (*In Service*, *Out of Service* or *Maintenance*).
- Tx Gain: *Default = 0dB*
  The transmit gain in dB.
- Rx Gain: *Default = 0dB*
  The receive gain in dB.

5.4.6.3 TNS

This tab is shown when the line Provider is set to AT&T. It allows the entry of the Network Selection settings. These are prefixes for alternative long distance carriers. When a number dialed matches an entry in the table, that pattern is stripped from the number before being sent out. This table is used to set field in the TNS (Transit Network Selection) information element for 4ESS and 5ESS exchanges. It is also used to set fields in the NSF information element.

Line | TNS (T1 PRI)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- TNS Code:
  The pattern for the alternate long distance carrier. For example: The pattern 10XXX is added to this tab. If 10288 is dialed, 10 is removed and 288 is placed in the TNS and NSF information.

5.4.6.4 Special

This tab is shown when the line Provider is set to AT&T. This table is used to set additional fields in the NSF information element after initial number parsing by the TNS tab. These are used to indicate the services required by the call. If the channel is set to Call by Call, then further parsing is done using the entries in the Call by Call tab.

Line | Special (T1 PRI)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Short code:
  The number which results from the application of the rules specified in the User or System Short code tables and the Network Selection table and the Call-by-call table to the number dialed by the user.
- Number:
  The number to be dialed to line.
- Special: *Default = No Operator*
  (*No Operator*, *Local Operator* or *Presubscribed Operator*).
- Plan: *Default = National*
  (*National* or *International*).

Typical values are:

Short Code	Number	Service
011N	N	No Operator, International
010N	N	Local Operator, International
01N	N	Local Operator, National
00N	N	Presubscribed Operator, National
0N	N	Presubscribed Operator, National
1N	1N	No operator, National

5.4.6.5 Call By Call

This tab is shown when the line Provider is set to AT&T. Settings in this tab are only used when calls are routed via a channel which has its Service set to *Call by Call*.

It allows short codes to be created to route calls to a different services according to the number dialed. Call By Call reduces the costs and maximizes the use of facilities. Call By Call chooses the optimal service for a particular call by including the Bearer capability in the routing decision. This is particularly useful when there are limited resources.

Line | Call By Call (T1 PRI)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Short Code:
  The number which results from the application of the rules specified in the User or System Short code tables and the Network Selection table to the number dialed by the user.
- Number:
  The number to be dialed to line.
- Bearer: *Default = Any*
  The type of channel required for the call (Voice, Data or Any).
- Service: *Default = AT&T*
  The service required by the call (*SDN* (inc GSDN), *MegaCom800*, *MegaCom*, *Inwats*, *Wats*, *Accunet*, *NLDS*, *I800*, *ETN*, *Private Line*, *AT&T Multiquest*).

5.4.7 S0 Line

These settings are used for S0 ports provided by an S08 expansion module connected to the control unit. Though displayed as lines, these BRI ports are used for connection of ISDN2 devices such as video conferencing units or ISDN PC cards. For full details of installation refer to the Manager Installation manual.

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

5.4.7.1 Line

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

Line | Line (S0)	
Control Unit	SOE ✗, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- **Line Number**
  This parameter is not configurable. It is allocated by the system.
- **Telephone Number:**
  Used to remember the telephone number of this line. For information only.
- **Prefix:** *Default = Blank.*
  The prefix is used in the following ways:
  - **For incoming calls**
    The ISDN messaging tags indicates the call type (National, International or Unknown). If the call type is unknown, then the number in the Prefix field is added to the ICLID.
  - **For outgoing calls**
    Pre-IP Office 4.0: When a outgoing call is presented to the line with a leading digit to dial that matches the Prefix, that digit is stripped from the number. IP Office 4.0+: The prefix is not stripped, therefore any prefixes not suitable for external line presentation should be stripped using short codes.
- **National Prefix:** *Default = 0*
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 1923000000 is converted to 01923000000.
- **International Prefix:** *Default = 00*
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 00441923000000.
- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.
- **TEI:** *Default = 0*
  Not used. The Control Unit will ignore any entry.
- **Number of Channels:** *Default = 2*
  Defines the number of operational channels that are available on this line. 2 for BRI and up to 30 for PRI - depending upon the number of channels subscribed.
- **Outgoing Channels:** *Default = 2*
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.
- **Voice Channels:** *Default = 2*
  The number of channels available for voice use.
- **Data Channels:** *Default = 2*
  The number of channels available for data use. If left blank the value is 0.

5.4.7.2 Short Codes

Line short codes can be applied to the digits received with incoming call. The stage at which they are applied also varies depending on the trunk type.

The line Short Code tab is shown for internal trunk types: *QSIG* (T1, E1, H323), *BRI S0*, *H323*, *SCN*, *SES*. Incoming calls are routed by looking for a match to the incoming digits in the following order:

- Extension number (including remote numbers in a Small Community Network).
- Line short codes (excluding ? short code).
- System short codes (excluding ? short code).
- Line ? short code.
- System ? short code.

Line | Short Codes	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

Short codes can be added and edited using the Add, Remove and Edit buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

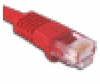
5.4.7.3 Channels

This tab allows settings for individual channels within the trunk to be adjusted. For So channels this form is not used.

Line | Channels (S0)	
Control Unit	SOE ✗, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Appearance ID
  Not used with So lines.

5.4.8 H323 Line



These lines are added manually. They allow voice calls to be routed over data links within the system. They are therefore dependent on the IP data routing between the system and the destination having being configured and tested.

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

Network Assessments

Not all data connections are suitable for voice traffic. A network assessment is required for internal network connections. For external network connections a service level agreement is required from the service provider. Avaya cannot control or be held accountable for the suitability of a data connection for carrying voice traffic. Refer to the Manager Installation Manual for further details of Network Assessments and VoIP requirements.

- QSIG trunks and H323 IP trunks are not supported on IP500 and IP500v2 systems without IP500 Voice Networking 833 licenses.

5.4.8.1 Line

- QSIG trunks and H323 IP trunks are not supported on IP500 and IP500v2 systems without IP500 Voice Networking 833 licenses.

Line | H323 Line | VoIP Line	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Line Number: *Default = 0, Range = 1 to 249.*
  Enter the line number that you wish. Note that this must be unique.

- Telephone Number:
  Used to remember the telephone number of this line. For information only.

- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.

  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.

- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.

- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.

- Prefix: *Default = Blank.*
  The prefix is used in the following ways:

  - For incoming calls
    The ISDN messaging tags indicates the call type (National, International or Unknown). If the call type is unknown, then the number in the Prefix field is added to the ICLID.

  - For outgoing calls
    Pre-IP Office 4.0: When a outgoing call is presented to the line with a leading digit to dial that matches the Prefix, that digit is stripped from the number. IP Office 4.0+: The prefix is not stripped, therefore any prefixes not suitable for external line presentation should be stripped using short codes.

- National Prefix: *Default = 0*
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 1923000000 is converted to 01923000000.

- International Prefix: *Default = 00*
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 00441923000000.

- Number of Channels: *Default = 20, Range 0 to 128.*
  Defines the number of operational channels that are available on this line.

- Outgoing Channels: *Default = 20, Range 0 to 128.*
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.

- Data Channels: *Default = 20, Range 0 to 128.*
  The number of channels available for data use. If left blank the value is 0.

- Voice Channels: *Default = 20, Range 0 to 128.*
  The number of channels available for voice use.

- TEI: *Default = 0, Range = 0 to 127.*
  The Terminal Equipment Identifier. Used to identify each Control Unit connected to a particular ISDN line. For Point to Point lines this is typically (always) 0. It can also be 0 on a Point to Multi-Point line, however if multiple devices are actually sharing a Point to Multi-Point line it should be set to 127 which will result in the exchange deciding on the TEI's to be used by this Control Unit.

5.4.8.2 Short Codes

Line short codes can be applied to the digits received with incoming call. The stage at which they are applied also varies depending on the trunk type.

The line Short Code tab is shown for internal trunk types: *QSIG* (T1, E1, H323), *BRI S0*, *H323*, *SCN*, *SES*. Incoming calls are routed by looking for a match to the incoming digits in the following order:

- Extension number (including remote numbers in a Small Community Network).
- Line short codes (excluding ? short code).
- System short codes (excluding ? short code).
- Line ? short code.
- System ? short code.

Line | Short Codes	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

5.4.8.3 VoIP

Line | H323 Line | VoIP Settings (IP)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

H323 Trunk (Non SCN)

The following settings are applicable to trunks with their Supplementary Services set to other than *IP Office SCN* or *IP Office SCN - Fallback*.

- Gateway IP Address: *Default = Blank*
  Enter the IP address of the gateway device at the remote end. This address must not be shared by any other IP line (H323, SIP, SES or IP DECT).

- Compression Mode: *Default = Automatic Selection*
  This field defines the compression mode (codec) or modes offered during call setup. The supported codecs are *G.711 ALAW 64K*, *G.711 ULAW 64K*, *G.729(a) 8K CS-ACELP*, *G.723.1 6K3 MP-MLQ*.

  - *Automatic Select*
    This setting is used as follows:

- Pre-IP Office 5.0: *Automatic Select* uses the codecs in the order of preference *G729a*, *G711 ALAW*, *G711 ULAW* and *G.723.1.*

- IP Office 5.0+: The Automatic Codec Preference 160 setting (System | Telephony | Telephony 160) can be used to set which codec should be first in the list of preferred codecs. The remaining codecs are used in the order as above. Note that the G711 codecs are treated as a pair, so if one is selected as the first preference, the other will automatically be second in the list.

  - If required, a specific codec can be selected. However, if during call setup negotiation with a specific codec, connect fails, the system will fallback to using automatic selection.

  - Codecs not selected for a SIP connection are not used in codec negotiation. For H.323 it is not possible to exclude codecs from negotiation, only to change the order of codec preference.

- Supplementary Services: *Default = IP Office SCN*
  Selects the supplementary service signaling method for use across the H323 trunk. The remote end of the trunk must support the same option. Options are:
  - *None*
    No supplementary services are supported.

  - *H450*
    Use for H323 lines connected to another PBX or device that uses H450.

  - *QSIG*
    Use for H323 lines connected to another PBX or device that uses QSIG.

  - *IP Office SCN*
    This option is used for H323 trunks within a Small Community Network (SCN). The systems with an SCN automatically exchange information about users and extensions, allowing remote users to be called without any additional configuration on the local system. For full details of SCN operation see Small Community Networking 810.

  - *IP Office SCN - Fallback (IP Office 5+ only)*
    This option is used for an SCN trunk connection as above, where the system at the end of the trunk will try to take over the selected SCN Backup Options if this system is not visible within the Small Community Network for a period of more than 3 minutes. Only one H323 trunk per system can be set as *IP Office SCN - Fallback*. See SCN Fallback 826.

- Call Initiation Timeout: *Default = 4 seconds, Range = 1 to 99 seconds. Software level = 4.2+.*
  This option sets how long the system should wait for a response to its attempt to initiate a call before following the alternate routes set in an ARS 409 form.

- VoIP Silence Suppression: *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using *G711* between systems.

- Enable FastStart for non-Avaya IP Phones: *Default = Off*
  A fast connection procedure. Reduces the number of messages that need to be exchanged before an audio channel is created.

- Fax Transport Support: *Off (Fixed)*
  This option is only supported on trunks with their Supplementary Services set to *IP Office SCN* or *IP Office SCN - Fallback*.

* IP Office 5.0+: IP500 and IP500 v2 systems with an IP500 VCM are able to support fax relay 749 across H323 SCN lines with Fax Transport Support selected. This will use 2 VCM channels in each of the systems.

* Local Tones: *Default = Off*
  When selected, the tones are generated by the local system to which the phone is registered. This option should not be used with lines being used for Small Community Networking. For the Small Office Edition control unit, this field should not be enabled.

* DTMF Support: *Default = Out of Band*
  DTMF tones can be sent to the remote end either as DTMF tones within the calls audio path (*In Band*) or a separate signals (*Out of Band*). Out of Band is recommended for compression modes such as G.729 and G.723 compression modes where DTMF in the voice stream could become distorted.

  * For trunks with Supplementary Services set to *IP Office SCN* or *IP Office SCN - Fallback*, this option is fixed to *Out of Band*.

* Allow Direct Media Path: *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the system) or can be routed alternately if possible within the network structure.

  * If enabled, H323 calls can take routes other than through the system. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.

  * If disabled or not supported at on one end of the call, the call is routed via the system.

    * On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.

    * On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel.

* Progress Ends Overlap Send: *Default = Off. Software level = 3.2+.*
  Some telephony equipment, primarily AT&T switches, over IP trunks send a H323 Progress rather than H323 Proceeding message to signal that they have recognized the digits sent in overlap state. By default the system expects an H323 Proceeding message. This option is not available by default. If required, the value *ProgressEndsOverlapSend* must be entered into the Source Numbers 272 tab of the *NoUser* 767 user.

* Default Name for Display IE: *Default = Off. Software level = 4.2+.*

H323 Trunk (SCN)

The following settings are applicable to trunks with their Supplementary Services set to *IP Office SCN* or *IP Office SCN - Fallback*.

* Gateway IP Address: *Default = Blank*
  Enter the IP address of the gateway device at the remote end. This address must not be shared by any other IP line (H323, SIP, SES or IP DECT).

* Compression Mode: *Default = Automatic Selection*
  This field defines the compression mode (codec) or modes offered during call setup. The supported codecs are *G.711 ALAW 64K*, *G.711 ULAW 64K*, *G.729(a) 8K CS-ACELP*, *G.723.1 6K3 MP-MLQ*.

  * *Automatic Select*
    This setting is used as follows:

* Pre-IP Office 5.0: *Automatic Select* uses the codecs in the order of preference *G729a*, *G711 ALAW*, *G711 ULAW* and *G.723.1.*

* IP Office 5.0+: The Automatic Codec Preference 160 setting (System | Telephony | Telephony 160) can be used to set which codec should be first in the list of preferred codecs. The remaining codecs are used in the order as above. Note that the G711 codecs are treated as a pair, so if one is selected as the first preference, the other will automatically be second in the list.

  * If required, a specific codec can be selected. However, if during call setup negotiation with a specific codec, connect fails, the system will fallback to using automatic selection.

  * Codecs not selected for a SIP connection are not used in codec negotiation. For H.323 it is not possible to exclude codecs from negotiation, only to change the order of codec preference.

* Supplementary Services: *Default = IP Office SCN*
  Selects the supplementary service signaling method for use across the H323 trunk. The remote end of the trunk must support the same option. Options are:

  * *None*
    No supplementary services are supported.

  * *H450*
    Use for H323 lines connected to another PBX or device that uses H450.

- *QSIG*
  Use for H323 lines connected to another PBX or device that uses QSIG.

- *IP Office SCN*
  This option is used for H323 trunks within a Small Community Network (SCN). The systems with an SCN automatically exchange information about users and extensions, allowing remote users to be called without any additional configuration on the local system. For full details of SCN operation see Small Community Networking 810.

- *IP Office SCN - Fallback (IP Office 5+ only)*
  This option is used for an SCN trunk connection as above, where the system at the end of the trunk will try to take over the selected SCN Backup Options if this system is not visible within the Small Community Network for a period of more than 3 minutes. Only one H323 trunk per system can be set as *IP Office SCN - Fallback*. See SCN Fallback 826.

- SCN Backup Options: *Software level = 5.0+.*
  These options are only available on when the Supplementary Services option is set to *IP Office - Fallback*. The system to which the trunk connects must be an IP Office 5+ system. The intention of this feature is to attempt to maintain a minimal level of operation while problems with the local system are resolved.

  - Backs up my IP Phones: *Default = On.*
    This option is used for Avaya 1600, 4600, 5600 and 9600 Series phones registered with the system. When selected, it will share information about the registered phones and users on those phones with the other system.

    - If the local system is no longer visible to the phones, the phones will reregister with the other system. The users who were currently on those phones will appear on the other system as if they had hot desked.

    - Note that when the local system is restored to the SCN, the phones will not automatically re-register with it. A phone reset via either a phone power cycle or using the System Status Application is required.

    - When phones have registered with the other system, they will show an R on their display.

  - Backs up my Hunt Groups: *Default = On.*
    When selected, any hunt groups the local system is advertising to the SCN are advertised from the other system when fallback is required. The trigger for this occurring is Avaya H323 phones registered with the local system registering with the other system, ie. Backs up my IP Phones above must also be enabled.

    - When used, the only hunt group members that will be available are as follows:

      - If the group was a distributed hunt group, those members who were remote members on other systems still visible within the SCN.

      - Any local members who have hot desked to another system still visible within the SCN.

    - When the local system becomes visible to the other system again, the groups will return to be advertised from the local system.

  - Backs up my Voicemail: *Default = On.*
    This option can be used if the local system is hosting the Voicemail Pro server being used by the SCN. If selected, when the local system is no longer visible to the voicemail server, the other system will act as host for the voicemail server.

    - This option requires the other system to have licenses for the Voicemail Pro features that are required to operated during any fallback period.

    - This option requires Voicemail Pro 5.0+.

- Call Initiation Timeout: *Default = 4 seconds, Range = 1 to 99 seconds. Software level = 4.2+.*
  This option sets how long the system should wait for a response to its attempt to initiate a call before following the alternate routes set in an ARS 409 form.

- VoIP Silence Suppression: *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using *G711* between systems.

- Fax Transport Support: *Off (Fixed)*
  This option is only supported on trunks with their Supplementary Services set to *IP Office SCN* or *IP Office SCN - Fallback*.

  - For IP Office 5+, IP500 and IP500v2 systems with an IP500 VCM are able to support fax relay 749 across H323 SCN lines with Fax Transport Support selected. This will use 2 VCM channels in each of the systems.

- Local Tones: *Default = Off*
  When selected, the tones are generated by the local system to which the phone is registered. This option should not be used with lines being used for Small Community Networking. For the Small Office Edition control unit, this field should not be enabled.

- DTMF Support: *Default = Out of Band*
  DTMF tones can be sent to the remote end either as DTMF tones within the calls audio path (*In Band*) or a separate signals (*Out of Band*). Out of Band is recommended for compression modes such as G.729 and G.723 compression modes where DTMF in the voice stream could become distorted.

- For trunks with Supplementary Services set to *IP Office SCN* or *IP Office SCN - Fallback*, this option is fixed to *Out of Band*.

- Allow Direct Media Path: *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the system) or can be routed alternately if possible within the network structure.

  - If enabled, H323 calls can take routes other than through the system. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.

  - If disabled or not supported at on one end of the call, the call is routed via the system.

    - On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.

    - On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel.

5.4.9 IP DECT Line

This type of line can be manually added. They are used to route voice calls over an IP data connection to an Avaya IP DECT system. Only one IP DECT line can be added to a system. Refer to the Manager IP DECT Installation manual for full details.

For North American locales, an IP DECT line is only supported with the IP Office 4.0 Q2 2007 maintenance release and higher.

5.4.9.1 Line

Currently only one IP DECT line is supported on a system. For North American locales, an IP DECT line is only supported with the IP Office 4.0 Q2 2007 maintenance release and higher.

Line | Line (IP DECT)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.1+.
Mergeable	✗.

- Line Number
  This number is allocated by the system and is not adjustable.
- Associated Extensions
  Lists all the DECT extensions associated with the IP DECT line.

5.4.9.2 Gateway

Line | Gateway (IP DECT)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.1+.
Mergeable	✗.

- Auto-Create Extension: *Default = Off. Software level = 4.0+.*
  If enabled, subscription of a handset with the DECT system causes the auto-creation of a matching numbered extension within the system configuration if one does not already exist.
- Enable DHCP Support: *Default = Off*
  This option is not supported for use with Avaya IP DECT R4. The IP DECT base stations require DHCP and TFTP support. Enable this option if the system is being used to provide that support, using IP addresses from its DHCP range (LAN1 or LAN2) and its TFTP server setting. If not enabled, alternate DHCP and TFTP options must be provided during the IP DECT installation.
    - If it is desired to use the system for DHCP support of the ADMM and IP DECT base stations only, the system address range should be set to match that number of addresses. Those addresses are then taken during the system restart and will not be available for other DHCP responses following the restart.
    - For Small Office Edition and IP406V2 control units, use of the embedded voicemail memory card slot for the TFTP server is recommended for small IP DECT installations. See System | TFTP IP Server Address 143. For other control units, or larger IP DECT installations, the use of a non-embedded TFTP software option other than Manager is recommended.
- Boot File: *Default = ADMM_RFP_1_0_0.tftp, Range = Up to 31 characters.*
  The name and path of the ADMM software file. The path is relative to the TFTP server root directory.
- ADMM MAC Address: *Default = 00:00:00:00:00:00*
  This field must be used to indicate the MAC address of the IP DECT base station that should load the ADMM software file and then act as the IP DECT system's ADMM. The address is entered in hexadecimal format using comma, dash, colon or period separators.
- VLAN ID: *Default = Blank, Range = 0 to 4095.*
  If VLAN is being used by the IP DECT network, this field sets the VLAN address assigned to the base stations by the system if Enable DHCP Support is selected.
    - The system itself does not apply or use VLAN marking. It is assumed that the addition of VLAN marking and routing of VLAN traffic is performed by other switches within the customer network.
    - An ID of zero is not recommended for normal VLAN operation.
    - When blank, no VLAN option is sent to the IP DECT base station.
- Base Station Address List: *Default = Empty*
  This box is used to list the MAC addresses of the IP DECT base stations, other than the base station being used as the ADMM and entered in the ADMM MAC Address field. Right-click on the list to select Add or Delete. or use the Insert and Delete keys. The addresses are entered in hexadecimal format using comma, dash, colon or period separators.

5.4.9.3 VoIP

- Gateway IP Address: *Default = Blank*
  Enter the IP address of the gateway device at the remote end. This address must not be shared by any other IP line (H323, SIP, SES or IP DECT).

- Compression Mode: *Default = Automatic Selection*
  This field defines the compression mode (codec) or modes offered during call setup. The supported codecs are *G.711 ALAW 64K*, *G.711 ULAW 64K*, *G.729(a) 8K CS-ACELP*, *G.723.1 6K3 MP-MLQ*.

  - *Automatic Select*
    This setting is used as follows:

- Pre-IP Office 5.0: *Automatic Select* uses the codecs in the order of preference *G729a*, *G711 ALAW*, *G711 ULAW* and *G.723.1.*

- IP Office 5.0+: The Automatic Codec Preference 160 setting (System | Telephony | Telephony 160) can be used to set which codec should be first in the list of preferred codecs. The remaining codecs are used in the order as above. Note that the G711 codecs are treated as a pair, so if one is selected as the first preference, the other will automatically be second in the list.

  - If required, a specific codec can be selected. However, if during call setup negotiation with a specific codec, connect fails, the system will fallback to using automatic selection.

  - Codecs not selected for a SIP connection are not used in codec negotiation. For H.323 it is not possible to exclude codecs from negotiation, only to change the order of codec preference.

- Gain: *Default = Default. Software level = 3.0 to 4.1.*
  Allows adjustment of the volume. The gain is selectable from -31dB to +31dB in 1dB increments.

- TDM->IP Gain: *Default = Default (0dB). Range = -31dB to +31dB. Software level = 4.1 Q1 2008+.*
  Allows adjustment of the gain on audio from the system TDM interface to the IP connection.

- IP->TDM Gain: *Default = Default (0dB). Range = -31dB to +31dB. Software level = 4.1 Q1 2008+.*
  Allows adjustment of the gain on audio from the IP connection to the system TDM interface.

- VoIP Silence Suppression: *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using *G711* between systems.

- Allow Direct Media Path: *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the system) or can be routed alternately if possible within the network structure.

  - If enabled, H323 calls can take routes other than through the system. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.

  - If disabled or not supported at on one end of the call, the call is routed via the system.

    - On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.

    - On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel.

5.4.10 SIP Line

IP Office 4.0 and higher supports SIP voice calls through the addition of SIP trunks to the system configuration. This approach allows users with non-SIP phones to make and receive SIP calls.

Use of SIP requires the following:

1. **SIP Service Account**
   An account or accounts with a SIP internet service provider (ITSP). The method of operation and the information provided will vary. The key requirement is a SIP URI, a web address of the form *name@example.com*. This is the equivalent of a SIP telephone number for making and receiving calls via SIP.

2. **Voice Compression Channels**
   SIP calls use system voice compression channels in the same way as used for standard IP trunks and extensions. These are provided by the installation of VCM modules within the control unit. RTP relay is applied to SIP calls where applicable.

3. **Licensing**
   SIP trunks require licenses in the system configuration. These set the maximum number of simultaneous SIP calls supported by the system.

4. **Firewall Traversal**
   Routing traditional H323 VoIP calls through firewalls often fails due to the effects of NAT (Network Address Translation). For SIP a number of ways of ensuring successful firewall traversal have been created.

   - **STUN (Simple Traverse of UDP NAT)**
     UDP SIP can use a mechanism called STUN to cross firewalls between the switch and the ITSP. This requires the ITSP to provide the IP address of their STUN server and the system to then select from various STUN methods how to connect to that server. The system can attempt to auto-detect the required settings to successfully connect. These settings are part of the System | LAN1 148 and System | LAN2 155 forms. STUN is not required is the ITSP used Session Border Control (SBC).

   - **TURN (Traversal Using Relay NAT)**
     TCP SIP can use a mechanism called TURN (Traversal Using Relay NAT). This is not currently supported.

   - **Firewall**
     The firewall between LAN1 and LAN2 is not applied to SIP calls.

5. **SIP Trunks**
   These trunks are manually added to the system configuration. Typically a SIP trunk is required for each SIP ITSP being used. As the configuration provides methods for multiple URI's from that ITSP to use the same trunk. For each trunk at least one SIP URI entry is required, up to 150 SIP URI's are supported on the same trunk. Amongst other things this sets the incoming and outgoing groups for call routing.

6. **Outgoing Call Routing**
   The initial routing uses any standard short code with a dial feature. The short code's Line Group ID should be set to match the Outgoing Group ID of the SIP URI 238 channels to use. However the short code must also change the number dialed into a destination SIP URI suitable for routing by the ITSP. In most cases, if the destination is a public telephone network number, a URI of the form *123456789@example.com* is suitable. For example:

   - **Code**: 9N#
   - **Feature**: Dial
   - **Telephone Number**: N"@example.com"
   - **Line Group ID**: 100

7. **Incoming Call Routing**
   Incoming SIP calls are routed in the same way as other incoming external calls. The caller and called information in the SIP call header can be used to match Incoming CLI and Incoming Number settings in normal system Incoming Call Route 344 entries.

8. **DiffServ Marking**
   DiffServ marking is applied to calls using the DiffServer Settings on the System | LAN | VoIP 150 tab of either LAN1 or LAN2 as set by the line's **Use Network Topology Info** setting.

SIP URIs
Calls across SIP require URI's (Uniform Resource Identifiers), one for the source and one for the destination. Each SIP URI consists of two parts, the user part (for example *name*) and the domain part (for example *example.com*) to form a full URI (in this case *name@example.com*). SIP URI's can take several forms:

- name@117.53.22.2
- name@example.com
- 012345678@example.com

Typically each account with a SIP service provider will include a SIP URI or a set of URI's. The domain part is then used for the SIP trunk configured for routing calls to that provider. The user part can be assigned either to an individual user if you have one URI per user for that ITSP, or it can also be configured against the line for use by all users who have calls routed via that line.

- IP Office 5.0+: If the wildcard * is used in the SIP trunk's Local URI, Contact and Display fields, that SIP trunk will accept any incoming SIP call. The incoming call routing is still performed by the system incoming call routes based on matching the values received with the call or the URI's incoming group setting. Outgoing calls using the trunk will need to use another URI set against the trunk.

Resource Limitation
A number of limits can affect the number of SIP calls. When one of these limits is reached the following occurs: any further outgoing SIP calls are blocked unless some alternate route is available using ARS; any incoming SIP calls are queued until the required resource becomes available. Limiting factors are:

- the number of licensed SIP channels.
- the number of SIP channels configured for a SIP URI.
- the number of voice compression channels.
  - SIP Line Call to/from Non-IP Devices
    Voice compression channel required.
  - Outgoing SIP Line Call from IP Device
    No voice compression channel required.
  - Incoming SIP Line Call to IP Device
    If using the same codec, voice compression channel reserved until call connected. If using differing codecs then 2 channels used.

SIP Information Display
The full from and to SIP URI will be recorded for use by SMDR, CBC and CCC. For all other applications and for telephone devices, the SIP URI is put through system directory matching the same as for incoming CLI matching. First a match against the full URI is attempted, then a match against the user part of the URI. Directory wildcards can also be used for the URI matching.

SIP Standards
The system implementation of SIP conforms to the following SIP RFC's.

RFC	Description
2833 [7]	RTP Payload for DTMF digits, telephony tones and telephony signals.
3261 [8]	SIP Session Initiation Protocol.
3263	Locating SIP Services (IP Office 6.1+)
3264 [11]	An Offer/Answer Model with Session Description Protocol (SDP).
3323 [14]	A Privacy Mechanism for SIP
3489 [18]	STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NAT's).
3824 [24]	Using E.164 Numbers with the Session Initiation Protocol (SIP). E.164 is the ITU-T recommendation for international public telecommunication numbering plans.

5.4.10.1 Incoming Call Routing

Incoming SIP calls are routed using Incoming Call Routes 342 in the same way as call arriving on other external trunks. The following Incoming Call Route fields are used to determine which route is the best match for a call.

- **Line Group ID**
  This field is matched against the Incoming Group settings of the SIP URI (Line | SIP URI). This must be an exact match.
- **Incoming Number**
  This field can be used to match the called details (TO) in the SIP header of incoming calls. It can contain a number, SIP URI or Tel URI. For SIP URI's the domain part of the URI is removed before matching by incoming call routing occurs. For example, for the SIP URI mysip@example.com , only the user part of the URI, ie. mysip, is used for matching.
- **Incoming CLI**
  This field can be used to match the calling details (FROM) in the SDP header of incoming SIP calls. It can contain a number, SIP URI, Tel URI or IP address received with SIP calls. For all types of incoming CLI except IP addresses a partial entry can be used to achieve the match, entries being read from left to right. For IP addresses only full entry matching is supported.

The fields Bearer Capability and Incoming Sub Address are not used for matching of incoming SIP calls. The remain Incoming Call Route fields, including those voice recording, as used as for all call types.

- IP Office 5.0+: If the wildcard * is used in the SIP trunk's Local URI, Contact and Display fields, that SIP trunk will accept any incoming SIP call. The incoming call routing is still performed by the system incoming call routes based on matching the values received with the call or the URI's incoming group setting. Outgoing calls using the trunk will need to use another URI set against the trunk.

5.4.10.2 SIP Line

Line | SIP Line	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	✗.

- Line Number: *Default = Automatically assigned.*
  By default a value is assigned by the system. This value can be changed but it must be unique.
- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.
- ITSP Domain Name: *Default = Blank.*
  This field is used to enter the domain part of the SIP URI provided by the ITSP. For example, in the SIP URI *name@example.com*, the domain part of the URI is *example.com*. For outgoing calls the user part of the SIP URI is determined in a number of ways:
  - For the user making the call, the user part of the FROM SIP URI is determined by the settings of the SIP URI 238 channel entry being used to route the call. This will use one of the following:
    - a specific name entered in Local URI field of the channel entry.
    - or specify using the primary or secondary authentication name set for the line below
    - or specify using the SIP Name set for the user making the call (User | SIP | SIP Name 303).
  - For the destination of the call, the user part of the TO SIP URI is determined by the dialing short codes of the form 9N / N"@example.com" where N is the user part of the SIP URI.
- ITSP IP Address: *Default = 0.0.0.0. Software level = 4.0 to 6.0.*
  This value is provided by the SIP ITSP. This address must not be shared by any other IP, IP DECT or SIP line in the system configuration.
  - IP Office 6.0+: Domain name resolution using the ITSP Domain Name value can be used instead. This requires a DNS server and the system to be acting as a DHCP client. If that is the case, leaving the ITSP IP Address as 0.0.0.0 will cause the system to attempt to resolve the domain name using DNS.
  - IP Office 6.1+: This setting has replaced by ITSP Proxy Address on the Transport 236 tab.
- Prefix: *Default = Blank. Software level = 6.0.*
  This prefix is removed from the called number on outgoing calls if present. See SIP Prefix Operation 235 below.
- National Prefix: *Default = 0. Software level = 6.0.*
  This prefix is added to calls identified as not being international.
- Country Code: *Default = Blank. Software level = 6.0.*
  Set to match the local country code of the system location.
- International Prefix: *Default = 00. Software level = 6.0.*
  This prefix is added to calls identified as not being national.
- Send Caller ID: *Default = None. Software level = 5.0+.*
  Select which value the SIP line should use for the original calling party ID when routing twinned calls.
    - IP Office 6.1: This setting is also used for forwarded calls. Note that the values on the System | Twinning 177 tab override this if set. For incoming calls to a hunt group, the hunt group details will be provided and not the details of the answering agent. This setting is mergeable.
    - IP Office 5.0+: The SIP line Send Caller ID setting takes priority.
    - IP Office 6.0+: The values on the System | Twinning 177 tab override the SIP lines Send Caller ID setting.
  - *Diversion Header (Software level = 6.0+)*
  - *Remote Party ID*
  - *P Asserted ID*
    This is the default setting for new SIP lines.

- *None*
  This option corresponds to the ISDN withheld setting. This is the default setting for existing SIP lines in configurations upgraded to IP Office 5+.

- **Refer Support**: *Default = On. Software level = 6.1+*
  REFER is the method used by many SIP device, including SIP trunks, to transfer calls. These settings can be used to control whether REFER is used as the method to transfer calls on this SIP trunk to another call on the same trunk. If supported, once the transfer has been completed, the system is no longer involved in the call. If not supported, the transfer may still be completed but the call will continue to be routed via the system.

  - **Incoming**: *Default = Auto*
    Select whether REFER can or should be used when an attempt to transfer an incoming call on the trunk results in an outgoing call on another channel on the same trunk. The options are:

    - *Always*
      Always use REFER for call transfers that use this trunk for both legs of the transfer. If REFER is not supported, the call transfer attempt is stopped.

    - *Auto*
      Request to use REFER if possible for call transfers that use this trunk for both legs of the transfer. If REFER is not supported, transfer the call via the system as for the *Never* setting below.

    - *Never*
      Do not use REFER for call transfers that use this trunk for both legs of the transfer. The transfer can be completed but will use 2 channels on the trunk.

  - **Outgoing**: *Default = Auto*
    Select whether REFER can or should be used when attempt to transfer an outgoing call on the trunk results in an incoming call on another channel on the same trunk. This uses system resources and may incur costs for the duration of the transferred call. The options available are the same as for the **Incoming** setting.

- **In Service**: *Default = On.*
  When this field is not selected, the SIP trunk is unregistered and not available to incoming and outgoing calls.

- **Registration Required**: *Default = Off. Software level = 4.0 to 6.0.*
  If selected, the SIP trunk will register with the ITSP using the value in the ITSP Domain Name field. For IP Office 6.1 this setting has moved to the SIP Credentials 242 tab.

- **Use Tel URI**: *Default = Off.*
  Use Tel URI format (for example TEL: +1-425-555-4567) rather than SIP URI format (for example name@example.com). This affects the From field of outgoing calls. The To field for outgoing calls will always use the format specified by the short codes used for outgoing call routing.

- **Check OOS**: *Default = On. Software level = 6.0+.*
  If enabled, the system will regularly check if the trunk is in service. Checking that SIP trunks are in service ensures that outgoing call routing is not delayed waiting for response on a SIP trunk that is not currently useable.

  - For both UDP and TCP trunks, the OPTIONS message is regularly sent. If no reply is received the trunk is taken out of service.

  - For TCP trunks, if the TCP connection is disconnected the trunk will be taken out of service.

  - For trunks using DNS, if the IP address is not resolved or the DNS resolution has expired, the trunk is taken out of service.

- **Call Routing Method**: *Default = Request URI. Software level = 6.0+.*
  This field allows selection of which incoming SIP information should be used for incoming number matching by the system's incoming call routes. The options are to match either the *Request URI* or the *To Header* element provided with the incoming call.

- **Originator number for forwarded and twinning calls**: *Default = Blank. Software level = 6.1+.*
  The field can be used to set a originator number for forwarded and twinned calls when using any of the Send Caller ID options above other than *None*.

For IP Office 6.1 and higher, the following fields have been replaced by those on the Transport 236 tab.

- **Network Configuration**: *Software level = 4.0 to 6.0.*

  - **Layer 4 Protocol**: *Default = UDP*
    This field sets whether the line uses UDP SIP or TCP SIP.

  - **Use Network Topology Info**: *Default = LAN1*
    This field associates the SIP line with the System | LAN1 | Network Topology 152 settings of either LAN1 or LAN2. If *None* is selected, STUN lookup is not applied and routing is determined by the system routing tables.

  - **Send Port**: *Default = 5060*
    This field sets the port to which the system sends outgoing SIP calls.

  - **Listen Port**: *Default = 5060*
    This field sets the port on which the system listens for incoming SIP calls.

For IP Office 6.0 and higher, the following fields have been replaced by those on the SIP Credentials 242 tab.

- Primary Authentication Name: *Default = Blank.*
  This value is provided by the SIP ITSP. Depending on the settings on the Local URI tab associated with the SIP call it may also be used as the user part of the SIP URI.
  - If the From field on the Local URI 238 being used for the call is set to Use Authentication Name and the Registration is set to Primary, this value is used as the user part of the SIP URI for calls.
- Primary Authentication Password: *Default = Blank.*
  This value is provided by the SIP ITSP.
- Primary Registration Expiry: *Default = 60 minutes.*
  This setting defines how often registration with the SIP ITSP is required following any previous registration.
- Secondary Authentication Name: *Default = Blank.*
  This value is provided by the SIP ITSP. Depending on the settings on the Local URI tab associated with the SIP call it may also be used as the user part of the SIP URI.
  - If the From field on the Local URI 238 being used for the call is set to Use Authentication Name and the Registration is set to Secondary, this value is used as the user part of the SIP URI for calls.
- Secondary Authentication Password: *Default = Blank.*
  This value is provided by the SIP ITSP.
- Secondary Registration Expiry: *Default = 60 minutes.*
  This setting defines how often registration with the SIP ITSP is required following any previous registration.

SIP Prefix Operation

IP Office 6.0+: The prefix fields Prefix, National Prefix, Country Code and International Prefix are available with the SIP Line settings. These fields are used in the following order:

1. If an incoming number (called or calling) starts with the + symbol, the + is replaced with the International Prefix .
2. If the Country Code has been set and an incoming number begins with that Country Code or with the International Prefix and Country Code, they are replaced with the National Prefix.
3. If the Country Code has been set and the incoming number does not start with the National Prefix or International Prefix, the International Prefix is added.
4. If the incoming number does not begin with either the National Prefix or International Prefix, then the Prefix is added.

For example, if the SIP Line is configured with prefixes as follows:

- Line Prefix: 9
- National Prefix: 90
- International Prefix: 900
- Country Code: 44

Number Received	Processing	Resulting Number
+441707362200	Following rule 1 above, the + is replace with the International Prefix (900), resulting in 900441707362200. The number now matches the International Prefix (900) and Country Code (44).Following rule 2 above they are replace with the National Prefix (90).	901707362200
00441707362200	Following rule 2 above the International Prefix (900) and the Country Code (44) are replaced with the National Prefix (90).	90107362200
441707362200	Following rule 2 above, the Country Code (44) is replace with the National Prefix (90).	901707362200
6494770557	Following rule 3 above the International Prefix (900) is added.	9006494770557

5.4.10.3 Transport

Line | SIP Line	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	6.1+.
Mergeable	✗ *(ITSP Proxy Address and Calls Route via Registrar are mergeable)*.

- ITSP Proxy Address: *Default = Blank*
  This is the SIP Proxy address used for outgoing SIP calls.
  - The address can be specified in the following ways:
    - If left blank, the ITSP Domain Name is used and is resolved by DNS resolution in the same way as if a DNS address had been specified as below.
    - An IP address.
    - A list of up to 4 IP addresses, with each address separated by a comma or space.
      - The addresses can include an indication of the relative call weighting of each address compared to the others. This is done by adding a w*N* suffix to the address where *N* is the weighting value. For example, in the list *213.74.81.102w3 213.74.81.100w2*, the weighting values assigns 1.5 times the weight of calls to the first address. The default weight if not specified is 1. A weight of 0 can be used to disable an address. Weight is only applied to outgoing calls.
      - If the Calls Route via Registrar setting below is enabled, the weighting is applied to registrations rather than calls.
    - A DNS address, for example *sbc.example.com*.
      - The DNS response may return multiple proxy addresses (RFC 3263). If that is the case, the system will resolve the address to use based on priority, TTL and weighting information included with each address.
      - A load balancing suffix can be added to specify that multiple proxy results should be returned if possible, for example *sbc.example.com(N)*. where *N* is the required number of addresses from 1 to 4.
  - This field is mergeable. However no more than 4 IP Addresses should be in use at any time. So, if the combined new and old address settings exceed 4, the new addresses are only phased into use as transactions in progress on the previous addresses are completed.
- Network Configuration
  These settings are applied to all calls using this SIP line.
  - Layer 4 Protocol: *Default = UDP*
    This field sets whether the line uses UDP SIP or TCP SIP.
  - Use Network Topology Info: *Default = LAN1*
    This field associates the SIP line with the System | LAN1 | Network Topology 152 settings of either LAN1 or LAN2. If *None* is selected, STUN lookup is not applied and routing is determined by the system routing tables.
  - Send Port: *Default = 5060*
    This field sets the port to which the system send outgoing SIP calls.
  - Listen Port: *Default = 5060*
    This field sets the port on which the system listens for incoming SIP calls.
- Explicit DNS Server(s): *Default = 0.0.0.0 (Off)*
  If specific DNS servers should be used for SIP trunk operation rather than the general DNS server specified or obtained for the system, the server addresses can be specified here.
- Calls Route via Registrar: *Default = On*
  If selected, all calls are routed via the same proxy as used for registration. If multiple ITSP proxy addresses have been specified, the weighting for those addresses is applied to the registrations.
- Separate Registrar: *Default = Blank*
  This field allows the SIP registrar address to be specified if it is different from that of the SIP proxy. The address can be specified as an IP address or DNS name.

Behaviour during Service unavailable

A proxy server is considered *Active* once the system has received a response to an INVITE, REGISTER or OPTIONS.

In the case of the proxy server responding with *503 - Service Unavailable*, it should be considered *Active - In Maintenance*. In this case, the following should occur:

- If the response *503 - Service Unavailable* was in response to an *INVITE* request:
  - If calls are tied to registrations (Calls Route via Registrar enabled) and there are other proxies available, the tied registrations should issue an *Un-REGISTER* and try to *REGISTER* with a different proxy. The call should fail with *cause = Temporary Fail*.
  - If calls are not tied, the *INVITE* should be immediately tried to a different proxy.
- If the response 503 - Service Unavailable was in response to a *REGISTER* request:
  - If there are other proxies available, this registration only should issue an *Un-REGISTER* and try to *REGISTER* with a different proxy.
- If Explicit DNS Server(s) are configured, a DNS request should be sent out to see whether the proxy server has disappeared from those being offered.
- An *Active-InMaintenance* proxy server should not be used for a new transactions (INVITE or REGISTER) until:
  - There is a change in DNS responses indicating the proxy has become active.
  - The configuration does not leave any better option available. In this case, there should be a throttle so that no more than 5 failures (without successes) in 1 minute should be allowed.
  - A config merge has occurred where the proxy string is changed.
  - 10 minutes has expired.

Behaviour during Not Responding

A proxy server that is not-responding (UDP) is indicated when 3 requests are sent and no replies are received. This would normally occur during a single INVITE transaction.

Consideration should be given whether this is caused by a local network fault or is caused by the Proxy being out of service. Since it is likely to be local, no action should be taken unless traffic is received from an alternative proxy while this proxy is actually not responding. The state should be "Possibly non responding".

If explicit DNS servers are configured, a DNS request should be sent out to see whether this Proxy server has disappeared from those being offered.

If possible, an alternative proxy should be stimulated simultaneously with stimulating the suspect server.

The server should be considered non-responding if it is persistently nonresponding while other proxies are responding or if it is non-responding and has disappeared from the DNS advertisement.

While in the "possibly not responding" state, it would be better to send an INVITE to an alternative proxy while simultaneously sending any appropriate message to this proxy. This will help to resolve whether it is really not responding rather than there being local network problems. However, there is no requirement to blacklist the proxy.

Once in the "definitely not responding" state:

- If there are other proxies available: this registration only should issue an Un-REGISTER, and try to REGISTER with a different proxy. Calls should not automatically clear.
- If a SIP message is received from it, the state should immediately go"Active".
- This proxy should be blacklisted unless there are no better options available. While blacklisted, only one transaction per 10 minutes is allowed.
- Even if not blacklisted, there should be a throttle so that no more than 5 failures (without successes) in 1 minute should be alowed.

5.4.10.4 SIP URI

Having setup the SIP trunk to the SIP ITSP, the SIP URI's registered with that ITSP are entered on this tab. A SIP URI (Uniform Resource Identifier) is similar to an internet email address, for example name@example.com, or 01555326978@example.com and represents the source or destination for SIP connection. The URI consists of two parts, the user part (eg. name) and the host part (eg. example.com).

- IP Office 5.0+: If the wildcard * is used in the SIP trunk's Local URI, Contact and Display fields, that SIP trunk will accept any incoming SIP call. The incoming call routing is still performed by the system incoming call routes based on matching the values received with the call or the URI's incoming group setting. Outgoing calls using the trunk will need to use another URI set against the trunk.

For the system, each SIP URI acts as a set of trunk channels. Outgoing calls can then be routed to the required URI by short codes that match that URI's Outgoing Group setting. Incoming calls can be routed by incoming call routes that match the URI's Incoming Group setting.

Note that the system only supports up to 150 URI entries on a SIP line.

Line | SIP URI	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	✗.

- Via
  This field is for information only and cannot be edited. It shows the IP address of the system LAN interface with which the SIP trunk is associated.
- Local URI: *Default = Use Authentication Name*
  This field sets the 'From' field for outgoing SIP calls using this URI. The value can either be entered manually or one of the following options can be selected.
  - *Use Authentication Name: Software level = 4.0 to 5.0.*
    Use the appropriate Authentication Name on the SIP Line tab as indicated by the Registration setting below.
  - *Use Credentials User Name: Software level = 6.0+.*
    Use the User Name from the SIP Credentials entry used for the call.
  - *Use Internal Data*
    Use the SIP Name value from the User | SIP 303 tab of the user making the call. IP Office 6.0: The system can also use SIP URI information configured for a hunt group (Hunt Group | SIP 322) or for the voicemail (System | Voicemail 157).
- Contact: *Default = Use Authentication Name*
  This field sets the 'Contact' field for SIP calls using this URI. The value can either be entered manually or one of the following options can be selected.
  - *Use Authentication Name: Software level = 4.0 to 5.0.*
    Use the appropriate Authentication Name on the SIP Line tab as indicated by the Registration setting below.
  - *Use Credentials User Name: Software level = 6.0+.*
    Use the User Name from the SIP Credentials entry used for the call.
  - *Use Internal Data*
    Use the SIP Name value from the User | SIP 303 tab of the user making the call. IP Office 6.0: The system can also use SIP URI information configured for a hunt group (Hunt Group | SIP 322) or for the voicemail (System | Voicemail 157).
- Display Name: *Default = Use Authentication Name*
  This field sets the 'Name' value for SIP calls using this URI. The value can either be entered manually or one of the following options can be selected.
  - *Use Authentication Name: Software level = 4.0 to 5.0.*
    Use the appropriate Authentication Name on the SIP Line tab as indicated by the Registration setting below.
  - *Use Credentials User Name: Software level = 6.0+.*
    Use the User Name from the SIP Credentials entry used for the call.
  - *Use Internal Data*
    Use the SIP Name value from the User | SIP 303 tab of the user making the call. IP Office 6.0: The system can also use SIP URI information configured for a hunt group (Hunt Group | SIP 322) or for the voicemail (System | Voicemail 157).
- PAI: *Default = None. Software level = 6.1*
  Select whether the PAI should be set and if so, the source of the value to use in the PAI field. The value can either be entered manually or one of the following options can be selected.

- *None*
  Do not use the PAI field.

- *Use Authentication Name: Software level = 4.0 to 5.0.*
  Use the appropriate Authentication Name on the SIP Line tab as indicated by the Registration setting below.

- *Use Credentials User Name: Software level = 6.0+.*
  Use the User Name from the SIP Credentials entry used for the call.

- *Use Internal Data*
  Use the SIP Name value from the User | SIP 303 tab of the user making the call. IP Office 6.0: The system can also use SIP URI information configured for a hunt group (Hunt Group | SIP 322) or for the voicemail (System | Voicemail 157).

- Registration: *Default = Primary*
  Pre-IP Office 6.0: This field sets whether the primary or secondary authentication name values set on the SIP Line tab should be used for calls by this SIP URI. IP Office 6.0: This field is used to select from a list of the account credentials configured on the line's SIP Credentials tab.

- Incoming Group ID: *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.

- Outgoing Group ID: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The system will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.

- Max Calls per Channel: *Default =10*
  This field sets the maximum number of simultaneous calls that can use the URI before the system returns busy to any further calls.

5.4.10.5 VoIP

- Compression Mode: *Default = Automatic Selection*
  This field defines the compression mode (codec) or modes offered during call setup. The supported codecs are *G.711 ALAW 64K*, *G.711 ULAW 64K*, *G.729(a) 8K CS-ACELP*, *G.723.1 6K3 MP-MLQ*.

  - *Automatic Select*
    This setting is used as follows:

- Pre-IP Office 5.0: *Automatic Select* uses the codecs in the order of preference *G729a*, *G711 ALAW*, *G711 ULAW* and *G.723.1*.

- IP Office 5.0+: The Automatic Codec Preference 160 setting (System | Telephony | Telephony 160) can be used to set which codec should be first in the list of preferred codecs. The remaining codecs are used in the order as above. Note that the G711 codecs are treated as a pair, so if one is selected as the first preference, the other will automatically be second in the list.

  - If required, a specific codec can be selected. However, if during call setup negotiation with a specific codec, connect fails, the system will fallback to using automatic selection.

  - Codecs not selected for a SIP connection are not used in codec negotiation. For H.323 it is not possible to exclude codecs from negotiation, only to change the order of codec preference.

  The Advanced button can be used to display a list of the codecs in their current order of preference. Codecs can be dragged up or down the list or deselected from the list. Note that once a list has been edited in this way, clicking Advanced again will return it to *Automatic Select*.

- Call Initiation Timeout: *Default = 4 seconds, Range = 1 to 99 seconds. Software level = 4.2+.*
  This option sets how long the system should wait for a response to its attempt to initiate a call before following the alternate routes set in an ARS 409 form.

- DTMF Support: *Default = RFC2833.*
  This setting is used to select the method by which DTMF key presses are signalled to the remote end. The supported options are *In Band*, *RFC2833* or *Info*.

- VoIP Silence Suppression: *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using *G711* between systems.

- Fax Transport Support: *Default = Off. Software level = 5.0+. IP500 and IP500v2 SIP lines.*
  This option is only available if Re-Invite Supported is selected. When enabled, the system performs fax tone detection on calls routed via the line and, if fax tone is detected, renegotiates the call codec as configured below. The SIP line provider must support the selected fax method and Re-Invite. The system must have available VCM resources using an IP500 VCM base card. For systems in a Small Community Network, fax relay 749 is supported for fax calls between the systems.

  - *None*
    Select this option if fax is not supported by the line provider.

  - *T38 (IP Office 6.0+)*
    T38 is supported for the sending and receiving of faxes on a SIP line if also supported for fax by the line provider.

- RE-Invite Supported: *Default = Off.*
  When enabled, Re-Invite can be used during a session to change the characteristics of the session, for example when the target of an incoming call or a transfer does not support the codec originally negotiated on the trunk. Requires the ITSP to also support Re-Invite.

- Use Offerer's Codec: *Default = Off. Software level = 4.2+.*
  Normally for SIP calls, the initiator of the calls sends a SIP offer which includes the codecs that they support in order of preference. The SIP response includes the codec they want to use for the call. This option can be used to override that behaviour and use the codec preference offered by the caller.

5.4.10.6 T38 Fax

The setting on this tab are only accessible if Re-invite Supported and Fax Transport Support are selected on the VoIP 240 tab. Fax relay 749 is only supported on IP500/IP500v2 systems with an IP500 VCM card.

SIP Line | T38 Fax	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✗, IP412 ✗, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✗.

- Use Default Values: *Default = On.*
  If selected, all the fields are set to their default values and greyed out.
- T38 Fax Version: *Default = 3.*
  The system can support Versions *0*, *1*, *2* and *3*. During fax relay, the two gateways will negotiate to use the highest version which they both support.
- Transport: *Default = UDPTL (fixed).*
  Currently only *UDPTL* is supported. *TCP* and *RTP* transport are not supported.
  - For *UDPTL*, redundancy error correction is supported. Forward Error Correction (FEC) is not supported.
- Redundancy:
  Redundancy sends additional fax packets in order to increase the reliability. However increased redundancy increases the bandwidth required for the fax transport.
  - Low Speed: *Default = 0 (No redundancy), Range = 0 to 5.*
    Sets the number of redundant T38 fax packets that should be sent for low speed V.21 T.30 fax transmissions.
  - High Speed: *Default = 0 (No redundancy). Range = 0 to 5.*
    Sets the number of redundant T38 fax packets that should be sent for V.17, V.27 and V.28 fax transmissions.
- TCF Method: *Default = Trans TCF.*
  TCF = Training Check Frame.
- Max Bit Rate (bps): *Default = 14400.*
  Lower rates can be selected if the current rate is not supported by the fax equipment or is found to not be reliable.
- EFlag Start Timer (msecs): *Default = 2600.*
- EFlag Stop Timer (msecs): *Default = 2300.*
- Tx Network Timeout (secs): *Default = 150.*
- Scan Line Fix-up: *Default = On.*
- TFOP Enhancement: *Default = On.*
- Disable T30 ECM: *Default = Off.*
  When selected, disabled the T.30 Error Correction Mode used for fax transmission.
- Disable EFlags For First DIS: *Default = Off.*
- Disable T30 MR Compression: *Default = Off.*
- NSF Override: *Default = Off.*
  If selected, the NSF (Non-Standard Facility) information sent by the T38 device can be overridden using the values in the fields below.
  - Country Code: *Default = 0.*
  - Vendor Code: *Default = 0.*

5.4.10.7 SIP Credentials

IP Office 6.0+. It is used to enter the ITSP username and password for the SIP account with the ITSP. If you have several SIP accounts going to the same ITSP IP address or domain name, you can enter up to 30 sets of ITSP account names and passwords on this tab.

Line | SIP Line	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	6.0+.
Mergeable	✓ (Mergeable for Release 6.1+.

Use the Add..., Edit... and Remove buttons to manage the set of credentials for the SIP trunk accounts. The settings for each account are:

- **Index**
  This number is assigned automatically and cannot be edited. If the From field on the SIP URI 238 being used for the call is set to Use Authentication Name , the registration field of the SIP URI will indicate the index number of the SIP credentials to use for calls by that SIP URI.
- **User Name**
  This name must be unique and is used to identify the trunk. The name can include the domain if necessary.
- **Authentication Name**: *Default = Blank.*
  This field can be blank but must be completed is a Password is also specified. This value is provided by the SIP ITSP . Depending on the settings on the Local URI tab associated with the SIP call, it may also be used as the user part of the SIP URI. The name can include the domain if necessary.
- **Contact**: *Default = Blank. Software level = 6.1.*
  This field is used to enter a contact and can include the domain if necessary.
- **Password**: *Default = Blank.*
  This value is provided by the SIP ITSP. If a password is specified the matching Authentication Name must also be set.
- **Expiry**: *Default = 60 minutes.*
  This setting defines how often registration with the SIP ITSP is required following any previous registration.
- **Registration Required**: *Default = On. Software level = 6.1*
  If selected, the fields above above are used for registration when making calls.

5.4.11 SES Line

SES lines are used for connection to an Avaya SIP Enablement Service (SES) server within an Avaya SIP for Branch network. This is a variant of TCP SIP line and requires SIP Trunk Channels licenses in the system configuration for the number of simultaneous SIP and or SES calls. The SES server supports up to 1000 branches arranged in a star network configuration.

With an Avaya SIP for Branch network, all extensions are reachable from any branch. Using short codes, the system should be configured to route calls for other branches in the network to the SES server via a SES line. The SES server acts as a SIP proxy. It holds a table of all branch prefixes and uses that table to reroute each call to the appropriate destination branch. The SES server will only route calls to the branch if the number match both the branch prefix and the expected extension number length.

Related Fields

There are a number of fields in the system configuration which are used with SES lines but are not on the SES Line form. The following are located on the System | System 143 tab:

- Branch Prefix: *Default = Blank, Range = 0 to 999999999. Software level = 4.1+.*
  Used to identify the system within an Avaya SIP for Branch network linked via an SES server. The branch prefixes of each systems within the network must unique and must not overlap. For example 85, 861 and 862 are okay, but 86 and 861 overlap.
- Local Number Length: *Default = Blank (Off), Range = Blank (Off) or 3 to 9. Software level = 4.1+.*
  Set the default length for extension numbers for extensions, users and hunt groups. Entry of an extension number of a different length will cause an error warning within Manager.This field is intended for systems being used in an Avaya SIP for Branch network linked via an SES server, where fixed extension lengths must be used. Note that the Branch Prefix and the Local Number Length must not exceed 15 digits.

Short Codes

Calls are routed to the SES line using short codes in the same was as for other line types. The short codes can route calls directly to the SES line's Outgoing Group ID or to an ARS form configured for the SES line. If at all possible, requiring a short code for each branch within the SIP for Branch Network should be avoided.

- Common Branch Prefix Leading Digit Routing
  If a common digit has been used at the start of all branch prefixes, that digit can be used as the key for a single SES routing short code. For example, first use a single range for branch prefixes, ie. 80 to 89 in a small network, 800 to 899 in a medium network or 8000 to 8999 in a large network. At each branch a single short code of the form 8N could then be used to start SES routing.
- Common Branch Prefix and Extension Length Routing
  If the above method cannot be used, maintaining a common branch prefix and local number length throughout the network is another option to simplify routing. For example, if all branches have a two digit branch prefixes and then 4 digit extension numbers, a short code of the form XXXXXX; could be used to only match dialed six digit numbers. Longer dialing is still possible so long as the first additional digit is dialed within the Dial Delay Time setting of the system.
- SES Line Prefix
  The SES line includes a prefix field that can be added to any calling number information supplied with calls received on a SES line. That prefix can then be used as the mechanism to route return calls back to the SES line. For example, if the line prefix is 3, a short code of the form 3N/N can be used.

Additional Notes

1. **Voice Compression Channels**
   SES calls use system voice compression channels in the same way as used for standard IP trunks and extensions. These are provided by the installation of VCM modules within the control unit. RTP relay is applied to SES calls where applicable.

2. **Licensing**
   SIP Trunk Channels licenses is required in the system configuration. These set the maximum number of simultaneous SIP and SES calls supported by the system. Multiple licenses can be added to achieve a cumulative maximum number of channels supported.

3. **Firewall**
   The system firewall between LAN1 and LAN2 is not applied to SES calls.

4. **Incoming Call Routing**
   Incoming SES calls are routed as if being internal calls. The dialed branch prefix is removed and the remaining extension number is used as if dialed on switch.

5. **DiffServ Marking**
   DiffServ marking is applied to calls using the **DiffServ Settings** on the System | LAN | VoIP 150 tab of either LAN1 or LAN2 as set by the line's **Use Network Topology Info** setting.

Resource Limitation
A number of limits can affect the number of SES calls. When one of these limits is reached the following occurs: any further outgoing SES calls are blocked unless some alternate route is available using ARS; any incoming SES calls are queued until the required resource becomes available.

Limiting factors are:

- the number of licensed SIP channels.
- the **Max Calls** setting of the SES line.
- the number of voice compression channels.
  - **SES Line Call to/from Non-IP Devices**
    Voice compression channel required.
  - **Outgoing SES Line Call from IP Device**
    No voice compression channel required.
  - **Incoming SIP Line Call to IP Device**
    Voice compression channel reserved until call connected.
- If the system also used SIP trunks, the same resources are shared between SES and SIP calls.

5.4.11.1 SES Line

Line | SES Line	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.1+.
Mergeable	✗.

- Line Number: *Default = Automatically assigned.*
  By default a value is assigned by the system. This value can be changed but it must be unique.

- Network Type: *Default = Public. Software level = 4.2+.*
  This option is available if Restrict Network Interconnect 160 (System | Telephony | Telephony 160) is enabled. It allows the trunk to be set as either *Public* or *Private*. The system will return number unobtainable indication to any attempt to connect a call on a *Private* trunk to a *Public* trunk or vice versa. This restriction includes transfers, forwarding and conference calls.

  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.

- SES Domain Name: *Default = Blank.*
  This field is used to enter the domain part of the SIP URI provided by the ITSP. For example, in the SIP URI mysip@example.com, the domain part of the URI is example.com. For outgoing calls the user part of the SIP URI is the Branch Prefix and the extension number.

- SES Address: *Default = 0.0.0.0*
  This value is the public IP address of the SES server. This address must not be shared by any other IP, IP DECT or SIP line in the system configuration.

- Inactivity Timeout (seconds): *Default = 120, Range = 0 to 99999 seconds.*
  If no SIP messages or signalling have been sent or received during this period the system will close the connection.

- Outgoing Group: *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The Manager will then seize a line with a matching Outgoing Group ID.

- Prefix: *Default = Blank.*
  This prefix will be added to any source number received with incoming calls. Normally this should match a dialing short code configured to route matching calls to the SES line's Outgoing Group ID number.
- Max Calls: *Default =10*
  This field sets the maximum number of simultaneous calls that can use the URI before the system returns busy to any further calls.

- In Service: *Default = On.*
  When this field is not selected, the SIP trunk is unregistered and not available to incoming and outgoing calls.

- Network Configuration

  - Layer 4 Protocol: *Default = TCP*
    This field cannot be changed.

  - Use Network Topology Info: *Default = LAN1*
    For various settings, this field indicates whether the SES line uses the System | LAN | Network Topology 152 settings of either LAN1 or LAN2. For instance the DiffServ Settings applied to outgoing SES calls and the IP address (see description of Via Nat below).

  - Send Port: *Default = 5060*
    This field cannot be changed.

  - Listen Port: *Default = 5060*
    This field cannot be changed.

  - Via NAT: *Default = Off.*
    This option controls which address should be used as the IP address of the system that is entered into the SES server configuration.

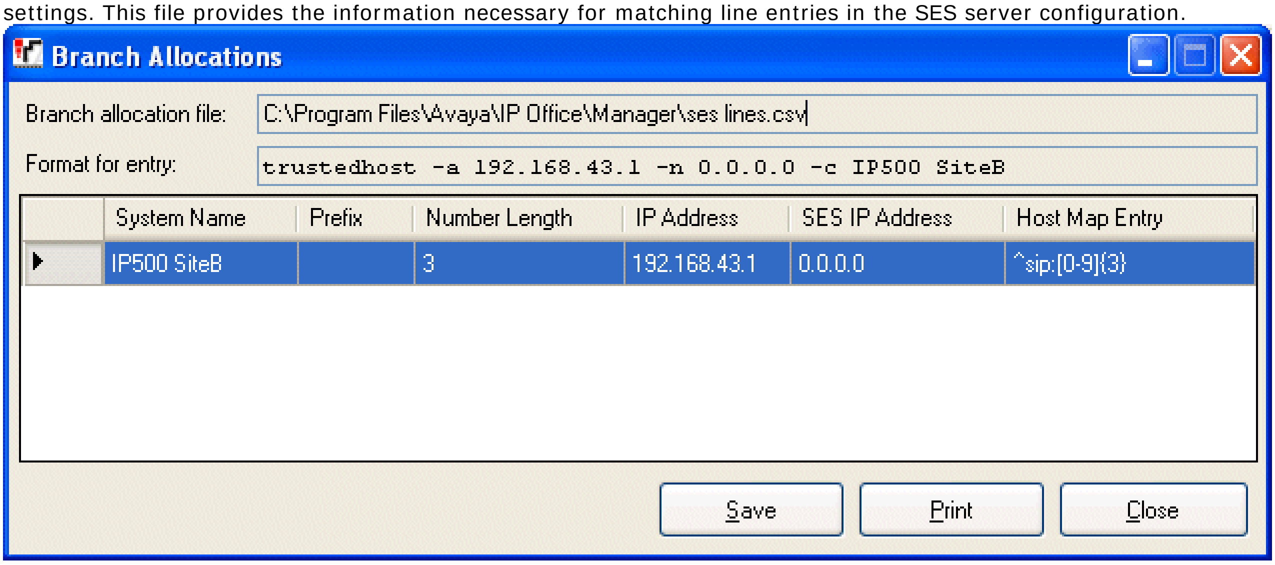
    - If disabled, the address used is the IP Address 149 of the system as set on the LAN Settings sub-tab of LAN1 or LAN2. The selection of LAN1 or LAN2 is determined by the Use Network Topology field above.

    - If enabled, the address used is the Public IP Address 149 as set on the Network Topology sub-tab of LAN1 or LAN2. The selection of LAN1 or LAN2 is determined by the Use Network Topology field above.

Prefix Allocation Check
As SES lines are configured within each system in the SIP for Branch network, details of the system can be written to a CSV file. This file can then be used to ensure the following:

1. Check branch allocation
   Check that the settings of the system and SES line being added are both complete and do not conflict with those of other systems already added to the CSV file. Select this option and click Execute , then select the CSV file containing details of the other systems with SES lines.

2. List branch allocations
   Provide the source of the details that need to be entered in the SES server configuration for each system. Select this option and click Execute to select and display the CSV file containing details of existing system SES line settings. This file provides the information necessary for matching line entries in the SES server configuration.



5.4.11.2 VoIP

Line | SES Line	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.1+.
Mergeable	✗.

- Gateway IP Address: *Default = Blank*
  Enter the IP address of the gateway device at the remote end. This address must not be shared by any other IP line (H323, SIP, SES or IP DECT).
- Compression Mode: *Default = Automatic Selection*
  This field defines the compression mode (codec) or modes offered during call setup. The supported codecs are *G.711 ALAW 64K*, *G.711 ULAW 64K*, *G.729(a) 8K CS-ACELP*, *G.723.1 6K3 MP-MLQ*.
  - *Automatic Select*
    This setting is used as follows:
- Pre-IP Office 5.0: *Automatic Select* uses the codecs in the order of preference *G729a*, *G711 ALAW*, *G711 ULAW* and *G.723.1.*
- IP Office 5.0+: The Automatic Codec Preference 160 setting (System | Telephony | Telephony 160) can be used to set which codec should be first in the list of preferred codecs. The remaining codecs are used in the order as above. Note that the G711 codecs are treated as a pair, so if one is selected as the first preference, the other will automatically be second in the list.
  - If required, a specific codec can be selected. However, if during call setup negotiation with a specific codec, connect fails, the system will fallback to using automatic selection.
  - Codecs not selected for a SIP connection are not used in codec negotiation. For H.323 it is not possible to exclude codecs from negotiation, only to change the order of codec preference.

  The Advanced button can be used to display a list of the codecs in their current order of preference. Codecs can be dragged up or down the list or deselected from the list. Note that once a list has been edited in this way, clicking Advanced again will return it to *Automatic Select*.
- Call Initiation Timeout: *Default = 4 seconds, Range = 1 to 99 seconds. Software level = 4.2+.*
  This option sets how long the system should wait for a response to its attempt to initiate a call before following the alternate routes set in an ARS 409 form.
- DTMF Support: *Default = RFC2833.*
  This setting is used to select the method by which DTMF key presses are signalled to the remote end. The supported options are *In Band*, *RFC2833* or *Info*.
- VoIP Silence Suppression: *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using *G711* between systems.
- Local Tones: *On (Fixed)*
  Indicates that the system will generate tones when required.
- Allow Direct Media Path: *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the system) or can be routed alternately if possible within the network structure.
  - If enabled, H323 calls can take routes other than through the system. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.
  - If disabled or not supported at on one end of the call, the call is routed via the system.
    - On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.
    - On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel.
- RE-Invite Supported: *Default = Off.*
  When enabled, Re-Invite can be used during a session to change the characteristics of the session, for example when the target of an incoming call or a transfer does not support the codec originally negotiated on the trunk. Requires the ITSP to also support Re-Invite.

- Use Offerer's Codec: *Default = Off. Software level = 4.2+.*
  Normally for SIP calls, the initiator of the calls sends a SIP offer which includes the codecs that they support in order of preference. The SIP response includes the codec they want to use for the call. This option can be used to override that behaviour and use the codec preference offered by the caller.

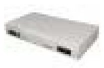
5.4.11.3 T38 Fax

The setting on this tab are only accessible if Re-invite Supported and Fax Transport Support are selected on the VoIP 247 tab. Fax relay 749 is only supported on IP500/IP500v2 systems with an IP500 VCM card.

SIP Line | T38 Fax	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✗, IP412 ✗, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✗.

- Use Default Values: *Default = On.*
  If selected, all the fields are set to their default values and greyed out.
- T38 Fax Version: *Default = 3.*
  The system can support Versions *0*, *1*, *2* and *3*. During fax relay, the two gateways will negotiate to use the highest version which they both support.
- Transport: *Default = UDPTL (fixed).*
  Currently only *UDPTL* is supported. *TCP* and *RTP* transport are not supported.
  - For *UDPTL*, redundancy error correction is supported. Forward Error Correction (FEC) is not supported.
- Redundancy:
  Redundancy sends additional fax packets in order to increase the reliability. However increased redundancy increases the bandwidth required for the fax transport.
  - Low Speed: *Default = 0 (No redundancy), Range = 0 to 5.*
    Sets the number of redundant T38 fax packets that should be sent for low speed V.21 T.30 fax transmissions.
  - High Speed: *Default = 0 (No redundancy). Range = 0 to 5.*
    Sets the number of redundant T38 fax packets that should be sent for V.17, V.27 and V.28 fax transmissions.
- TCF Method: *Default = Trans TCF.*
  TCF = Training Check Frame.
- Max Bit Rate (bps): *Default = 14400.*
  Lower rates can be selected if the current rate is not supported by the fax equipment or is found to not be reliable.
- EFlag Start Timer (msecs): *Default = 2600.*
- EFlag Stop Timer (msecs): *Default = 2300.*
- Tx Network Timeout (secs): *Default = 150.*
- Scan Line Fix-up: *Default = On.*
- TFOP Enhancement: *Default = On.*
- Disable T30 ECM: *Default = Off.*
  When selected, disabled the T.30 Error Correction Mode used for fax transmission.
- Disable EFlags For First DIS: *Default = Off.*
- Disable T30 MR Compression: *Default = Off.*
- NSF Override: *Default = Off.*
  If selected, the NSF (Non-Standard Facility) information sent by the T38 device can be overridden using the values in the fields below.
  - Country Code: *Default = 0.*
  - Vendor Code: *Default = 0.*

5.5 Control Unit



The Control Unit configuration form gives details for devices connected to the system. This includes some modules within the control unit as well as external expansion modules.

For most units, this information is allocated by the system and is not configurable.

The **New** and **Delete** actions on this form have special functions.

- **New**
  This action is used to added a WAN3 expansion module. If when a WAN3 is added to the system, the WAN3 is not recognized following a system reboot, New on this form can be used to scan for the WAN3 module.

- **Delete**
  This action can only be used with external expansion modules. It cannot be applied to the control unit. The action should used with caution as deleting a module will also delete any extensions or lines associated with that module. If the module is physically present, those entries will be recreated following a reboot but with default settings.

Control Unit | Control Unit	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✓.

- **Device Number**
  This is automatically allocated by the system.

- **Unit Type**
  The name of the device.

- **Version**
  The version of software running on each unit.

- **Serial Number**
  This is the number the system uses to tie a physical Control Unit to a device configuration (device number). For the control unit and WAN3 modules this is the MAC address. For a device connected to an Expansion port it is the Expansion port number plus 1.

- **Unit IP Address**
  This field shows the IP address for the control unit (LAN1) and if present, WAN3 expansion module.

- **Interconnect Number**
  For external expansion modules this is the control unit expansion port used for connection. For other devices this is 0.

- **Module Number**
  For external expansion modules this is the control unit expansion port used for connection. For internal devices in the control unit, **Control Unit** is displayed.

5.6 Extension Settings

The system supports both physical extensions and IP extensions.

By default, each extension is normally associated with a user and uses that user's directory number and other setting. Users with a log in code can move between extensions by logging in and out, so the directory number is not a fixed property of the extension.

- Physical Extensions
  Physical extension ports are either integral to the control unit or added by the installation of an analog or digital phone expansion module. Extension entries are automatically created for each physical extension port within the system. These ports cannot be added or deleted manually.

  -  Standard Telephone
    An analog extension port (PHONE or POT) or an Avaya digital station port (DS) within the system.

  -  Quiet Headset
    Used for analog extension devices that are permanently off-hook.

  -  IVR Port
    Used for analog ports connected to devices that require a specific disconnect clear signal at the end of each call.

  -  Paging Speaker
    An analog extension port set to be used as a paging speaker connection.

  -  FAX Machine
    Indicates that the extension is connected to a FAX machine.

  -  MOH Source
    Indicates that the extension is being used as a music on hold source.

- IP Extensions
  These are used for IP phone devices or applications.

  -  H323 or SIP Extension
    This icon indicates an IP extension. For Avaya IP hardphones the IP extensions is either added manually or by the automatic detection the phone being connected, refer to the Manager IP Phone Installation Manual. IP extensions can also be added manually to support a Phone Manager Pro PC Softphone or a third-party IP phone device. Note that third-party IP phone devices require entry of an IP End-Points license.

  -  IP DECT
    An extension port manually added to match extensions within an Avaya IP DECT system connected to the system via an IP DECT line.

For systems with users using the IP Office Video Softphone, an extension entry is created while the user is logged into the application. This entry is not editable. The softphone extension entry is automatically deleted within a few minutes of the user closing the softphone application.

5.6.1 Extn

This tab contains settings applicable to most types of extension.

Extension | Extn	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗. For Manager 4.1+, Base Extension, Disable Speakerphone and Reset Volume After Calls are mergeable.

- Extension ID
  The physical ID of the extension port. Except for IP extensions, this settings is allocated by the system and is not configurable.

- Base Extension: *Range = 2 to 9 digits.*
  This is the directory number of the extension's default associated user.

  - Following a restart, the system will attempt to log in the user with the same extension number (if they are not already logged in elsewhere in the Small Community Network). This does not occur it that user is set to Force Login (User | Telephone 274).

  - If another user logs onto an extension, when they log out, the extension returns to its default associated user unless they have logged in elsewhere or are set to Force Login.

  - In the US the Base Extension number is used for E911 calls. Any change to an extension's base extension must be matched by changes to the E911 adjunct database, see E911 Overview 421. The extensions default associated user should not be deleted.

  - Extensions associated with IP phones should not be given extension numbers greater than 7 digits.

  - For pre-IP Office 6.0, extension numbers in the range 8897 to 9999 are reserved for use by the Delta Server.

  - Users for Delta Server, CBC and CCC should only use up to 4 digit extension numbers.

- Caller Display Type: *Default = On.*
  Controls the presentation of caller display information for analog extensions, see Caller Display 755. For digital and IP extensions, this value is fixed as *On*. The table below lists the supported options, all others are currently not used and default to matching *UK*.

Type	Description
Off	Disables caller display.
On	Enables caller display using the caller display type appropriate to the System Locale, see Supported Country and Locale Settings 844. If a different setting is required it can be selected from the list of supported options. For an analog extension connected to a fax server or other device that requires the pass through of DTMF tones, select DTMFF.
UK	FSK before the first ring conforming to BT SIN 227. Name and number.
UK20	As per UK but with a maximum length of 20 characters. Name and number.
DTMFA	Caller ID in the DTMF pattern A<caller ID>C. Number only.
DTMFB	Caller ID in DTMF after call connection. Number only.
DTMFC	Caller ID in the DTMF pattern A<caller ID>#. Number only.
DTMFF	Sends the called number in DTMF after call connection. Number only. Used for fax servers. When calls are delivered via a hunt group it is recommended that hunt group queuing is not used. If hunt group queuing is being used, set the Queue Type to Assign Call on Agent Alert.
DTMFD	Caller ID in the DTMF pattern D<caller ID>C. Number only.
FSKA	Variant of UK used for BT Relate 1100 phones. Name and number.
FSKB	ETSI specification with 0.25 second leading ring. Name and number.
FSKC	ETSI specification with 1.2 second leading ring. Name and number.
FSKD	Conforms to Belcore specification. Name and number.

- Reset Volume after Calls: *Default = Off.*
  Resets the phone's handset volume after each call. This option is supported on Avaya 1400, 1600, 2400, 4400, 4600, 5400, 5600, 6400 and 9600 Series phones.

- Device Type
  This field indicates, the last known type of phone connected to the extension port.

  - Analogue extension ports always report as *Analog Handset* since the presence or absence of actual analog phone cannot be detected.

- Digital extension ports report the type of digital phone connected or *Unknown digital handset* if no phone is detected.
- H323 extensions report the type of IP phone registered or *Unknown H323 handset* if no phone is currently registered as that extension.
- SIP extensions report the type of SIP phone registered or *Unknown SIP device* if no SIP device is currently registered as that extension.
- For some types of phone, the phone can only report its general type to the system but not the specific model. When that is the case, the field acts as a drop-drown to allow selection of a specific model. The value selected here is also reported in other applications such as the System Status Application, SNMP, etc.

Default Type	Possible Phone Models
T7100	M7100, M7100N, T7100, Audio Conferencing Unit.
T7208	M7208, M7208N, T7208.
M7310	M7310, M7310N, T7406, T7406E.
M7310BLF	M7310BLF, T7316.
M7324	M7324, M7324N.

- Module
  This field indicates the external expansion module on which the port is located. *BP* indicates an analog phone extension port on the base or control unit. *BD* indicates a digital station (DS) port on the control unit. For an IP500/IP500v2 control unit, *BD* and *BP* is also followed by the slot number. VoIP extensions report as *0*.
- Port
  This field indicates the port number on the Module indicated above. VoIP extensions report as *0*.
- Disable Speakerphone: *Default = Off (Speakerphone enabled). Software level = 4.1+.*
  When selected, disables the fixed SPEAKER button if present on the phone using this extension port. Only supported on Avaya DS, TCM and H323 IP phones. An audible beep is sounded when a disabled SPEAKER button is pressed. Incoming calls such as pages and intercom calls are still connected but the speech path is not audible until the user goes off-hook using the handset or headset. Similarly calls made or answered using other buttons on the phone are not audible unless the user goes off-hook using the handset or headset. Currently connected calls are not affected by changes to this setting.
- Force Authorization: *Default = On. Software level = 5.0+, SIP Extensions only.*
  This setting is used with SIP extension devices.
-

5.6.2 Analog

This tab contains settings that are applicable to analog extensions. These extensions are provided by ports marked as *POT* or *PHONE* on control units and expansion modules.

Extension | Analog	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗. For IP Office 6.1+, Equipment Classification is mergeable.

- Equipment Classification: *Default = Standard Telephone*
  Only available for analog extension ports. Note that changing this settings should be followed by a system reboot. For IP Office 6.1 changes to this setting are mergeable.

  - *Quiet Headset*
    On extensions set to *Quiet Headset*, the audio path is disabled when the extension is idle. Ringing is presented in the audio path. Caller ID is not supported on the phone. This option can be used with analog extensions where the handset is replaced by a headset since in such a scenario audio is only desired when a call is connected. Since the audio path is disabled when idle, the *Quiet Headset* extension cannot dial digits to make calls. Therefore to make and answer calls this option is typically used with the user Offhook Station 274 (User | Telephony | Call Settings 274) setting which allows the extension user to make and answer calls using applications.

  - *Paging Speaker*
    Used for analog ports connected to a paging amplifier. This extension will present busy and cannot be called or be used to make calls. It can only be accessed using Dial Paging 459 features.

    - IP Office 4.0+: When using a UPAM connected to an analog extension port, the extension's Equipment Classification 254 (Extension | Analog) 254 should be set to *IVR Port* and not *Paging Speaker*.

  - *Standard Telephone*
    Use for normal analog phones.

  - *Door Phone 1/Door Phone 2*
    These two options are currently not used and so are grayed out.

  - *IVR Port*
    Used for analog ports connected to devices that require a disconnect clear signal (ie. a break in the loop current) at the end of each call. When selected the Disconnect Pulse Width is used. For pre-3.2 systems, this option was only supported on systems with the locale set to *United States* or *Saudi Arabia*.

  - *FAX Machine (IP Office 5+)*
    If fax Relay 749 is being used, this setting should be selected on any analog extension connected to an analog fax machine.

  - MOH Source *(IP Office 6.1+)*
    If selected, the port can be used as a music on hold source in the Tones and Music 163 settings. An extension set as a music on hold source cannot make or receive calls. The audio input can be monitored through the extension music on hold 763 controls. A suitable interface device is required to provide the audio input to the extension port. It must look to the system like an off-hook analog phone. For example a transformer with a 600 Ohm winding (such as a Bogen WMT1A) or a dedicated MoH device with a 600Ohm output designed for connection to a PBX extension port which is providing loop current can be used.

- Message Waiting Lamp Indication Type: *Default = None*
  Allows the selection of the message waiting indication (MWI) mode for analog and IP DECT extensions.

  - For control unit and Phone V1 module analog extensions, the options *None*, *On*, *51V Stepped*, *81V*, *Line Reversal A* and *Line Reversal B* are available.

  - For Phone V2 external module extensions and IP500 Phone base cards, the additional option *101V* is available.

  - *On* defaults the message waiting indication as follows using the system locale.

Locale	'On' =
Argentina, Australia, Brazil, Canada, Chile, China, Colombia, Japan, Korea, Mexico, New Zealand, Peru, Russia, Saudi Arabia, South Africa, Spain, United States, Venezuela.	*51V Stepped*
Bahrain, Belgium, Denmark, Egypt, Finland, France, Germany, Greece, Hong Kong, Hungary, Iceland, Italy, India, Kuwait, Morocco, Netherlands, Norway, Oman, Pakistan, Poland, Portugal, Qatar, Singapore, Sweden, Switzerland, Taiwan, Turkey, United Arab Emirates, United Kingdom.	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

- IP500v2: If the option Restrict Analog Extension Ringer Voltage is selected (System | Telephony | Telephony 160), the MWI options are restricted to *Line Reversal A*, *Line Reversal B* or *None*. Any extensions set to another option are forced to *Line Reversal A*.

- Hook Persistency: *Default = 100ms, Range = 50 to 255ms.*
Defines the time frame (in milliseconds) in which the system will wait before determining that the phone is off-hook.

- Flash Hook Pulse Width
The following options are only available for analog extension ports. They define the length of loop break that will be considered a time break recall (TBR) signal.

  - Use System Defaults: *Default = Selected (On)*
  Use the default values appropriate to the system's locale. See Appendix A: Locale Settings 844.

  - Minimum Width: *Range = 0 to 2540 milliseconds.*
  Minimum hook flash length used if Use System Defaults is not selected. Shorter breaks are ignored a glitches.

  - Maximum Width: *Range = 0 to 2550 milliseconds.*
  Maximum hook flash length used if Use System Defaults is not selected. Longer breaks are treated as clearing.

- Disconnect Pulse Width: *Default = 0ms, Range = 0 to 2550ms*
This setting is used with analog extensions where the Equipment Classification above has been set to *IVR Port*. It sets the length of loop current break used to indicate call clearing.

5.6.3 VoIP

This tab is only available for H323 and SIP extensions. The settings available will vary depending on the extension type:

- H323 IP Extension 256
- SIP Extension 258

H323 IP Extension

Extension | VoIP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- IP Address: *Default = 0.0.0.0*
  The IP address of the phone. The default entry accepts connection from any address. For phones using DHCP, the field is not updated to show the IP address being used by the phone except for T3 IP phones.

- MAC Address: *Default = 0000000000000 (Grayed out)*
  This field is grayed out and not used.

- Compression Mode: *Default = Automatic Selection*
  This field defines the compression mode (codec) or modes offered during call setup. The supported codecs are *G.711 ALAW 64K*, *G.711 ULAW 64K*, *G.729(a) 8K CS-ACELP*, *G.723.1 6K3 MP-MLQ*.

  - *Automatic Select*
    This setting is used as follows:

- Pre-IP Office 5.0: *Automatic Select* uses the codecs in the order of preference *G729a*, *G711 ALAW*, *G711 ULAW* and *G.723.1.*

- IP Office 5.0+: The Automatic Codec Preference 160 setting (System | Telephony | Telephony 160) can be used to set which codec should be first in the list of preferred codecs. The remaining codecs are used in the order as above. Note that the G711 codecs are treated as a pair, so if one is selected as the first preference, the other will automatically be second in the list.

  - If required, a specific codec can be selected. However, if during call setup negotiation with a specific codec, connect fails, the system will fallback to using automatic selection.

  - Codecs not selected for a SIP connection are not used in codec negotiation. For H.323 it is not possible to exclude codecs from negotiation, only to change the order of codec preference.

- Gain: *Default = Default. Software level = 3.0 to 4.1.*
  Allows adjustment of the volume. The gain is selectable from -31dB to +31dB in 1dB increments.

- TDM->IP Gain: *Default = Default (0dB). Range = -31dB to +31dB. Software level = 4.1 Q1 2008+.*
  Allows adjustment of the gain on audio from the system TDM interface to the IP connection.

- IP->TDM Gain: *Default = Default (0dB). Range = -31dB to +31dB. Software level = 4.1 Q1 2008+.*
  Allows adjustment of the gain on audio from the IP connection to the system TDM interface.

- Supplementary Services: *Default = H450. Software level = 4.0+.*
  Selects the supplementary service signaling method for use with non-Avaya IP devices. Options are *None*, *QSIG* and *H450*. For H450, hold and transfer are supported. Note that the selected method must be supported by the remote end.

- VoIP Silence Suppression: *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using *G711* between systems.

- Enable FastStart for non-Avaya IP Phones: *Default = Off*
  A fast connection procedure. Reduces the number of messages that need to be exchanged before an audio channel is created.

- Out of Band DTMF: *Default = On*
  When on, DTMF is sent as a separate signal ("Out of Band") rather than as part of the encoded voice stream ("In Band"). The "Out of Band" signaling is inserted back into the audio by the remote end. This is recommended for low bit-rate compression modes such as G.729 and G.723 where DTMF in the voice stream can become distorted. Switch off for T3 IP extensions.

  - For Avaya 1600, 4600, 5600 and 9600 Series phones the system will enforce the appropriate setting for the phone type.

  - For Avaya T3 IP phones, when Out-Of-Band is unchecked, the Allow Direct Media Path option is ignored and calls are via the system in order to provide tones.

- Local Tones: *Default = Off*
  When selected, the H.323 phones generate their own tones. This option is not supported by Avaya IP phones and Phone Manager Pro PC Softphone.

- Allow Direct Media Path: *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the system) or can be routed alternately if possible within the network structure.

  - If enabled, H323 calls can take routes other than through the system. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.

  - If disabled or not supported at on one end of the call, the call is routed via the system.

    - On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.

    - On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel.

    - T3 IP phones must be configured to 20ms packet size to use RTP relay. With the IP Office 4.0 Q2 2007 maintenance release, previous restrictions on T3 IP phones using direct media were removed. The phone must have firmware T246 or higher.

- VPN Phone Allowed: *Default = Off. Software level = 4.1 - 5.0.*
  Indicates that the extension can be used with Avaya VPNremote Phone firmware. The phones must be licensed using a VPN IP Extensions licenses added to the system configuration for the number of VPN IP phones required. Note that the field is grayed out if there are no available licenses. Note that this license is no longer used for IP Office 6.0, phones using VPNremote being licensed through standard Avaya IP Phone licenses.

- Reserve Avaya IP Endpoint License: *Software level = 6.0+.*
  Avaya IP phones require an Avaya IP Endpoint license 834. Normally licenses are issued in the order that devices register. This option allows this extension to be pre-licensed before the device has registered.

- Reserve 3rd Party IP Endpoint License: *Software level = 6.0+.*
  Non-Avaya IP phones require an 3rd Party IP Endpoint license 834. Normally licenses are issued in the order that devices register. This option allows this extension to be pre-licensed before the device has registered.

SIP Extension

Extension | VoIP (SIP)	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✗.

- **IP Address:** *Default = 0.0.0.0*
  The IP address of the phone. The default entry accepts connection from any address. If an address is entered, registration is only accepted from a device with that address.

- **Compression Mode:** *Default = Automatic Selection*
  This field defines the compression mode (codec) or modes offered during call setup. The supported codecs are *G.711 ALAW 64K*, *G.711 ULAW 64K*, *G.729(a) 8K CS-ACELP*, *G.723.1 6K3 MP-MLQ*.

  - ***Automatic Select***
    This setting is used as follows:

- Pre-IP Office 5.0: ***Automatic Select*** uses the codecs in the order of preference *G729a*, *G711 ALAW*, *G711 ULAW* and *G.723.1*.

- IP Office 5.0+: The **Automatic Codec Preference** 160 setting (System | Telephony | Telephony 160) can be used to set which codec should be first in the list of preferred codecs. The remaining codecs are used in the order as above. Note that the G711 codecs are treated as a pair, so if one is selected as the first preference, the other will automatically be second in the list.

  - If required, a specific codec can be selected. However, if during call setup negotiation with a specific codec, connect fails, the system will fallback to using automatic selection.

  - Codecs not selected for a SIP connection are not used in codec negotiation. For H.323 it is not possible to exclude codecs from negotiation, only to change the order of codec preference.

- **Fax Transport Support:** *Default = Off. Software level = 5.0+. IP500 and IP500v2 SIP lines.*
  This option is only available if **Re-Invite Supported** is selected. When enabled, the system performs fax tone detection on calls routed via the line and, if fax tone is detected, renegotiates the call codec as configured below. The SIP line provider must support the selected fax method and Re-Invite. The system must have available VCM resources using an IP500 VCM base card. For systems in a Small Community Network, fax relay 749 is supported for fax calls between the systems.

  - ***None***
    Select this option if fax is not supported by the line provider.

  - ***T38 (IP Office 6.0+)***
    T38 is supported for the sending and receiving of faxes on a SIP line if also supported for fax by the line provider.

- **Gain:** *Default = Default. Software level = 3.0 to 4.1.*
  Allows adjustment of the volume. The gain is selectable from -31dB to +31dB in 1dB increments.

- **TDM->IP Gain:** *Default = Default (0dB). Range = -31dB to +31dB. Software level = 4.1 Q1 2008+.*
  Allows adjustment of the gain on audio from the system TDM interface to the IP connection.

- **IP->TDM Gain:** *Default = Default (0dB). Range = -31dB to +31dB. Software level = 4.1 Q1 2008+.*
  Allows adjustment of the gain on audio from the IP connection to the system TDM interface.

- **DTMF Support:** *Default = RFC2833.*
  This setting is used to select the method by which DTMF key presses are signalled to the remote end. The supported options are ***In Band***, *RFC2833* or ***Info***.

- **VoIP Silence Suppression:** *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using *G711* between systems.

- **Local Hold Music:** *Default = Off*

- **Allow Direct Media Path:** *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the system) or can be routed alternately if possible within the network structure.

  - If enabled, H323 calls can take routes other than through the system. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.

  - If disabled or not supported at on one end of the call, the call is routed via the system.

    - On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.

  - On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel.

- **RE-Invite Supported:** *Default = Off.*
  When enabled, Re-Invite can be used during a session to change the characteristics of the session, for example when the target of an incoming call or a transfer does not support the codec originally negotiated on the trunk. Requires the ITSP to also support **Re-Invite**.

- **Use Offerer's Codec:** *Default = Off. Software level = 4.2+.*
  Normally for SIP calls, the initiator of the calls sends a SIP offer which includes the codecs that they support in order of preference. The SIP response includes the codec they want to use for the call. This option can be used to override that behaviour and use the codec preference offered by the caller.

- **Reserve Avaya IP Endpoint License:** *Software level = 6.0+.*
  Avaya IP phones require an Avaya IP Endpoint license 834. Normally licenses are issued in the order that devices register. This option allows this extension to be pre-licensed before the device has registered.

- **Reserve 3rd Party IP Endpoint License:** *Software level = 6.0+.*
  Non-Avaya IP phones require an 3rd Party IP Endpoint license 834. Normally licenses are issued in the order that devices register. This option allows this extension to be pre-licensed before the device has registered.

5.6.4 T38 Fax

The setting on this tab are only accessible if Re-invite Supported and Fax Transport Support are selected on the VoIP 256 tab. Fax relay 749 is only supported on IP500/IP500v2 systems with an IP500 VCM card.

SIP Extension | T38 Fax	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✗, IP412 ✗, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✗.

- Use Default Values: *Default = On.*
  If selected, all the fields are set to their default values and greyed out.
- T38 Fax Version: *Default = 3.*
  The system can support Versions *0*, *1*, *2* and *3*. During fax relay, the two gateways will negotiate to use the highest version which they both support.
- Transport: *Default = UDPTL (fixed).*
  Currently only *UDPTL* is supported. *TCP* and *RTP* transport are not supported.
  - For *UDPTL*, redundancy error correction is supported. Forward Error Correction (FEC) is not supported.
- Redundancy:
  Redundancy sends additional fax packets in order to increase the reliability. However increased redundancy increases the bandwidth required for the fax transport.
  - Low Speed: *Default = 0 (No redundancy), Range = 0 to 5.*
    Sets the number of redundant T38 fax packets that should be sent for low speed V.21 T.30 fax transmissions.
  - High Speed: *Default = 0 (No redundancy). Range = 0 to 5.*
    Sets the number of redundant T38 fax packets that should be sent for V.17, V.27 and V.28 fax transmissions.
- TCF Method: *Default = Trans TCF.*
  TCF = Training Check Frame.
- Max Bit Rate (bps): *Default = 14400.*
  Lower rates can be selected if the current rate is not supported by the fax equipment or is found to not be reliable.
- EFlag Start Timer (msecs): *Default = 2600.*
- EFlag Stop Timer (msecs): *Default = 2300.*
- Tx Network Timeout (secs): *Default = 150.*
- Scan Line Fix-up: *Default = On.*
- TFOP Enhancement: *Default = On.*
- Disable T30 ECM: *Default = Off.*
  When selected, disabled the T.30 Error Correction Mode used for fax transmission.
- Disable EFlags For First DIS: *Default = Off.*
- Disable T30 MR Compression: *Default = Off.*
- NSF Override: *Default = Off.*
  If selected, the NSF (Non-Standard Facility) information sent by the T38 device can be overridden using the values in the fields below.
  - Country Code: *Default = 0.*
  - Vendor Code: *Default = 0.*

5.6.5 IP DECT

This tab is displayed for IP DECT extensions. These are created manually after an IP DECT 227 line has been added to the configuration or added automatically as DECT handsets subscribe to the DECT system.

Extension | IP DECT	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.1+.
Mergeable	✗.

- DECT Line ID
  Use the drop-down list to select the IP DECT line 227 from the system to the Avaya IP DECT system.

- Message Waiting Lamp Indication Type: *Default = On*
  Allows selection of the message waiting indication to use with the IP DECT extension. Options are: *None*, *On*.

- Reserve Avaya IP Endpoint License: *Software level = 6.0+.*
  Avaya IP phones require an Avaya IP Endpoint license in order to register with the system. Normally licenses are issued in the order that devices register. This option allows this extension to be pre-licensed before the device has registered.

5.7 User Settings

Users are the people who use the system. They do not necessary have to be an extension user, for example users are used for RAS dial in data access. In addition, more users can be created than there are extensions, with users logging in to an extension when they want to receive calls.

By default, a user is automatically created to match each extension. They are numbered from 201 upwards and the first 16 are placed in the hunt group Main (200), which is the default destination for incoming calls.

- Standard User: A standard user.
- No User: Used to apply settings for extensions which currently have no associated user.
- Remote Manager: Used as the default settings for dial in user connections.
- Hot Desking User: Users with a Login Code can move between extensions by logging in and off.

When a user is deleted

When a user is deleted, any calls in progress continue until completed. The ownership of the call is shown as the NoUser user. Merging the deletion of a user causes all references to that deleted user to be removed from the system.

Changing a user's extension

Changing a user's extension number automatically logs the user in on the matching base extension if available and the user doesn't have Forced Login 274 enabled. If Forced Login is enabled, then the user remains on the current extension being used until they log out and log in at the new extension.

Note that changing a user's extension number affects the user's ability to collect Voicemail messages from their own extension. Each user's extension is set up as a "trusted location" under the Source Numbers tab of the User configuration form. This "trusted location" allows the user to dial *17 to collect Voicemail from his own extension. Therefore if the extension number is changed so must the "trusted location".

The following related configuration items are automatically updated when a user extension is changed:

- User, Coverage and Bridged Appearance buttons associated with the user.
- Hunt group membership (disabled membership state is maintained).
- Forwards and Follow Me's set to the user as the destination.
- Incoming call routes to this destination.
- Dial in source numbers for access to the user's own voicemail.
- Direct call pickup buttons are updated.
- The extension number of an associated extension is updated.

Creating a User Rights Based on an Existing User

1. Select User Rights.
2. In the group pane, right-click and select New User Rights from a User.
3. Select the user and click OK.

Associating User Rights to a User

1. Select User Rights or User.
2. In the group pane, right-click and select Apply User Rights to Users.
3. Select the user rights to be applied.
4. On the Members of this User Rights sub tab select the users to which the user rights should be applied as their Working Hours User Rights.
5. On the Members when out of hours sub tab select which users should use the selected user rights as their out of hours user rights.
6. Click OK.

or

1. Select the required user to display their settings in the details pane.
2. Select the User tab.
3. Use Working Hours User Rights drop-down to select the user rights required.
4. If required a Working Hours Time Profile and Out of Hours User Rights can be selected.
5. Click OK.

Copy User Rights Settings over a User's Settings

This process replaces a user's current settings with those that are part of the selected user rights. It does not associate the user with the user rights.

1. Select User Rights.
2. In the group pane, right-click and select Copy user rights values to users.
3. Select the user rights to be applied.
4. Click OK.

5.7.1 User

Users are the people who use the system or are Dial In users for data access. A system User may or may not have an Extension Number that physical exists - this is useful if users do not require a physical extension but wish to use system features, for example voicemail, forwarding etc.

- **NoUser** is used to apply settings to extensions which have no associated user.
- **Remote Manager** is used as the default settings for dial in connections.

User | User	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the **Working Hour User Rights** 264 and **Out of Hours User Rights** 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's **Working Hours Time Profile** 264 setting. The effect of the user rights can be displayed using the **User Right View** 264 control.

- **Name**: *Range = Up to 15 characters*
  This is the user's account name used for RAS Dial In, Caller Display and voicemail mailbox. As the display on Caller Display telephones is normally only 16 digits long it is useful to keep the name short. Only alphanumeric characters and space are supported in this field. Do not use punctuation characters such as #, ?, /, -, _ and ,. Start names with a alphabetic character. This field is case sensitive and must be unique.
  - Voicemail uses the name to match a user to their mailbox. Changing a user's name will route their voicemail calls to a new mailbox. Note however that Voicemail Pro is not case sensitive and will treat names such as "Steve Smith", "steve smith" and "STEVE SMITH" as being the same.
- **Password**: *Default = Blank, Range = Up to 31 alphanumeric characters.*
  This password is used by user applications such as Phone Manager, SoftConsole and TAPI. It is also used for user's with Dial In access. Note that this is not the user's voicemail mailbox password (see User | Voicemail | Voicemail Code 267) or their phone log in code (see User | Telephony | Supervisor Settings | Login Code 276).
  - **IP Office 4.2+**: Users can change their Password using the *File | **Change Password*** option within Phone Manager.
- **Full Name**: *Default = Blank*
  Use this field to enter the user's full name. The recommended format is <first name><space><last name> in order for this value to be used correctly by voicemail dial by name features. When set, the **Full Name** is used in place of the **Name** for display by phones and user applications. Only alphanumeric characters and spaces are supported in this field. Do not use punctuation characters such as #, ?, /, -, _ , ^, > and ,. The entry in this field also cannot start with either a space or a numeral.
- **Extension**: *Range = 2 to 9 digits.*
  Any number up to 9 digits. In general all extensions should have the same number of digits. This setting can be left blank for users used just for dial in data connections.
  - Users for Delta Server, CBC and CCC should only use up to 4 digit extension numbers.
  - Users associated with IP phones or who may log in as such devices should not be given extension numbers greater than 7 digits.
  - For pre-IP Office 6.0, extension numbers in the range 8897 to 9999 are reserved for use by the Delta Server.
- **Locale**: *Default = Blank (Use system locale 143)*
  Configures the language used for voicemail prompts played to the user, assuming the language is available on the voicemail server. See Supported Country and Locale Settings 844. On a digital extension it also controls the display language used for messages from the system. Note however that some phones have their own menu options for the selected language for the phone menus.
  - When the system routes a call to the voicemail server it indicates the locale for which matching prompts should be provided if available. The locale sent to the voicemail server by the system is determined as follows:

Locale Source	Usage
Short Code	The short code locale, if set, is used if the call is routed to voicemail using the short code.
System	If no user or incoming call route locale is set system locale is used unless overridden by a short code locale.
Incoming Call Route	The incoming call route locale, if set, is used if caller is external.
User	The user locale, if set, is used if the caller is internal.

- Priority: *Default = 5, Range = 1 (Lowest) to 5 (Highest)*
  Pre-IP Office 4.0: This setting is used by Least Cost Routing 370. IP Office 4.0+: This setting is used by ARS 407.

- System Phone Rights: *Default = None.*
  This option replaces the System Phone 274 option used on pre-IP Office 6.0 systems. Users set as a system phone user are able to access additional functions 765.

  - None
    The user cannot access any system phone options.

  - Level 1
    The user can access all system phone options supported on the type of phone they are using except system management and memory card commands.

  - Level 2
    The user can access all system phone options supported on the type of phone they are using including system management and memory card commands. Due to the nature of the additional commands a login code should be set for the user to restrict access.

- Profile : *Default = Basic User. Software level = 6.0+.*
  A user's configured profile controls whether they can be configured for a number of features. The table below lists the different user profiles and the settings accessible by each profile. Setting a user to a particular profile will enable those settings by default, however they can still be manually disabled if required. The number of users that can be configured for each profile other than *Basic User* is controlled by the user licenses 835 present in the configuration.

	Basic User	Office Worker	Teleworker	Mobile Worker	Power User
one-X Portal for IP Office	Yes[1]	Yes	Yes	–	Yes
" Telecommuter options	Yes[1]	–	Yes	–	Yes
UMS Web Services	Yes[1]	Yes	Yes	–	Yes
Mobility Features	Yes[1]	–	–	Yes	Yes
TTS for Email Reading	–	–	–	Yes	Yes
IP Office SoftPhone	–	–	Yes	–	Yes

  1. These features are supported for Basic User users on systems with the appropriate pre-IP Office 6.0 legacy licenses.

- Receptionist : *Default = Off. Software level = 6.0+.*
  This settings allows the user to use the SoftConsole application. This requires the Manager configuration to contain Receptionist licenses 830. Up to 4 users can be licensed.

- Enable Softphone : *Default = Off. Software level = 6.0+.*
  This option can be enabled for users whose Profile 265 is set to *Teleworker* or *Power User*. If selected, the user is able to use the IP Office Softphone application.

- Enable one-X Portal Services: *Default = Off. Software level = 5.0+.*
  This option can be enabled for users whose Profile 265 is set to *Office Worker*, *Teleworker* or *Power User*. If selected, the user is able to use the one-X Portal for IP Office application to access their phone settings and to control phone calls.

  - For systems upgraded to IP Office 6.0 with existing licenses for one-X Portal for IP Office, those licenses can be used with users whose Profile is set to *Basic User*.

- Enable one-X TeleCommuter: *Software level = 6.0+.*
  This option can be enabled for users whose Profile 265 is set to *Teleworker* or *Power User*. If selected, the user is able to use the telecommuter mode features of the one-X Portal for IP Office application.

- Ex Directory: *Default = Off*
  When on, the user does not appear in the directory list shown by the user applications and on phones with a directory function.

- Restrictions: *Default = None. Software level = Up to 3.1.*
  Sets which set of User Restrictions 387 applies to the user.

- Phone Manager Type: *Default = Lite. Software level = Up to 3.1.*
  Determines the mode in which the user's copy of the Phone Manager application will operate. Modes are Lite, Pro and VoIP (Phone Manager Pro PC Softphone). Note that the number of users able to simultaneously use Pro and VoIP modes is controlled by licenses entered into the system configuration. *In IP Office 3.2 configurations this option has moved to the Phone Manager Options tab.

- Book a Conferencing Center in Phone Manager: *Default = Off. Software level = Up to 3.1.*
  When enabled, displays links in the user's Phone Manager application for access to the Conferencing Center application if installed. Note that to book a conference requires the user to have a Conferencing Center user ID and password. This feature also requires the Conferencing Center IP Address and Conferencing Center URL to be set (System | System).
  *In IP Office 3.2 configurations, this option has moved to the Phone Manager Options tab.

- Device Type:
  This field shows the type of phone at which the user is current logged in. If the user is logged out but is associated with a Base Extension, the device type for that extension port is shown. If the user has been logged out and is not associated with a Base Extension, the device type is listed as *Device Type Unknown*.

- User Rights View: *Software level = 3.2+.*
  This field affects Manager only. It allows you to switch between displaying the user settings as affected by their associated Working Hours User Rights or Out of Hours User Rights.

- Working Hours Time Profile: *Default = <None> (Continuous). Software level = 3.2+.*
  If set, the selected time profile defines when the user's Working Hours User Rights are applied. Outside the time profile, the user's Out of Hours User Rights are applied.

- Working Hours User Rights: *Default = Blank (No rights restrictions). Software level = 3.2+.*
  This field allows selection of user rights which may set and lock some user settings. If a Working Hours Time Profile has been selected, the Working Hours User Rights are only applied during the times defined by that time profile, otherwise they are applied at all times.

- Out of Hours User Rights: *Default = Blank (No rights restrictions). Software level = 3.2+.*
  This field allows selection of alternate user rights that are used outside the times defined by the user's Working Hours Time Profile.

5.7.2 Voicemail

If a voicemail server application is being used on your system, each user has use of a voicemail mailbox. You can use this form to enable this facility and various user voicemail settings.

User | Voicemail	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A 🔒 symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Voicemail Code: *Default = Blank, Range = 0 (no code) to 15 digits.*
  A code (1-15 digits) used by the voicemail server to validate access to this mailbox. If remote access is attempted to a mailbox that has no voicemail code set, the prompt "Remote access is not configured on this mailbox" is played. Whether the caller will be prompted to enter this code varies as follows:
  - Embedded Voicemail
    The voicemail code is used if set.
  - Trusted Source Access
    The voicemail code is required when accessing the mailbox from a location that is not set as a trusted number in the user's Source Numbers 272 list. Also Voicemail Pro call flows containing an action where the action's PIN code set to $ will prompt the user for their voicemail code.
    - Users can set their own code through the mailbox telephone user interface. In this case the user is forced to enter at least 4 digits.
    - Codes set through the Voicemail Pro telephone user interface are restricted to valid sequences. For example, attempting to enter a code that matches the mailbox extension, repeat the same number (1111) or a sequence of numbers (1234) are not allowed. If these types of code are required they can be entered through Manager.
  - Changing the Code
    All of the voicemail interfaces, except IMS and IMAP, provide options for the user to change the voicemail code themselves. In addition, Voicemail Pro running in Intuity emulation mode will request that the user sets a code when they first log in to their mailbox using the phone.
- Voicemail On *Default = On*
  When on, the mailbox is used by the system to answer the user's unanswered calls or calls when the user's extension returns busy. Note that selecting off does not disable use of the user's mailbox. Messages can still be forward to their mailbox and recordings can be placed in it. The mailbox can also still be accessed to collect messages.
  - When a caller is directed to voicemail to leave a message, the system indicates the target user or hunt group mailbox.
    - Pre-IP Office 4.0: The mailbox of the user or hunt group whose settings have caused the call to go to voicemail is used.
    - For IP Office 4.0+: The mailbox of the originally targeted user or hunt group is used. This applies even if the call has been forwarded to another destination. It also includes scenarios where a hunt group call overflows or is in fallback to another group.
    - Voicemail Pro can be used to customize which mailbox is used separately from the mailbox indicated by the system.
- Voicemail Help *Default = Off*
  For voicemail systems running IP Office mailbox mode, this option controls whether users retrieving messages are automatically given an additional prompt *"For help at any time press 8."* If switched off, users can still press 8 for help. For voicemail systems running in Intuity emulation mode, this option has no effect. On those systems the default access greeting always includes the prompt *"For help at any time, press *4"* (*H in the US locale).
- Voicemail Ringback: *Default = Off* 🔒
  When enabled and a new message has been received, the voicemail server calls the user's extension to attempt to deliver the message each time the telephone is put down. Voicemail will not ring the extension more than once every 30 seconds.
- Voicemail Email Reading: *Default = Off*
  This option can be enabled for users whose Profile 265 is set to *Mobile Worker* or *Power User*. If enabled, when you log into you voicemail box, it will detect your email messages and read them to you. This email text to speech feature is set-up through Voicemail Pro.

- UMS Web Services: *Default = Off. Software level = 4.2+.*
  This option can be enabled for users whose Profile 265 is set to *Teleworker*, *Office Worker* or *Power User*. If selected, the user can use either any of the Voicemail Pro UMS services to access the voicemail messages (IMAP email client, web browser or Exchange 2007 mailbox). Note that the user must have a voicemail code set in order to use the UMS services.

    - For systems upgraded to IP Office 6.0 with existing UMS Web Service licenses, those licenses can be used with users whose Profile is set to *Basic User*.

- Voicemail Email: *Default = Blank (No voicemail email features)*
  This field is used to set the user or group email address used by the voicemail server for voicemail email operation. When an address is entered, the additional Voicemail Email control below are selectable to configure the type of voicemail email service that should be provided.

  - Use of voicemail email requires the Voicemail Pro server to have been configured to use either a local MAPI email client or an SMTP email server account. For embedded voicemail, voicemail email is supported with IP Office 4.2+ (except not on Small Office Edition) and uses the system's SMTP 177 settings.

  - The use of voicemail email for the sending (automatic or manual) of email messages with wav files attached should be considered with care. A one-minute message creates a 1MB .wav file.

- Voicemail Email *Default = Off*
  If an email address is entered for the user or group, the following options become selectable. These control the mode of automatic voicemail email operation provided by the voicemail server whenever the voicemail mailbox receives a new voicemail message.

    - Users can change their voicemail email mode using visual voice. If the voicemail server is set to IP Office mode, user can also change their voicemail email mode through the telephone prompts. The ability to change the voicemail email mode can also be provided in a call flow using a Play Configuration Menu action or a Generic action.

    - If the voicemail server is set to IP Office mode, users can manually forward a message to email.

  - *Off*
    If off, none of the options below are used for automatic voicemail email. Users can also select this mode by dialing *03 from their extension.

  - *Copy*
    If this mode is selected, each time a new voicemail message is received in the voicemail mailbox, a copy of the message is attached to an email and sent to the email address. There is no mailbox synchronization between the email and voicemail mailboxes. For example reading and deletion of the email message does not affect the message in the voicemail mailbox or the message waiting indication provided for that new message.

  - *Forward*
    If this mode is selected, each time a new voicemail message is received in the voicemail mailbox, that message is attached to an email and sent to the email address. No copy of the voicemail message is retained in the voicemail mailbox and their is no message waiting indication. As with Copy, their is no mailbox synchronization between the email and voicemail mailboxes. Users can also select this mode by dialing *01 from their extension.

    - UMS Exchange 2007 *(IP Office 5.0+)*
      With Voicemail Pro, the system supports voicemail email to an Exchange 2007 server email account. For users and groups also enabled for UMS Web Services this significantly changes their mailbox operation. The Exchange Server inbox is used as their voicemail message store and features such as message waiting indication are set by new messages in that location rather than the voicemail mailbox on the voicemail server. Telephone access to voicemail messages, including Visual Voice access, is redirected to the Exchange 2007 mailbox.

  - *Alert*
    If this mode is selected, each time a new voicemail message is received in the voicemail mailbox, a simple email message is sent to the email address. This is an email message announcing details of the voicemail message but with no copy of the voicemail message attached. Users can also select this mode by dialing **02* from their extension.

- DTMF Breakout
  When a caller is directed to voicemail to leave a message, they can be given the option to be transferred to a different extension. The greeting message needs to be recorded telling the caller the options available. The extension numbers that they can be transferred to are entered in the fields below. IP Office 5+: System default values can be set for these numbers and are used unless a different number is set within these user settings. The values can be set using User Rights 399.

  - Reception / Breakout (DTMF 0)
    The number to which a caller is transferred if they press *0* while listening to the mailbox greeting rather than leaving a message (**0* on embedded voicemail).

    - For systems set to Intuity emulation mode, the mailbox user can also access this option when collecting their messages by dialing **0*.

    - If the mailbox has been reached through a call flow containing a Leave Mail action, the option provided when *0* is pressed are:

  - For IP Office mode, the call follows the Leave Mail action's *Failure* or *Success* results connections depending on whether the caller pressed *0* before or after the record tone.
  - For Intuity mode, pressing *0* always follows the Reception / Breakout (DTMF 0) setting.

- Breakout (DTMF 2)
  The number to which a caller is transferred if they press 2 while listening to the mailbox greeting rather than leaving a message (*2 on embedded voicemail). Pre-IP Office 5 this option is not support for Voicemail Pro running in IP Office mailbox mode.

- Breakout (DTMF 3)
  The number to which a caller is transferred if they press *3* while listening to the mailbox greeting rather than leaving a message (*3 on embedded voicemail). Pre-IP Office 5 this option is not support for Voicemail Pro running in IP Office mailbox mode.

5.7.3 DND

Do not disturb prevents the user from receiving hunt group and page calls. Direct callers hear busy tone or are diverted to voicemail if available. It overrides any call forwarding, follow me and call coverage settings. A set of exception numbers can be added to list numbers from which the user still wants to be able to receive calls when they have do not disturb in use. See Do Not Disturb 769 in the Telephone Features section for full details of Do Not Disturb operation.

User | DND	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A 🔒 symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Do Not Disturb: *Default = Off* 🔒
  When checked the user's extension is considered busy, except for calls coming from sources listed in their Do Not Disturb Exception List. When a user has do not disturb in use, their normal extension will give alternate dialtone 846 when off hook. IP Office 4.2+: Users with DND on are indicated as 'busy' on any BLF indicators set to that user.

- Do Not Disturb Exception List: *Default = Blank*
  This is the list of telephone numbers that are still allowed through when Do Not Disturb is set. For example this could be an assistant or an expected phone call. Internal extension numbers or external telephone numbers can be entered. If you wish to add a range of numbers, you can either enter each number separately or make use of the wildcards "N" and "X" in the number. For example, to allow all numbers from 7325551000 to 7325551099, the DND Exception number can be entered as either 73255510XX or 73255510N. Note that this list is only applied to direct calls to the user.

  - Calls to a hunt group of which the user is a member do not use the Do Not Disturb Exceptions list.

5.7.4 Short Codes

Short codes entered in this list can only be dialed by the user. They will override any matching user rights or system short code. See Short Codes 426 for details.

User and User Rights short codes are only applied to numbers dialed by that user. For example they are not applied to calls forwarded via the user.

- **WARNING**
  User dialing of emergency numbers must not be blocked by the addition of short codes. If short codes are added, the users ability to dial emergency numbers must be tested and maintained.

User | Short Codes	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A 🔒 symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

- ***FWD**
  Short codes of this form are inserted by the system. They are used in conjunction with the **User | Forwarding** settings to remember previously used forwarding numbers. They can be accessed on that tab by using the drop-down selector on the forwarding fields.

5.7.5 Source Numbers

This form is used to enter values that have special usages. These are entered using the **Add**, **Edit** and **Remove** buttons.

User | Source Numbers	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

The following types of entry can be added to a user's source numbers.

- **V<Caller's ICLID>**
  For systems using Voicemail Lite, Voicemail Pro or Embedded Voicemail, strings prefixed with a V indicate numbers from which access to the users mailbox is allowed without requiring entry of the mailbox's voicemail code. This is referred to as "trusted source".
  - For Voicemail Pro running in Intuity mode, trusted source is used for calls from programmable buttons set to Voicemail Collect and Visual Voice. Other controls are prompted for the mailbox number and then password.
- **R<Caller's ICLID>**
  To allow Dial In/RAS call access only from a specified number prefix the number with a "R", for example *R7325551234* .
- **H<Group Name>**
  Allows the user to receive message waiting indication of new group messages. Enter H followed by the group name, for example *HMain*.
  - On suitable display extensions, the hunt group name and number of new messages is displayed. Refer to the appropriate telephone user guide.
  - If the user is using Phone Manager, the Messages tab shows the hunt group name and number of new messages.
  - If the user is not a member of the group, a voicemail code must be set for the group's mailbox. See Voicemail Code on the Hunt Group | Voicemail 311 tab.
- **P<Telephone Number>**
  This entry sets the destination for callback (outbound alert) calls from voicemail. Enter P followed by the telephone number including any necessary external dialing prefix, for example P917325559876. This facility is only available when using Voicemail Pro through which a default Callback or a user specific Callback start point has been configured. Refer to the Voicemail Pro documentation. This feature is separate from voicemail ringback and Voicemail Pro outcalling.
- **RESERVE_LAST_CA=** *Software level = Up to 3.2.*
  Used for users with multiple call appearance buttons. When present, this string stops the users last call appearance button from being used to receive incoming calls. This ensures that the user always has a call appearance button available to make outgoing calls and to initiate transfers and conferences. For IP Office 4.0 and higher this option has been replaced by Reserve Last CA 278 (User | Telephony | Multi-line Options 278).
- **AT<string>**
  Strings beginning with AT can be used with a user called *DTEDefault* to configure the default settings of the control unit's DTE port.
- **Enable_OTT**
  Enable one touch transfer 786 operation for the user.

NoUser User Source Numbers
The following source numbers can also be used on the **Source Numbers** tab of the *NoUser* 767 user. These affect all users on the system. Note that changes to these source numbers require a reboot of the system to become effective.

- **ACD_QUEUE_DELAY=***nn Software level = Up to 3.2.*
  Used to change the timeout for still queued messages. The parameter *nn* can be replace with a time in seconds between 20 and 180. For IPOffice 4.0+ this has been replaced by Hunt Group | Announcements 320.
- **ALLOW_5410_UPGRADES** *Software level - 4.1 Q1 2008 Maintenance Release+.*
  Previously the only control over the upgrading of 5410 phones was controlled by the use of the turn_on.bat and turn_off.bat batch files installed with the Manager application. Now in addition this option must be present for 5410 phones to update their firmware. Refer to the Manager Installation manual for full details.
- **DISTINCT_HOLD_RINGBACK** *(Software level = 4.1+)*
  Used to display a specific message about the call type for calls returning after timing out from being parked or held. If set, such calls display *Return Call - Held* or *Return Call – Parked* rather than connected party name or line name.

- ExtendDirectLimit <*optional limit*> *(Software level = 4.1 – 4.2)*
This command allows the number of directory entries that can be added to the configuration to be controlled. By default Manager imposes a limit of 1000 directory entries. If this command is used without specifying an optional limit, the limit set is determined by the type of control unit; refer to the table below. Note that specifying or allowing a large directory limit may affect the performance of applications that use and display the directory, for example Phone Manager.

  - ! Warning
  Specifying a large directory limit may affect the performance of applications that use and display the directory, for example Phone Manager, SoftConsole and phones with access to the directory. It may also delay the inward routing of calls as the system attempts to match ICLID's received to names using the directory entries.

- ExtendLDAPDirectLimit <*optional limit*> *(Software level = 4.1 – 4.2)*
This command allows the number of LDAP directory entries that the system will read to be controlled. By default the system will only support up to 500 LDAP directory entries. If this command is used without specifying an optional limit, the limit set is determined by the type of control unit; refer to the table below. Note that specifying or allowing a large directory limit may affect the performance of applications that use and display the directory, for example Phone Manager.

Control Unit	Directory Entries	LDAP Entries
Small Office Edition	100	1000
IP406 V2	2500	10000
IP412	10000	10000
IP500	10000	10000

- FORCE_HANDSFREE_TRANSFER *(Software level = 4.2 Q4 Maintenance release+)*
If set, when using the handsfree announced transfer 784 process, both the transfer enquiry and transfer completion calls are auto-answered. Without this setting only the transfer enquiry call is auto-answered.

- H323SetupTimerNoLCR *(Software level = 3.2 only)*
Used to set the fallback time from VoIP trunks to non-VoIP trunks within LCR. See IP Trunk Fallback. For IP Office 4.2+ the setting Call Initiation Timeout is available on SIP and IP trunks.

- HIDE_CALL_STATE
Used to hide the call status information, for example Dial, Conn, etc, on DS phones. Used in conjunction with the LONGER_NAMES option. Not supported for 1600 and 9600 Series phones.

- LONGER_NAMES
Used to increase the length of names sent for display on DS phones. See Caller Display 755. Not supported for 1600 and 9600 Series phones.

- ProgressEndsOverlapSend
See Line | VoIP 223.
- VM_TRUNCATE_TIME=*X (Range X = 0 to 7 seconds. Software level = 3.2 Maintenance Releases and 4.0+)*
On analog trunks, call disconnection can occur though busy tone detection. When such calls go to voicemail to be recorded or leave a message, when the call ends the system indicates to the voicemail system how much to remove from the end of the recording in order to remove the busy tone segment. This amount varies by system locale 844, the defaults being listed below. For some systems it may be necessary to override the default if analog call recordings are being clipped or include busy tone. That can be done by adding a VM_TRUNCATE_TIME= setting with the required value in the range 0 to 7 seconds.

  - New Zealand, Australia, China, Saudi Arabia and Custom: 5 seconds.
  - Korea: 3 seconds.
  - Italy, Mexico, Chile, Colombia and Brazil: 2 seconds.
  - Argentina, United States, Canada and Turkey: 0 seconds.
  - All other locales: 7 seconds.

- VMAIL_WAIT_DURATION=X *(Software level = 4.2 2Q09+)*
The number of milliseconds to wait before cutting through the audio to Voicemail. Some delay is required to allow for codec negotiation.

5.7.6 Telephony

This form allows you to set telephony related features for the user. These override any matching setting in the System | Telephony 160 tab. The settings are grouped into a number of sub-tabs.

5.7.6.1 Call Settings

For details of the ringing tones, see Ring Tones 762. DefaultRing uses the system default setting set through the System | Telephony 160 tab.

User | Telephony | Call Settings	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A 🔒 symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Outside Call Sequence: *Default = Default Ring (Use system setting 160)*
  Applies only to analog phones. Sets the ring pattern used for external calls to the user. The distinctive ring patterns used for other phones are fixed. Note that changing the pattern for users associated with fax and modem device extensions may cause those devices to not recognize and answer calls.
- Inside Call Sequence: *Default = Default Ring (Use system setting 160)*
  Applies only to analog phones. Sets the ring pattern used for internal calls to the user. The distinctive ring patterns used for other phones are fixed.
- Ring Back Sequence: *Default = Default Ring (Use system setting 160)*
  Applies only to analog phones. Sets the ring pattern used for ringback calls to the user. The distinctive ring patterns used for other phones are fixed.
- No Answer Time: *Default = Blank (Use system setting 160), Range = 6 to 99999 seconds.* 🔒
  Sets how long a call rings the user before following forwarded on no answer if set or going to voicemail. Leave blank to use the system default setting.
- Wrap-up Time (secs): *Default = 2 seconds, Range 0 to 99999 seconds.* 🔒
  Specifies the amount of time after ending one call before another call can ring. During this interval the user is treated as still being on a call. You may wish to increase this in a "call center" environment where users may need time to log call details before taking the next call. It is recommended that this option is not set to less than the default of 2 seconds. 0 is used to allow immediate ringing.
  - For users set as an CCR Agent, the After Call Work Time 276 (User | Telephony | Supervisor Settings 276) setting is also used.
- Transfer Return Time (secs): *Default = Blank (Off), Range 1 to 99999 seconds.* 🔒
  Sets the delay after which any call transferred by the user, which remains unanswered, should return to the user. A return call will continue ringing and does not follow any forwards or go to voicemail.
  - Pre-3.2 IP Office: The transfer return only occurs if the user has no other connected call.
  - IP Office 3.2+: Transfer return will occur if the user has an available call appearance button.
  - Transfer return is not applied if the transfer is to a hunt group that has queuing enabled.
- Call Cost Mark-Up: *Default = 100. Software level = 4.0+.*
  This setting is used for ISDN advice of charge (AOC). The markup is applied to the cost calculations based on the number of units and the line base cost per charging unit. The field is in units of 1/100th, for example an entry of 100 is a markup factor of 1. This value is included in the system SMDR output.
- Call Waiting On: *Default = Off* 🔒
  For users on phones without appearance buttons, if the user is on a call and a second call arrives for them, an audio tone can be given in the speech path to indicate a waiting call (the call waiting tone varies according to locale). The waiting caller hears ringing rather than receiving busy. There can only be one waiting call, any further calls receive normal busy treatment. If the call waiting is not answered within the no answer time, it follows forward on no answer or goes to voicemail as appropriate. User call waiting is not used for users on phones with multiple call appearance buttons. Call waiting can also be applied to hunt group calls, see Hunt Group | Hunt Group | Call Waiting 306. Call waiting should not be used for fax and modem devices.
- Answer Call Waiting on Hold (Analog): *Default = On*
  Applies to analog and IP DECT extension users only. If the user has a call waiting and places their current call on hold, the waiting call is automatically connected.

- Busy on Held: *Default = On*
  If on, when the user has a call on hold, new calls receive busy treatment. They will follow the users forward on busy setting or are diverted to voicemail. Otherwise busy tone (ringing for incoming analog calls) is played. This overrides call waiting when the user has a call on hold. The use of Busy on Held for users with multiple call appearance buttons is deprecated and Manager will prompt whether it should switch off busy on held for such a user.

- Offhook Station: *Default = Off*
  Off-hook station allows an analog extension to be left permanently off-hook, with calls being made and answered using an application or TAPI. When enabled, the analog extension user is able to control calls using the application in the following ways:

  - Offhook station does not disable the physical off-hook on the phone. When starting with the phone on-hook, making and answering calls is the same as normal analog extension operation. Additionally however calls can be initiated from the application. After entering the required number and making the call, the on-hook analog extension receives a ringback showing the users own caller ID and when answered the outgoing call leg to the dialed number is started.

    - Pre-IP Office 4.0: Calls to a busy destination are cleared immediately without hearing busy tone.

    - IP Office 4.0+: Calls to a busy destination present busy tone before being cleared. Except Phone Manager PC Softphone.

  - The application can be used to end a call with the analog extension still off-hook. Instead of hearing disconnect tone the user hears silence and can use the application to make another call. Though off-hook the user is indicated as idle on BLF indicators. Without off-hook Station set the user would be indicated as busy when off-hook, whether on a call or not.

  - If off-hook and idle (having cleared a previous call), incoming call alerts by presenting ringing through the audio path. The call can be answered using the application or going on-hook/off-hook or by pressing recall. Note that if the phone normally displays call ID, any caller ID displayed on the phone is not updated in this mode, however the call ID in the application will be that of the current call.

  - If on-hook, an incoming call alerts as normal using the phone's ringer and is answered by going off-hook. The answer call option in the application cannot be used to answer calls to an on-hook analog extension.

  - While off-hook and idle, the analog extension user will receive page calls.

  - If the analog extension handset is replaced with a headset, changing the Extension Classification 254 *(Extn | Analog)* 254 to *Quiet Handset* is recommended.

  - Offhook Station is not intended for non-analog phone extension. However, since it enables the answer calls control in Phone Manager it is recommended that the option is selected for non-analog extension users.

- System Phone: *Default = Off*
  Users set as a system phone user are able to access additional functions 765. For IP Office 6.0 and higher systems, the setting has been replaced by the System Phone Rights 264 setting on the User | User 264 tab.

- Remote Homeworker/Agent: *Default = Off. Software level = Up to 3.2 only.*
  Select if the user has been configured as a remote extension on an Avaya INDeX telephone system. Refer to the INDeX Level 10 documentation for full details. Only available in Locales where the Avaya INDeX switch is supported.

5.7.6.2 Supervisor Settings

User | Telephony | Supervisor Settings	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Login Code: *Default = Blank, Range = Up to 31 digits.*
  The code that has to be entered, as part of a log in sequence, to allow a user to make use of an extension as if it was their own phone. This log in code can be used for hot desking as well as logging back onto your phone after it has been used by a hot desking user. This entry must be at least 4 digits for DS port users. Login codes of up to 15 digits are supported with Extn Login 619 buttons 619. Login codes of up to 31 digits are supported with Extn Login 468 short codes 468.
  - For IP phone users, the login code should be limited to 13 digits. The user's login code is used by IP phones during registration with the system.
  - IP Office 4.0+: Users can hot desking between systems. To log in at a pre-IP Office 5 system requires that Manager to have a Advanced Small Community Networking license.
  - IP Office 4.0+: Users can only log out if they have a Login Code set.
  - IP Office 4.1 2008 Q2 maintenance release+: Supports the short code feature Change Login Code 451.
  - IP Office 4.2+: Users can log out without having a Login Code set if they are currently logged in at an extension whose Base Extension Number 252 (Extension | Extn) 252 no longer matches their own Extension 264 (User | User) 264.
  - If the user has a login code set, it is used by the Outgoing Call Bar Off 483 short code feature.
  - If the user has a login code set, access to a range of programmable button features will require entry of the login code. For example access Self Admin 651 and System Phone features.
- Login Idle Period (secs): *Default = Blank (Off), Range = 0 (Off) to 99999.*
  If the telephone is not used for this period; the user currently logged in is automatically logged out. This option should be used only in conjunction with Force Login (see below).
- Monitor Group: *Default = <None>*
  Sets the hunt group whose members the user can monitor if silent monitoring is setup. See Call Listen 443.
- Coverage Group: *Default = <None>. Software level = 5.0+*
  If a group is selected, then in scenarios where an external call would normally have gone to voicemail, it instead continues ringing and also starts alerting the members of the coverage group. For further details refer to Coverage Groups 748.
- Status on No Answer: *Default = Logged On. Software level = 4.0+.*
  Hunt groups can change the status of call center agents (users with a log in code and set to forced log in) who do not answer a hunt group call presented to them before it is automatically presented to the next agent. Use of this is controlled by the Agent's Status on No Answer Applies To 306 setting of the hunt group. This option is not used for calls ringing the agent because the agent is in another group's overflow group.
  - *Logged On*
    If this option is selected, the user's status is not changed.
  - *Busy Wrap-Up*
    If this option is selected the user's membership status of the hunt group triggering the action is changed to disabled. The user can still make and receive calls and will still continue to receive calls from other hunt groups to which they belong.
  - *Busy Not Available*
    If this option is selected the user's status is changed to do not disturb. This is the equivalent of DND and will affect all calls to the user.
  - *Logged Off*
    If this option is selected the users status is changed to logged out. In that state they cannot make calls or receive calls. Hunt group calls go to the next available agent and personal calls treat the user as being busy.
- Reset Longest Idle Time: *Default = All Calls. Software level = 4.0+.*
  This setting is used in conjunction with hunt groups set to Longest Waiting (also known as Idle and Longest Waiting). It defines what type of calls reset the idle time of users who are members of these hunt groups. Options are *All Calls* and *External Incoming*.

- Force Login: *Default = Off*
  If checked, the user must log in using their Login Code to use any extension including an extension to which they are the default associated user (Base Extension). For example, if Force Login is ticked for user A and user B has logged onto A's phone, when B logs off user A is not automatically associated with their normal phone and instead must log back on. If Force Login was not ticked, A would be automatically logged back in.

  - For users set as CCR Agents, Forced Login is automatically enabled and cannot be switched off.

  - Note that users with a Login Code and set to Forced Login are treated as call center agents. These users consume CCC agents licenses and their status is reported within CBC and CCC applications.

- Force Account Code: *Default = Off*
  If checked, the user must enter a valid account code to make an external call.

- Force Authorization Code: *Default = Off. Software level = 3.2+.*
  If checked, the user must enter a valid authorization code to make an external call. That authorization code must be one associated with the user or the user rights to which the user belongs. See Authorization Codes 404.

- Outgoing Call Bar: *Default = Off*
  When enabled, this setting stops a user from making any external calls except those that use dial emergency features. On many Avaya display phones, this causes a B to be displayed. The following features can be used with outgoign call bar: Outgoing Call Bar On 483, Outgoing Call Bar Off 483 and Change Login Code 451.

- Inhibit Off-Switch Forward/Transfers: *Default = Off. Software level = 3.2+.*
  When enabled, this setting stops the user from transferring or forwarding calls externally. This does not stop another user transferring the restricted users calls off-switch on their behalf. Note that a number of other controls may inhibit the transfer operation, see Off-Switch Transfer Restriction 782.

- Can Intrude: *Default = Off*
  Check this option if the User can interrupt other user's calls. This setting and the setting below are used to control the use of the following short code and button features: Call Intrude, Call Listen, Call Steal and Dial Inclusion.

- Cannot be Intruded: *Default = On (Pre-4.0 - Off in Italy)*
  If checked, this user's calls cannot be interrupted or acquired. In addition to the features listed above, this setting also affects whether other users can use their appearance buttons to bridge into a call to which this user has been the longest present user.

- Can Trace Calls: *Default = Off. Software level = 4.0+.*
  This settings controls whether the user is able to make used of ISDN MCID controls.

- Can Accept Collect Calls: *Default = Off [Brazil Only]*
  Determines whether the user is able to receive and accept collect calls.

- CCR Agent: *Default = Off. Software level = 4.2+.*
  This field is used by the CCR application to indicate which users are Agents monitored by that application. It also indicate to the system those users who can use other CCR features within the system configuration. If a user is set as an CCR Agent, Forced Login is enabled and greyed out from being changed and a warning is given if the user does not have a log in code set.

  - The number of simultaneous logged in CCR Agents supported by the system is controlled by licenses 839 entered into the configuration. If all agent licenses on a system have been used, additional agents are prevented from logging in.

- Automatic After Call Work: *Default = Off. Software level = 4.2.*
  CCR Agents (see above) can be automatically put into *After Call Work* (ACW) state after ending a hunt group call. During ACW state, further hunt group calls are not presented to the agent. Unless ended manually, the After Call Work state is automatically cleared after the agent's After Call Work Time setting. Automatic after call work is only supported when the agent is using a phone that supports an After Call Work 579 button.

- After Call Work Time (secs): *Default = System Default, Range = 0 (No ACW) to 999 seconds. Software level = 4.2 +.*
  For CCR Agents with Automatic After Call Work enabled, this value sets the duration of the ACW period. If set to *System Default*, the value set in System | CCR | Default After Call Work Time 184 is used. A value of *0* disables the user from using ACW.

5.7.6.3 Multi-line Options

User | Telephony | Multi-line Options	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Individual Coverage Time (secs): *Default = 10 seconds, Range 1 to 99999 seconds. Software level 3.0+.*
  This function sets how long the phone will ring at your extension before also alerting at any call coverage users. This time setting should not be equal to or greater than the No Answer Time applicable for the user.

- Ring Delay: *Default = Blank (Use system setting), Range = 0 (use system setting 160) to 98 seconds. Software level = 3.2+.*
  This setting is used when any of the user's programmed appearance buttons is set to Delayed ringing. Calls received on that button will initially only alert visually. Audible alerting will only occur after the ring delay has expired.

- Coverage Ring: *Default = Ring. Software level = 5.0+.*
  This field selects the type of ringing that should be used for calls alerting on any the user's call coverage and bridged appearance buttons. *Ring* selects normal ringing. *Abbreviated Ring* selects a single non-repeated ring. *No Ring* disables audible ringing. Note that each button's own ring settings (*Immediate*, *Delayed Ring* or *No Ring*) are still applied.

  - The ring used for a call alerting on a call coverage or bridged appearance button will vary according to whether the user is currently connected to a call or not.

    - If not currently on a call, the Coverage Ring setting is used.

    - If currently on a call, the quieter of the Coverage Ring and Attention Ring settings is used.

Attention Ring Setting	Coverage Ring Setting		
	Ring	Abbreviated	Off
Ring	Ring	Abbreviated	Off
Abbreviated	Abbreviated	Abbreviated	Off

- Attention Ring: *Default = Abbreviated Ring. Software level = 4.1+.*
  This field selects the type of ringing that should be used for calls alerting on appearance buttons when the user already has a connected call on one of their appearance buttons. *Ring* selects normal ringing. *Abbreviated Ring* selects a single ring. Note that each button's own ring settings (*Immediate*, *Delayed Ring* or *No Ring*) are still applied.

- Ringing Line Preference: *Default = On. Software level = 3.0+.*
  For users with multiple appearance buttons. When the user is free and has several calls alerting, ringing line preference assigns currently selected button status to the appearance button of the longest waiting call. Ringing line preference overrides idle line preference.

- Idle Line Preference: *Default = On. Software level = 3.0+.*
  For users with multiple appearance buttons. When the user is free and has no alerting calls, idle line preference assigns the currently selected button status to the first available appearance button.

- Delayed Ring Preference: *Default = Off. Software level = 4.0+.*
  This setting is used in conjunction with appearance buttons set to delayed or no ring. It sets whether ringing line preference should use or ignore the delayed ring settings applied to the user's appearance buttons.

  - When on, ringing line preference is only applied to alerting buttons on which the ring delay has expired.

  - When off, ringing line preference can be applied to an alerting button even if it has delayed ring applied. This is the same as pre-4.0 ringing line preference operation.

- Answer Pre-Select: *Default = Off. Software level = 4.0+.*
  Normally when a user has multiple alerting calls, only the details of the call on current selected button are shown. Pressing any of the alerting buttons will answer the call on that button, going off-hook will answer the current selected button. Enabling Answer Pre-Select allows the user to press any alerting button to make it the current selected button and displaying its call details without answering that call until the user either presses that button again or goes off-hook. Note that when both Answer Pre-Select and Ringing Line Preference are enabled, once current selected status is assigned to a button through ringing line preference it is not automatically moved to any other button.

- Reserve Last CA: *Default = Off. Software level = 4.0+.*
  Used for users with multiple call appearance buttons. When selected this option stops the user's last call appearance button from being used to receive incoming calls. This ensures that the user always has a call appearance button available to make an outgoing call and to initiate actions such as transfers and conferences. For pre-4.0 IP Office, this option is set by adding the RESERVE_LAST_CA= option on the User | Source Numbers 272 tab.

- Abbreviated Ring:
  This option has been replaced by the Attention Ring setting above.

5.7.6.4 Call Log

IP Office 5.0+: The system can store a centralized call log 745 for users. Each users' centralized call log can contain up to 30 call records for user calls (10 on IP412 and IP406 V2 systems). When this limit is reached, new call records replace the oldest record.

On 1400, 1600 and 9600 Series phones with a Call Log or History button, that button can be used to display the user's centralized call log. The user can use the call log to make calls or to store as a personal speed dial. They can also edit the call log to remove records. The same call log is also if the user logs into one-X Portal for IP Office application.

The centralized call log moves with the user if they log on and off from different phones. This includes if they hot desk within a Small Community Network.

User | Telephony | Call Log	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✓.

A 🔒 symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Centralized Call Log: *Default = System Default (On)* 🔒
  This setting allows the use of centralized call logging to be enabled or disabled on a per user basis. The default is to match the system setting Default Centralized Call Log On 165 (System | Telephony | Call Log 165). The other options are *On* or *Off* for the individual user. If off is selected, the call log shown on the users phone is the local call log stored by the phone.

- Delete entries after (hours:minutes): *Default = 00:00 (Never). Software level = 6.1+.* 🔒
  If a time period is set, entries in the user's call log are automatically deleted after this period.

- Groups: *Default = System Default (On).* 🔒
  This section contains a list of hunt groups on the system. If the system setting Log Missed Huntgroup Calls 160 (System | Telephony | Call Log) 160 has been enabled, then missed calls for those groups selected are shown as part of the users call log. The missed calls are any missed calls for the hunt group, not just group calls presented to the user and not answered by them.

5.7.7 Forwarding

This form can be used to check and adjust a user's call forwarding and follow me settings.

Follow Me is intended for use when the user is present to answer calls but for some reason is working at another extension. For example; temporarily sitting at a colleague's desk or in another office or meeting room. As a user, you would use Follow Me instead of Hot-Desking if you don't have a log in code or you don't want to interrupt you colleague also receiving their own calls. Multiple users can use follow me to the same phone.

Forwarding is intended for use when, for some reason, the user is unable to answer a call. They may be busy on other calls, unavailable or simply don't answer. Calls may be forwarded to internal or, subject to the user's call barring controls, external numbers.

To bar a user from forwarding calls to an external number, the Inhibit Off-Switch Forward/Transfers 276 (User | Telephony | Supervisor Settings 276) option should be selected. To bar all users from forwarding calls to external numbers the Inhibit Off-Switch Forward/Transfers 160 (System | Telephony | Telephony 160) option should be selected.

Note that analog lines doe not provide call progress signalling. Therefore calls forwarded off-switch via an analog line are treated as answered and are not recalled.

- Calls Forwarded
  Once a call has been forwarded to an internal destination, it will ignore any further Forward No Answer or Forward on Busy settings but may follow additional Forward Unconditional settings.

User | Forwarding	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

A symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Follow Me Number: *Default = Blank. Range = Internal extension number.*
  Redirects the user's calls to the internal extension number entered. If the redirected call receives busy or is not answered, it follows the user's forwarding and or voicemail settings as if it had been presented to their normal extension. When a user has follow me in use, their normal extension will give alternate dialtone 846 when off hook. For further details see Follow Me 771.
  - Calls targeting longest waiting type hunt groups ignore Follow Me.
  - Calls triggered by actions at the user's original extension, for example voicemail ringback, ignore Follow Me.
  - Park, hold and transfer return calls will go to the extension at which the user initiated the park, hold or transfer action.
- Forward Unconditional: *Default = Off*
  This option, when checked and a Forward Number is also set, forwards all external calls immediately. Additional options allow this forwarding to also be applied to internal calls and to hunt group calls if required. Using Follow Me overrides Forward Unconditional. When a user has forward unconditional in use, their normal extension will give alternate dialtone 846 when off hook. If the destination is an internal user on the same system, they are able to transfer calls back to the user, overriding the Forward Unconditional.
  - To Voicemail: *Default = Off. Software level = 5.0+.*
    If selected and forward unconditional is enabled, calls are forwarded to the user's voicemail mailbox. The Forward Number and Forward Hunt Group Calls settings are not used. This option is not available if the system's Voicemail Type is set to *None*. 1400, 1600 and 9600 Series phone users can select this setting through the phone menu. Note that if the user disables forward unconditional, for example using Phone Manager or a short code, the To Voicemail setting is cleared.
  - Forward Number: *Default = Blank. Range = Internal or External number. Up to 32 characters.*
    This option sets the destination number to which calls are forwarded when Forward Unconditional is checked. The number can be an internal or external number. This option is also used for Forward on Busy and Forward on No Answer if no separate Forward Number is set for those features. If a user forwards a call to a hunt group of which they are a member, the group call is not presented to them but is presented to other members of the hunt group.
  - Forward Internal Calls: *Default = On. Software level = 3.2+.*
    This option, when checked, sets that internal calls should be also be forwarded immediately when forward unconditional is active.

* **Forward Hunt Group Calls:** *Default = Off*
  Hunt group calls (internal and external) are not normally presented to a user who has forward unconditional active. Instead they are presented to the next available member of the hunt group. This option, when checked, sets that hunt group calls (internal and external) are also forwarded when forward unconditional is active. The group's **Ring Type** must be ***Sequential*** or ***Rotary***, not ***Collective*** or ***Longest Waiting***. The call is forwarded for the period defined by the hunt group's **No Answer Time** after which it returns to the hunt group if unanswered. Note also that hunt group calls cannot be forwarded to another hunt group.

* **Forward On Busy:** *Default = Off*
  When checked and a forward number is set, external calls are forwarded when the user's extension is busy. The number used is either the **Forward Number** set for **Forward Unconditional** or if set, the separate **Forward Number** set under **Forward On Busy**. Having **Forward Unconditional** active overrides **Forward on Busy**.

  * If the user has **Busy on Held** selected, if forward on busy is active it is applied when the user is free to receive calls but already has a call on hold.

  * If the user's phone has multiple call appearance buttons, the system will not treat them as busy until all the call appearance buttons are in use unless the last appearance button has been reserved for outgoing calls only.

* **Forward On No Answer:** *Default = Off*
  When checked and a forward number is set, calls are forwarded when the user does not answer within their set **No Answer Time** 274 (User | Telephony | Call Settings 274). Having **Forward Unconditional** active overrides **Forward on No Answer**.

  * **Forward Number:** *Default = Blank. Range = Internal or External number. Up to 32 characters.*
    If set, this number is used as the destination for **Forward On Busy** and **Forward On No Answer** when on. If not set, the **Forward Number** set for **Forward Unconditional** is used. If a user forwards a call to a hunt group of which they are a member, the group call is not presented to them but is presented to other members of the hunt group.

  * **Forward Internal Calls:** *Default = On. Software level = 3.2+.*
    When checked, this option sets that internal calls should be also be forwarded when forward on no answer or forward on busy is active.

5.7.8 Dial In

Use this dialogue box to enable dial in access for a remote user. An Incoming Call Route and RAS service must also be configured.

User | Dial In	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Dial In On: *Default = Off*
  When enabled, dial in access into the system is available via this User account.
- Dial In Time Profile: *Default = <None>*
  Select the Time Profile 360 applicable to this User account. A Time Profile can be used to set time restrictions on dial in access via this User account. Dial In is allowed during the times set in the Time Profile form. If left blank, then there are no restrictions.
- Dial In Firewall Profile: *Default = <None>*
  Select the Firewall Profile 362 to restrict access to the system via this User account. If blank, there are no Dial In restrictions.

5.7.9 Voice Recording

This tab is used to activate the automatic recording of user's external calls. IP Office 6.1: The recording of internal calls as well is also supported.

Call recording requires Voicemail Pro to be installed and running. Call recording also requires available conference resources similar to a 3-way conference.

- IP Office 4.0+ introduces the following changes to recording:
  - Calls to and from IP devices, including those using Direct media, can be recorded.
  - Calls parked or held pause recording until the unparked or taken off hold.
  - Recording is stopped if:
    - User recording stops if the call is transferred to another user.
    - User account code recording stops if the call is transferred to another user.
    - Hunt group recording stops if the call is transferred to another user who is not a member of the hunt group.
    - Incoming call route recording continues for the duration of the call on the system.

User | Voice Recording	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Record Outbound: *Default = None*
  Select whether automatic recording of outgoing calls is enabled. The Auto Record Calls option sets whether just external calls or external and internal calls are included. Options for recording are:
  - None: Do not automatically record calls.
  - On: Record the call if possible. If not possible to record, allow the call to continue.
  - Mandatory: Record the call if possible. If not possible to record, block the call and return busy tone.
  - Percentages of calls: Record a selected percentages of the calls.
- Record Inbound: *Default = None*
  Select whether automatic recording of incoming calls is enabled. Options for recording are:
  - None: Do not automatically record calls.
  - On: Record the call if possible. If not possible to record, allow the call to continue.
  - Mandatory: Record the call if possible. If not possible to record, block the call and return busy tone.
  - Percentages of calls: Record a selected percentages of the calls.
- Record Time Profile: *Default = <None> (Any time)*
  Used to select a time profile 360 during which automatic call recording of incoming calls is applied. If no profile is selected, automatic recording of incoming calls is active at all times.
- Recording (Auto): *Default = Mailbox*
  Sets the destination for automatically triggered recordings.
  - Mailbox
    This option sets the destination for the recording to be a selected user or hunt group mailbox. The adjacent drop down list is used to select the mailbox.
  - Voice Recording Library: *Software level = 3.0+.*
    This options set the destination for the recording to be a VRL folder on the voicemail server. The ContactStore application polls that folder and collects waiting recordings which it then places in its own archive. Recording is still done by the Voicemail Pro.
- Auto Record Calls: *Default = External. Software level = 6.1*
  This setting allows selection of whether *External* or *External & Internal* calls are subject to automatic call recording.
- Recording (Manual): *Default = Mailbox*
  Sets the destination for manually triggered recordings.
  - Mailbox
    This option sets the destination for the recording to be a selected user or hunt group mailbox. The adjacent drop down list is used to select the mailbox.

- Voice Recording Library: *Software level = 3.0+.*
  This options set the destination for the recording to be a VRL folder on the voicemail server. The ContactStore application polls that folder and collects waiting recordings which it then places in its own archive. Recording is still done by the Voicemail Pro.

5.7.10 Coverage

Call coverage allows calls ringing at one extension (the 'Sender') to also be presented and answered at other defined extensions (the 'Covering Extensions').

User | Coverage	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✗, IP500v2 ✗.
Software Level	1.3 to 3.0DT only.
Mergeable	✓.

- Covering Extension
  The number of the extension that will be receiving the calls from the selected extension.
- Covering User
  This is the user's account name associated with the covering extension.

To add a covering extension

1. Right-click within the Coverage window and select Add.
2. Choose from the list of extension/users.
3. Click OK.

Senders
Senders are extensions that share their alerting calls with other extensions, referred to as their covering extensions. The only calls that are not shared are:

- Hunt Group calls that alert at the sender.
- Automatic Intercom calls.
- Calls that have been forwarded/diverted to the sender.
- Paging calls.
- Calls that are being covered for another station.
- Calls from one of their covering extensions.

Covering Extensions
When the sender's extension rings, the covering extensions also ring and show the call on a free call appearance button. The display indicates that the call is from the sender by showing the incoming call's name or number and the sender's name.

Covering Extensions can receive their own calls as well as calls for the Sender. A Covering Extension can receive a call when:

- Send All Calls/Do Not Disturb is not active.
- Forwarding/Divert is not active.
- They have an available Call Appearance button to accept the call.

Notes

To help covering extensions handle coverage calls efficiently it is suggested that the following buttons are programmed.

- **Program additional Call Appearance buttons**
  Covering extensions must have enough call appearance buttons for their own calls and for the extensions they are covering. By default each extension has three call appearance buttons. A suggested minimum extra is one less than the number of call appearance buttons on the sender's extension.
- **Program a Voicemail Collect button for the Sender**
  This will allow the covering extension to transfer a call directly to the sender's voicemail.
- **Program an Automatic Intercom button for the Sender**
  This allows the covering extension to place a voice announcement. If you do not wish to make voice announcement calls, use Dial Intercom instead.
- **Program a Send All Calls button**
- **Program a Drop Button**
  This helps in transferring calls.

Call Alerting Scenarios

Listed below are examples of how calls to the sender's extension are handled in specific scenarios.

- **Sender and Covering Extensions available**
  An incoming call alerts both the sender's and covering extension's on call appearance buttons. It alert the sender's extension for their set No Answer Time and then alerts the covering extension only until the call is answered or the caller hangs up.
- **Sender available/Covering Extension not available**
  An incoming call alerts the sender only. The call remains alerting until it is answered or the caller hangs up.
- **Sender not available/Covering Extension available**
  The call will alert the covering extension but not the sender. The call remains alerting until the call is answered or the call hangs up.
- If voicemail is available and enabled for the sender, then in all the above scenarios, following the sender's No Answer Time the call is redirected to the Sender's voicemail.
- **Sender and Covering Extension not available**
  The caller hears busy tone or is redirected to the sender's voicemail.

5.7.11 Button Programming

This tab is used to assign functions to the programmable keys provided on many Avaya telephones. For full details of button programming refer to the section Button Programming 506.

- T3 Phones
  T3 phone buttons have default functions. These are not shown in the configuration file but can be overridden by settings added to the configuration file. Buttons left blank or set to call appearance will use the phone's default function for that button.

User | Button Programming	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.1+.
Mergeable	✓.

A 🔒 symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Button No.
  The number of the DSS key against which the function is being set. To set a function against a button double-click it or select it and then click Edit.
- Label 514
  This is a text label for display on the phone. If no label is entered, the default label for the selected action is used.
- Action 569
  Defines the action taken by the menu item.
- Action Data
  This is a parameter used by the selected action. The options here will vary according to the selected button action.
- Display All
  The number of button displayed is based on the phone associated with the user when the configuration was loaded. This can be overridden by selecting Display All Buttons. This may be necessary for users who switch between different phones using hot desking or have an expansion unit attached to their phone.

5.7.12 Menu Programming

These menus control a range of options that are specific to different types of phones. The functions become accessible when the user logs in on the appropriate type of phone.

5.7.12.1 T3 Telephony

These settings are applied to the user when they are using a T3 phone.

User | Menu Programming | T3 Options	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.1+.
Mergeable	✓.

- Third Party Forwarding
  Avaya T3 phone users can be given menu options to change the forwarding settings of other users. In addition to the following controls, this functionality is protected by the forwarding user's log in code.
  - Allow Third Party Forwarding: *Default = Off*
    Sets whether this user can change the forwarding settings of other users.
  - Protect from Third Party Forwarding: *Default = Off*
    Sets whether this user's forwarding settings can be changed by other users.
- Advice of Charge
  - Display Charges: *Default = On. Software level = 4.0+.*
    This setting is used to control whether the user sees ISDN AOC information when using a T3 phone.
- Allow Self Administer: *Default = Off. Software level = 5.0+.*
  If selected, this option allows the user to self-administer button programming.

5.7.12.2 Huntgroup

Avaya T3, 1400, 1600 and 9600 Series phone users can control various settings for selected hunt groups. These settings are also used for one-X Portal for IP Office.

User | Menu Programming | Huntgroup	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.1+.
Mergeable	✓.

- Can Change Membership: *Default = Off*
  This list shows the hunt groups of which the user is a member. Up to 10 of these groups can be checked; those group and the users current membership status are then displayed on the phone. The user can change their membership status through the phone's menus.
  - T3 Series Phones Pre-Manager 4.2: For the selected group, their status can be viewed and changed from the phone by the user using *Menu -> Settings -> Display/Audio ->Group Membership*.
  - T3 Series Phones - Manager 4.2+: The selected hunt groups and the user's current membership status are displayed on the T3 phones status display. That display can be used to change the status.
- Can Change Service Status: *Default = Off*
  This list shows all the hunt groups on the system. Up to 10 of these groups can be checked.
  - T3 Series Phones: The user is then able to view and change the service status of the checked groups through their T3 phones menus (*Menu -> Group State*).
  - T3 Series Phones - Manager 4.2+: In addition to changing the status of the individual hunt groups displayed via Menu -> Group State, the menu also displays option to change the status of all the groups; All in service, All night service and All out service.
- Can Change Night Service Group: *Default = Off. Software level = 5.0+.*
  If selected, the user can change the fallback group used when the hunt group is in Night Service mode.
- Can Change Out of Service Group: *Default = Off. Software level = 5.0+.*
  If selected, the user can change the fallback group used when the hunt group is in Out of Service mode.

5.7.12.3 1400/1600

This menu applies to 1400, 1600 and 9600 Series phones.

User | Menu Programming | 1400/1600	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✓.

- Include Forwarding in Menu: *Default = On. Software level = 5.0+.*
  This setting controls whether the user is able to view and use options to control their forwarding within the phone's menus.

5.7.12.4 4400/6400

4412, 4424, 4612, 4624, 6408, 6416 and 6424 phones have a Menu key, sometimes marked with an icon. When Menu is pressed, a number of default functions are displayed. The < and > keys can be used to scroll through the functions while the keys below the display can be used to select the required function.

The default functions can be overwritten by selections made within this tab.

User | Menu Programming	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- Menu No.
  The menu position which the function is being set.
- Label 514
  This is a text label for display on the phone. The label is limited to 5 characters. If no label is entered, the default label for the selected action is used.
- Action 569
  Defines the action taken by the menu button.
- Action Data
  This is a parameter used by the selected action. The options here will vary according to the selected button action.

5.7.13 Phone Manager Options

This tab is used to configure the user's Phone Manager application options.

User | Phone Manager Options	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

A symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Allow user to modify Phone Manager settings: *Default = On*
  This setting is used with the Phone Manager Status Options, Screen Pop Options and Hide Options below. It controls whether those options are applied every time the user starts Phone Manager or only the first time the user starts Phone Manager.
  - If this setting is enabled, then the system configuration setting of those options are only applied the first time a user starts Phone Manager on a PC. Those settings become part of the user's Phone Manager profile on that PC. They can be changed by the user through Phone Manager. On subsequent Phone Manager starts the Manager settings are ignored.
  - If this setting is not enabled, the system configuration settings are applied every time the user starts Phone Manager and cannot be overridden by the user.
- Agent Mode: *Default = Off*
  This option controls the setting of the Agent Mode option on the Configure Preferences | Agent Mode tab within Phone Manager Pro. When enabled, the user has additional toolbar controls for Busy Wrap Up, Busy Not Available and Select Group. Note that the options on the Phone Manager Pro Agent Mode tab can be grayed out from user changes by the Agent Mode setting in Configuration Options below.
- Phone Manager Type: *Default = Lite*
  Determines the mode in which the user's copy of the Phone Manager application operates. Note that the number of users able to simultaneously use modes other than Lite is controlled by licenses entered into the system configuration. This setting cannot be changed by the user. For pre-3.2 systems this setting was located on the User | User 264 tab.
  - *Lite*
    Basic Phone Manager mode. This mode does not require any licenses.
  - *Pro*
    Advanced Phone Manager mode that enables a range of additional functions. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.
  - *Phone Manager PC Softphone*
    This is the VoIP IP phone mode of Phone Manager Pro. This mode requires both an available Phone Manager Pro license and a Phone Manager Pro IP Audio Enabled license. The user must be associated with an VoIP extension within the system configuration.
  - *Pro Telecommuter: Software level = 4.1+.*
    This version of Phone Manager Pro is supported with Phone Manager 4.1+. It allows the user to make and receive calls via an external phone specified at Phone Manager log in. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.
- VoIP Mode: *Default = On*
  This option only appears if the selected Phone Manager Type is set to *Phone Manager PC Softphone*. It sets the Enable VoIP control within the user's Phone Manager PC Softphone. For IP Office 4.2+, this setting is automatically enabled and greyed out.
- Book a Conference in Phone Manager: *Default = Off*
  When enabled, displays links in the user's Phone Manager application for access to the Conferencing Center application if installed. Note that to book a conference requires the user to have a Conferencing Center user ID and password. This feature also requires the Conferencing Center URL 143 (System | System) 143 to be set. This setting cannot be changed by the user. *For pre-3.2 systems this setting was located on the User | User 264 tab.
- Configuration Options
  These options allow the user to changes the settings on the indicated configure preferences tabs within Phone Manager.
  - The controllable tabs for Phone Manager Lite are Telephone and Do Not Disturb. For IP Office 4.0+, the Mobile Twinning option on the Forwarding tab is also controllable (the twinning number remains editable even if the Mobile Twinning option has been restricted).

- The additional controllable tabs for Phone Manager Pro and Phone Manager PC Softphone are Screen Pop, Compact Mode, Agent Mode, Voicemail (Voicemail and Voicemail Ringback controls only).

- Screen Pop Options
These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone screen pop options *Ringing*, *Answering*, *Internal*, *External* and *Outlook*.

  - The Allow user to modify Phone Manager setting controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.

- Phone Manager Status Options
These options allow selection of the tabs to show within the call history area of the user's Phone Manager.

  - The tabs selectable for Phone Manager are All, Missed, Status and Messages.

  - The additional tabs selectable for Phone Manager Pro and PC Softphone are Incoming, Outgoing and Account Codes.

  - The Allow user to modify Phone Manager setting controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.

- Hide Options
These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone options Hide on close and Hide on no calls.

  - The Allow user to modify Phone Manager settings option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.

5.7.14 Hunt Group Memberships

This tab displays the hunt group of which the user has been made a member. The tick boxes indicate whether the user's membership of each of those groups is currently enabled or disabled.

5.7.15 Mobility (Twinning)

These settings relate to twinning features. These are where a user has a main or primary extension but also regularly answer calls at a secondary or twinned phone. These features are intended for a single user, they are not aimed at two users answering calls presented to a single primary extension.

Twinning

Twinning allows a user's calls to be presented to both their current extension and to another number. The system supports two modes of twinning:

	Internal	Mobile
Twinning Destination	Internal extensions only	External numbers only.
Supported in	All locales.*	All locales.
License Required	No	Yes

*IP Office 4.0 Q2 2007 maintenance release and higher only. Prior to IP Office 4.0 Q2 2007 maintenance release, Internal twinning was not supported in North American locales.

5.7.15.1 Internal Twinning

IP Office 3.1+: Internal twinning can be used to link two system extensions to act as a single extension. Typically this would be used to link a users desk phone with some form of wireless extension such as a DECT or WiFi handset.

Internal twinning is an exclusive arrangement, only one phone may be twinned with another. When twinned, one acts as the primary phone and the other as the secondary phone. With internal twinning in operation, calls to the user's primary phone are also presented to their twinned secondary phone. Other users cannot dial the secondary phone directly.

- If the primary or secondary phones have call appearance buttons, they are used for call alerting. If otherwise, call waiting tone is used, regardless of the users call waiting settings. In either case, the Maximum Number of Twinned Calls setting applies.
- Calls to and from the secondary phone are presented with the name and number settings of the primary.
- The twinning user can transfer calls between the primary and secondary phones.
- Logging out or setting do not disturb at the primary stops twinned calls alerting at the secondary also.
- Logging out or setting do not disturb at the secondary only affects the secondary.
- User buttons set to monitor the status of the primary also reflect the status of the secondary.
- Depending on the secondary phone type, calls alerting at the secondary but then answered at the primary may still be logged in the secondary's call log. This occurs if the call log is a function of the phone rather than the system.
- Call alerting at the secondary phone ignoring any Ring Delay settings applied to the appearance button being used at the primary phone. The only exception is buttons set to No Ring, in which case calls are not twinned.
- If the internal secondary extension is a Phone Manager Pro PC Softphone, it will only support basic call functions. Advanced functions such as mailbox access to the primary mailbox are not supported.

For IP Office 5.0+, the following enhancements apply to internal twinned extensions:

- If using a T3, 1400, 1600 or 9600 Series phone as the secondary extension:
  - The secondary extension's directory/contacts functions access the primary user's Centralized Personal Directory 744 entries in addition to the Centralized System Directory 741.
  - The secondary extension's call Log/call List functions access the primary user's Centralized Call Log 745.
  - The secondary extension's redial function uses the primary users Centralized Call Log 745. Note: The list mode or single number mode setting is local to the phone.
  - On 1400, 1600 and 9600 Series phones, twinned status is indicated by a T in the display of the secondary extension.
- For all phone types, changing the following settings from either the primary or secondary extension, will apply the setting to the primary user. This applies whether using a short code, programmable button or phone menu. The status of the function will be indicated on both extensions if supported by the extension type.
  - Forwarding settings.
  - Group membership status and group service status.
  - Voicemail on/off.
  - Do Not Disturb on/off and DND Exceptions Add/Delete.

Internal Twinning Settings

User | Mobility | Internal Twinning	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.1+ (For North American locales, only supported from the IP Office 4.0 Q2 2007 maintenance release).
Mergeable	✓.

- **Internal Twinning:** *Software level = 3.1+.*
  Select this option to enable internal twinning for a user. **Internal Twinning** cannot be selected for a user if they already have **Mobility Features** selected.
  - **Twinned Handset:** *Default = Blank.*
    For internal twinning, the drop-down list can be used to select an available user as the twinned calls destination. Users not displayed in the list are already twinned with another user. If the list is grayed out, the user is a twinning destination and the primary to which they are twinned is displayed. The secondary phone must be on the same system.
  - **Maximum Number of Twinned Calls:** *Default = 1.*
    If set to one, when either the primary or secondary phone are in use, any additional incoming call receives busy treatment. If set to two, when either phone is in use, it receives call waiting indication for any second call. Any further calls above two receive busy treatment.
  - **Twin Bridge Appearances:** *Default = Off. Software level = 4.1+.*
    By default only calls alerting on the primary phone's call appearance buttons also alert at the secondary. When this option is enabled, calls alerting on a bridged appearance button at the primary can also alert at the secondary.
  - **Twin Coverage Appearances:** *Default = Off. Software level = 4.1+.*
    By default only calls alerting on the primary phone's call appearance buttons also alert at the secondary. When this option is enabled, calls alerting on a coverage appearance button at the primary can also alert at the secondary.
  - **Twin Line Appearances:** *Default = Off. Software level = 4.1+.*
    By default only calls alerting on the primary phone's call appearance buttons also alert at the secondary. When this option is enabled, calls alerting on a line appearance button at the primary can also alert at the secondary.

5.7.15.2 Mobile Twinning

This method of twinning can be used with external numbers. Calls routed to the secondary remain under control of the system and can be pulled back to the primary if required. If either leg of an alerting twinned call is answered, the other leg is ended.

- IP Office 4.2+: The Mobile Twinning license is now called the Mobility Features license, reflecting the fact that it can be used for Mobile Call Control and one-X Mobile Client support in addition to just mobile twinning. The mode of license operation has also changed. Prior to IP Office 4.2, the license was only consumed by users who had Mobile Twinning enabled. For IP Office 4.2, it is consumed by a user when they are configured for any of the mobility features, including mobile twinning even if they have turned mobile twinning off.
- IP Office 6.0: Mobility options can be enabled for users whose Profile 265 is set to *Mobile Worker* or *Power User*.
- For systems upgraded to IP Office 6.0 with existing UMS Web Service licenses, those licenses can be used with users whose Profile is set to *Basic User*.

A number of controls are available in addition to those on this tab.

- Button Programming Actions
  The Emulation | Twinning action can be used to control use of mobile twinning. Set on the primary extension, when that extension is idle the button can be used to set the twinning destination and to switch twinning usage on/off. When a twinned call has been answered at the twinned destination, the button can be used to retrieve the call at the primary extension.
- Mobile Twinning Handover (IP Office Release 6.1)
  When on a call on the primary extension, pressing the Twinning button will make an unassisted transfer to the twinning destination. This feature can be used even if the user's Mobile Twinning setting was not enabled.
  - During the transfer process the button will wink.
  - Pressing the twinning button again will halt the transfer attempt and reconnect the call at the primary extension.
  - The transfer may return if it cannot connect to the twinning destination or is unanswered within the user's configured Transfer Return Time (if the user has no Transfer Return Time configured, a enforced time of 15 seconds is used).
- Short Code Features
  The following short code actions are available for use with mobile twinning.
  - Set Mobile Twinning Number.
  - Set Mobile Twinning On.
  - Set Mobile Twinning Off.
  - Mobile Twinned Call Pickup.
- Caller ID
  The options on the System | Twinning tab can be used to control which caller ID is sent with calls sent to the twinned destination. The use of those options may be restricted by the trunk type carrying the twinned call and the services provided by the line provider.

Mobile twinning is only applied to normal calls. It is not applied to:

- Intercom, dial direct and page calls.
- Calls alerting on line appearance, bridged appearance and call coverage buttons.
- Returning held, returning parked, returning transferred and automatic callback calls.
- Follow me calls.
- Forwarded calls (except in IP Office 4.2+ if the user's Forwarded Calls Eligible for Mobile Twinning setting is enabled).
- Hunt group calls (except in IP Office 4.2+ if the user's Hunt Group Calls Eligible for Mobile Twinning setting is enabled)
- Additional calls when the primary extension is active on a call or the twinning destination has a connected twinned call.

Do Not Disturb and Twinning

- Mobile Twinning
  Selecting DND disables mobile twinning.
- Internal Twinning
  - Logging out or setting do not disturb at the primary stops twinned calls alerting at the secondary also.

- Logging out or setting do not disturb at the secondary only affects the secondary.

- Do Not Disturb Exceptions List
  For both types of twinning, while DND is selected calls from numbers entered in the user's Do Not Disturb Exception List 270 are only presented to the primary phone.

User BLF indicators and application speed dials set to the primary user will indicate busy when they are connected to a twinned call including twinned calls answered at the mobile twinning destination.

Analog Lines
These types of lines do not provide call progress signalling. Once a twinned call has been sent to an analog line, the system assumes that it has been answered and stops ringing the primary extension.

Mobility Features Settings

User | Mobility | Mobility Features	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

A 🔒 symbol indicates that the setting can also be set and locked within a set of user rights 389 with which the user is associated using the Working Hour User Rights 264 and Out of Hours User Rights 264 settings. The user rights applied can be controlled by a time profile 360 selected as the user's Working Hours Time Profile 264 setting. The effect of the user rights can be displayed using the User Right View 264 control.

- Mobility Features:
  If enabled this option allows any of the mobility features to be enabled for the user. This is subject to license requirements of the system software release.
    - For systems being upgraded to IP Office 4.2 where users have been previously enabled for Mobile Twinning, this option will be off by default. This will disables the user's mobile twinning until their Mobility Features option is also enabled.
    - For IP Office 6.0, this option can be enabled for users whose Profile 265 is set to *Teleworker* or *Power User*.
        - For systems upgraded to IP Office 6.0 with existing UMS Web Service licenses, those licenses can be used with users whose Profile is set to *Basic User*.
  - Mobile Twinning: *Software level = 3.2+.*
    If selected, the user is enable for mobile twinning. The user can control this option through a Twinning 663 programmable button on their a phone.
        - For user's setup for one-X Mobile Client, changes to their Mobile Twinning status made through the system configuration or using a Twinning button are not reflected in the status of the Extension to Cellular icon on their mobile client. However, changes to the Extension to Cellular status made from the mobile client are reflected by the Mobile Twinning field in the system configuration. Therefore, for one-X Mobile Client users, it is recommended that they control their Mobile Twinning status through the one-X Mobile Client rather than through a Twinning button.
    - Twinned Mobile Number: *Default = Blank*.
      This field sets the external destination number for mobile twinned calls. It is subject to normal short code processing and should include any external dialing prefix if necessary. For users of one-X Mobile Client 734 and or Mobile Call Control 730 the number in this field is used to match the users setting to the incoming CLI.
    - Twinning Time Profile: *Default = <None> (Any time)*
      This field allows selection of a time profile during which mobile twinning will be used.
    - Mobile Dial Delay: *Default = 2 seconds* 🔒
      This setting controls how long calls should ring at the user's primary extension before also being routed to ring at the twinning destination number. This setting may be used at the user's choice, however it may also be a necessary control. For example, if the twinning number is a mobile device that has been switched off, the mobile service provider may immediately answer the call with their own voicemail service. This would create a scenario where the user's primary extension does not ring or ring only briefly.
    - Mobile Answer Guard: *Default = 0 (Off), Range = 0 to 99 seconds. Software level = 4.2+.*
      This control can be used in situations where calls sent to the twinned destination are automatically answered by a voicemail service or automatic message if the twinned device is not available. If a twinned call is answered before the Mobile Answer Guard expires, the system will drop the call to the twin.
    - Hunt group calls eligible for mobile twinning: *Default = Off* 🔒
      This setting controls whether hunt group calls ringing the user's primary extension should also be presented to the mobile twinning number.

- **Forwarded calls eligible for mobile twinning:** *Default = Off*
  This setting controls whether calls forwarded to the user's primary extension should also be presented to the mobile twinning number.

- **Twin When Logged Out:** *Default = Off. Software level = 4.2+.*
  If enabled, if the user logs off their primary extension, calls to that extension will still alert at their twinned device rather than going immediately to voicemail or busy.

  - When logged out but twinned, **Mobile Dial Delay** is not applied.

  - Hunt group calls (all types) will be twinned if **Hunt group calls eligible for mobile twinning** is enabled. When this is the case the user's idle time is reset for each externally twinned call answered (note that calls twinned over analog and analog emulation trunks are automatically treated as answered).

  - When the user's **Mobile Time Profile**, if configured, is not active they will not get twinning calls. Calls will be treated the same as the user was logged out user with no twinning.

  - Callback calls initiated by the user will mature to the **Twinned Mobile Number**. It will also be possible to initiate Automatic Callback to the user with external twinning and their busy/free state will be tracked for all calls via the system.

  - Any Bridged Appearance set to the user will not alert. Coverage appearance buttons for the user will continue to operate.

  - The BLF/user button status shown for a logged out user with **Logged Off Mobile Twinning** is as follows:

    - If there are any calls alerting or in progress through the system to the twin the user status is shown as alerting or in-use as appropriate. This includes the user showing as busy/in-use if they have such a call on hold and they have Busy on Held enabled.

    - If the user enables DND through Mobile Call Control or one-X Mobile client their status will show as DND/busy.

    - Calls from the system dialed direct to the users twinned destination rather than directed by twinning from their primary extension will not change the user's status.

- **one-X Mobile Client:** *Default = Off. Software level = 4.2+ (IP500/IP500v2 digital trunks only).*
  one-X Mobile Client is a software application that can be installed on Windows Mobile and Symbian mobile cell phones. It allows the user to access a number of system features. For details see one-X Mobile Client 734.

- **Mobile Call Control:** *Default = Off. Software level = 4.2+ (IP500/IP500v2 digital trunks only).*
  Mobile call control is only supported on digital trunks. It allows a user receiving a call on their twinned device to access system dial tone and then perform dialing action including making calls and activating short codes. For details see Mobile Call Control 730.

- **Mobile Callback :** *Default = Off. Software level = 6.0+ (IP500/IP500v2 digital trunks only).*
  Mobile callback 733 allows the user to call the system and then hang up. The system will then make a call to the user's CLI and when answered, provide them with dial tone from the system to make calls.

5.7.16 Announcements

Announcements are played to callers waiting to be answered. This includes callers being presented to hunt group members, ie. ringing, and callers queued for presentation.

- The system supports announcements using Voicemail Pro, Voicemail Lite (pre-5.0 only) or Embedded Voicemail.
- If no voicemail channel is available for an announcement, the announcement is not played.
- In conjunction with Voicemail Pro, the system allows a number of voicemail channels to be reserved for announcements. See System | Voicemail 157.
- With Voicemail Pro, the announcement can be replaced by the action specified in a Queued (1st announcement) or Still Queued (2nd announcement) start point call flow. Refer to the Voicemail Pro Installation and Maintenance documentation for details.
- Calls can be answered during the announcement. If it is a mandatory requirement that announcements should be heard before a call is answered, then a Voicemail Pro call flow should be used before the call is presented.
    - **Warning: Call Billing and Logging**
      Note that a call becomes connected when the first announcement is played to it. That connected state is signaled to the call provider who may start billing at that point. The call will also be recorded as answered within the SMDR 898 output once the first announcement is played.
- If a call is rerouted, for example forwarded, the announcement plan of the original user is still applied until the call is answered. The exception is calls rerouted to a hunt group at which point the hunt group announcement settings are applied.
- For announcements to be used effectively, either the user's no answer time must be extended beyond the default 15 seconds or Voicemail On should be deselected.

Recording Announcements

- **Voicemail Pro**
  There is no mechanism within the telephony user interfaces (TUI) to record user announcements. To provide custom announcements, user queued and still queued start points must be configured with Voicemail Pro with the required prompts played by a generic action.
- **Embedded Voicemail**
  Embedded Voicemail does not include any default announcement or method for recording an announcement. The Record Message 486 short code feature is provided to allow the recording of announcements. The telephone number field of short codes using this feature requires the extension number followed by either ".1" for announcement 1 or ".2" for announcement 2. For example, for extension number 300, the short codes ***91N# | Record Message | N".1"*** and ****92N# | Record Message | N".2"*** could be used to allow recording of the announcements by dialing ****91300#*** and ****92300#***.

Hunt Group | Announcements	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	✓.

- **Announcements On**: *Default = Off.*
  This setting enables or disables announcements.

- **Wait before 1st announcement**: *Default = 10 seconds. Range = 0 to 9999 seconds.*
  This setting sets the time delay from the calls presentation, after which the first announcement should be played to the caller. If Synchronize Calls is selected, the actual wait may differ, see below.

- **Flag call as answered**: *Default = Off.*
  This setting is used by the CCC and CBC applications. By default they do not regarded a call as answered until it has been answered by a person or by a Voicemail Pro action with **Flag call as answered** selected. This setting allows calls to be marked as answered once the caller has heard the first announcement. This setting is not used by the Customer Call Reporter application.

  - **Warning: Call Billing and Logging**
    Note that a call becomes connected when the first announcement is played to it. That connected state is signaled to the call provider who may start billing at that point. The call will also be recorded as answered within the SMDR 898 output once the first announcement is played.

- **Post announcement tone**: *Default = Music on hold.*
  Following the first announcement, you can select whether the caller should hear *Music on Hold* 763, *Ringing* or *Silence* until answered or played another announcement.

- **2nd Announcement**: *Default = On.*
  If selected, a second announcement can be played to the caller if they have still not been answered.

- **Wait before 2nd announcement**: *Default = 20 seconds. Range = 0 to 9999 seconds.*
  This setting sets the wait between the 1st and the 2nd announcement. If Synchronize Calls is selected, the actual wait may differ, see below.

- **Repeat last announcement**: *Default = On.*
  If selected, the last announcement played to the caller is repeated until they are answered or hang-up.

- **Wait before repeat**: *Default = 20 seconds. Range = 0 to 9999 seconds.*
  If **Repeat last announcement** is selected, this setting sets is applied between each repeat of the last announcement. If Synchronize Calls is selected, this value is grayed out and set to match the **Wait before 2nd announcement** setting.

- **Synchronize calls**: *Default = Off*
  This option can be used to restrict how many voicemail channels are required to provide the announcements.

  - *Off*
    When Synchronize calls is off, announcement are played individually for each call. This requires a separate voicemail channel each time an announcement is played to each caller. While this ensures accurate following of the wait settings selected, it does not make efficient use of voicemail channels.

  - *On*
    When Synchronize calls is on, if a required announcement is already being played to another caller, further callers wait until the announcement been completed and can be restarted. In addition, when a caller has waited for the set wait period and the announcement is started, any other callers waiting for the same announcement hear it even if they have not waited for the wait period. Using this setting, the maximum number of voicemail channels ever needed is 1 or 2 depending on the number of selected announcements.

    - **Note: Interaction with Voicemail Pro Queued and Still Queued Start Points**
      If either custom Queued or Still Queued start point call flows are being used for the announcements, when Synchronize Calls is enabled those call flows will support the playing of prompts only. Voicemail Pro actions such as Speak ETA, Speak Position, Menu, Leave Mail, Transfer and Assisted Transfer, etc. are not supported.

5.7.17 Personal Directory

Each user is able to have up to 100 personal directory records, up to the overall system limit.

Manager	Total User Directory Records
IP500/IP500v2	10800
IP412	3600
IP406 V2	1900

These records are used as follows:

- Pre-IP Office 5
  - When using a T3 phone, the user is able to view and call their personal directory numbers.
  - Unlike system directory numbers, these entries are not used for name matching against the incoming ICLID of calls the user received.
- IP Office 5+
  - When using ETR, T3, 1400, 1600, or 9600 Series phones, the user is able to view and call their personal directory numbers.
  - When using a 1400, 1600, or 9600 Series phone, the user is also able to edit and add personal directory entries.
  - If the user hot desks to a T3, 1400, 1600, or 9600 Series phone on another system in a Small Community Network, they can still access their personal directory.

Directory entries are used for two types of function:

- Directory Dialing
  Directory numbers are displayed by user applications such as Phone Manager and SoftConsole. Directory numbers are viewable through the Dir 615 function on many Avaya phones (Contacts or History). They allow the user to select the number to dial by name. The directory will also contain the names and numbers of users and hunt groups on the system.
  - The Dir function groups directory entries shown to the phone user into the following categories. Depending on the phone, the user may be able to select the category currently displayed. In some scenarios, the categories displayed may be limited to those supported for the function being performed by the user:
    - External
      Directory entries from the system configuration. IP Office 5.0+: This includes HTTP and LDAP imported entries.
    - Groups
      Groups on the system. If the system is in a Small Community Network it will also include groups on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - Users or Index
      Users on the system. If the system is in a Small Community Network it will also include users on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - Personal
      Available for T3 phones, T3 IP phones, 1400, 1600 and 9600 Series phones. These are the user's personal directory entries stored within the system configuration.
- Name Matching
  Directory entries are also used to associate a name with the dialled number on outgoing calls or the received CLI on incoming calls. When name matching is being done, a match in the users personal directory overrides any match in the system directory. Note that some user applications also have their own user directory.
  - The IP Office Phone Manager and SoftConsole applications have their own user directories which are also used by the applications name matching. Matches in the application directory may lead to the application displaying a different name from that shown on the phone.
  - Name matching is not performed when a name is supplied with the incoming call, for example QSIG trunks.
  - Directory name matching is not supported for DECT handsets.

User | Personal Directory	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.

Software Level	3.1+.
Mergeable	✓.

- Name: *Range = Up to 31 characters.*
  Enter the text to be used to identify the number.

- Number: *Range = Up to 31 digits plus * and #.*
  Enter the number, without spaces, to be dialed. Wildcards are not supported in user personal directory entries. Note that if the system has been configured to use an external dialing prefix, that prefix should be added to directory numbers.

5.7.18 SIP

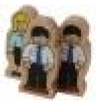
This tab is available when a SIP trunk with a SIP URI 238 entry has been added to the IP Office configuration. It is also available when an H323 trunk 220 set to *IP Office SCN* or *IP Office SCN - Fallback* has been added to the IP Office configuration.

Various fields within the URI settings used by SIP trunks can be set to Use Internal Data. When that is the case, the values from this tab are used inserted into the URI when the user makes or receives a SIP call. Within a Small Community Network, that includes calls which break out using a SIP trunk on another system within the SCN.

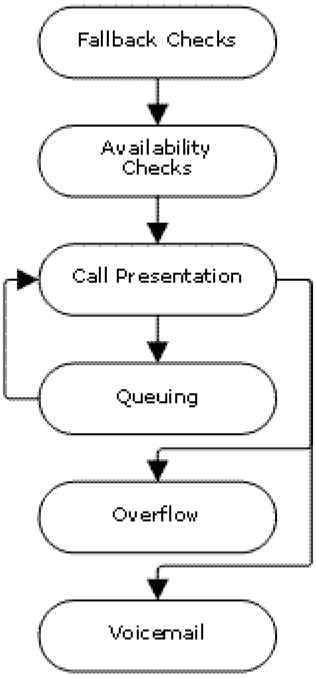
User | SIP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	✓.

- SIP Name: *Default = User name.*
  The value from this field is used when the From field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- SIP Display Name (Alias): *Default = User name.*
  The value from this field is used when the Display Name field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- Contact: *Default = User name.*
  The value from this field is used when the Contact field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- Anonymous: *Default = Off.*
  If the From field in the SIP URI is set to *Use Internal Data*, selecting this option inserts *Anonymous* into that field rather than the SIP Name set above.

5.8 Hunt Group Settings



A hunt group is a collection of users accessible through a single directory number. Calls to that hunt group can be answered by any available member of the group. The order in which calls are presented can be adjusted by selecting different group types and adjusting the order in which group members are listed.



- Call Presentation
  The order in which the available members of the hunt group are used for call presentation is selectable.
- Availability
  There are a range of factors which control whether hunt group calls are presented to a user in addition to that user being a member of the hunt group.
- Queuing
  This optional feature allows calls to be queued when the number of calls to be presented exceeds the number of available hunt group members to which call can be presented.
- Announcements
  On systems with a voicemail server (Voicemail Pro, Voicemail Lite (pre-IP Office Release 5) or Embedded Voicemail), announcements can be played to callers waiting to be answered. That includes calls that are ringing and calls that are queued.
- Overflow
  This optional feature can be used to redirect calls to an overflow group or groups when not answered within a set time.
- Fallback
  A hunt group can be taken out of operation manually or using a time profile. During fallback, calls can be redirected to a fallback group or sent to voicemail or just receive busy tone. Two types of fallback are supported; night service and out of service.
- Voicemail
  Calls can be redirected to voicemail. The system allows selection of whether hunt group calls remain in the hunt group mailbox or are copied (broadcast) to the individual mailboxes of the hunt group members. When messages are stored in the hunt group's own mailbox, selection of who receives message waiting indication is possible.

Changing the name of a hunt group has the following effects:

- A new empty mailbox is created on voicemail with the new hunt group name.
- Entries in other groups' Overflow lists will be updated.
- Out-of-Service and Night-Service fallback references are updated.

Modifying the extension number of a hunt group updates the following:

- Group buttons.
- Overflow, Out of Service Fallback and Night Service Fallback group entries.
- Incoming call route entries.

When a hunt group is deleted, all references to the deleted group will be removed including:

- Entry in Incoming call routing table.
- Transfer target in internal auto-attendant.
- Overflow, Night-Service or Fallback-Service on other groups.
- DSS keys monitoring group status.

Hunt Groups in a Small Community Network (SCN)

In a Small Community network, the extension numbers of users are automatically shared between IP Office systems and become diallable from other systems without any further programming.

For IP Office 4.0+, the following features are available for hunt groups within a Small Community Network. For pre-IP Office Release 5 systems, these features requires the IP Offices to have *Advanced Small Community Networking* licenses.

- **Advertised Hunt Groups**
  Each hunt group can be set as being 'advertised'. The hunt group can then be dialed from other systems within the SCN. The hunt groups extension number and name must be unique within the network. Non-advertised hunt group numbers remain local only to system hosting the hunt group.

- **Distributed Hunt Groups**
  Hunt groups on a system can include users located on other IP Office systems within the SCN network. Distributed hunt groups are automatically advertised to other systems within the SCN. Note that distributed hunt groups can only be edited on the system on which they were created.

5.8.1 Hunt Group

Hunt Group | Hunt Group	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Name: *Range = Up to 15 characters*
  The name to identify this hunt group. Only alphanumeric characters with no spaces should be used. Start names with a alphabetic character. This field is case sensitive and must be unique.
  - Voicemail uses the name to match a group and its mailbox. Changing a group's name will route its voicemail calls to a new mailbox. Note however that Voicemail Pro will treat names such as "Sales", "sales" and "SALES" as being the same.
- CCR Agent Group: *Default = Off. Software level = 4.2+.*
  This option is used in conjunction with IP Office CCR application to indicate the groups for which it collects information. CCR Agent Hunt Groups should only contain users who have been configured as CCR Agents 276 (User | Telephony | Supervisor Settings 276) option. When selected, the menus to select hunt group members will only show users configured as CCR Agents and a warning will be given if the group already contains any users who are not CCR Agents.
- Extension: *Range = 2 to 9 digits.*
  This sets the directory number for calls to the hunt group.
  - Groups for CBC and CCC should only use up to 4 digit extension numbers.
  - Extension numbers in the range 8897 to 9999 are reserved for use by the IP Office Delta Server.
- Ring Mode: *Default = Sequential*
  Sets how the system determines which hunt group member to ring first and the next hunt group member to ring if unanswered. This is used in conjunction with the User List which list the order of group membership.
  - *Collective* (previously known as *Group*)
    All available phones in the User List ring simultaneously.
  - *Collective Call Waiting (Software level = 3.0+)*
    This is a *Collective* hunt group as above but with hunt group call waiting also enabled (previous versions of Manager used a separate Call Waiting On control to select this option for a *Collective* group). When an additional call to the hunt group call is waiting to be answered, users in the group who are already on a call will receive call waiting indication. On phones with call appearance buttons, the call waiting indication takes the form of an alert on the next available call appearance button. On other phones, call waiting indication is given by a tone in the speech path (the tone is locale specific).
    - Pre-IP Office 4.0: All the users in the group must also have their own Call Waiting On 274 setting enabled.
    - IP Office 4.0+: The user's own Call Waiting On setting is overridden when they are using a phone with call appearances. Otherwise the user's Call Waiting On setting is used in conjunction with the hunt group setting.
  - *Sequential* (previously known as *Linear* and *Hunt*)
    Each extension is rung in order, one after the other, starting from the first extension in the list each time.
  - *Rotary* (previously known as *Circular*)
    Each extension is rung in order, one after the other. However, the last extension used is remembered. The next call received rings the next extension in the list.
  - *Longest Waiting* (previously known as *Idle* and *Most Idle*)
    The extension that has been unused for the longest period rings first, then the extension that has been idle second longest rings, etc. For extensions with equal idle time, 'sequential' mode is used.
    - IP Office 4.2+: Where hunt group calls are being presented to a twinned extension, the longest waiting status of the user can be reset by calls answered at either their master or twinned extension.
- No Answer Time (secs): *Default = System Default. Range = System Default or 6 to 99999 seconds.*
  The number of seconds an extension rings before the call is passed to another extension in the list. This applies to all telephones in this group and the Overflow Groups (if used). For collective hunt groups, the idea of moving to the next member when the No Answer Time expires does not apply, instead calls will continue ringing unless overflow or voicemail is applied.
- Hold Music Source: *Default = No Change. Software level =4.2+ (Not Small Office Edition).*
  The system can support up to 4 music on hold sources; the *System Source* (either an internal file or the external source port or tones) plus up to 3 additional internal wav files, see System | Telephony | Tones & Music 163. Before reaching a hunt group, the source used is set by the system wide setting or by the Incoming Call Route 344 that routed the call. If the system has several hold music sources available, this field allows selection of the source to associate with calls presented to this hunt group or to leave it unchanged. The new source selection will then apply even if the call is forwarded or transferred out of the hunt group unless changed again by another hunt group. If the call is routed to another IP Office 4.2+ system in a Small Community Network, the matching source is used if available.

- Calls overflowing from a hunt group will use the hold music source setting of the original hunt group and ignore the setting of the overflow group.
- Calls going to night service or out of service fallback group use the hold music source setting of the original hunt group and then, if different, the setting of the fallback group. The setting of further fallback groups from the first are ignored.

- **Voicemail Answer Time:** *Default = 45 seconds, Range = Off, 1 to 99999 seconds. Software level = 4.0+.* This setting sets how long a call should be presented to a hunt group, and its overflow groups if set, before going to voicemail. When exceeded the call goes to voicemail (if available) regardless of any announcements, overflow, queuing or any other actions. If set to Off, voicemail is used when all available members of the hunt group have been altered for the no answer time.
- **Agent's Status on No-Answer Applies To:** *Default = None (No status change). Software level = 4.0+.* For call center agents, that is hunt group members with a log in code and set to forced log in, the system can change the agent's status if they do not answer a hunt group call presented to them before being automatically presented to the next available agent.
  - This setting defines what type of hunt group calls should trigger use of the agent's **Status on No Answer** setting. The options are *None*, *Any Call* and *External Inbound Calls Only*.
  - The new status is set by the agent's Status on No Answer 276 (User | Telephony | Supervisor Settings 276) setting.
  - This action is only applied if the call is unanswered at the agent for the hunt group's **No Answer Time** or longer. It does not apply if the call is presented and, before the **No Answer Time** expires, is answered elsewhere or the caller disconnects.
  - This option is not used for calls ringing the agent because the agent is in another group's overflow group.
- **Central System:** *Software level = 4.0+.* The field is for information only. It displays the IP Office system where the hunt group was created and can be configured. For pre-IP Office Release 5 systems, this field is only visible if the IP Office has an *Advanced Small Community Networking* license.
- **Advertise Group:** *Default = Off. Software level = 4.0+.* If selected, details of the hunt group are advertised to the other systems within a Small Community Network and the hunt group can be dialled from those other systems without the need for routing short codes. For pre-IP Office Release 5 systems, this field is only visible if the IP Office has an *Advanced Small Community Networking* license.
  - Advertised groups must have an extension number that is unique within the SCN. If an advertised hunt group's extension number conflicts with a local groups extension number, the advertised group is ignored.
  - Groups set as advertised will appear in the configuration of other IP Office systems. However an advertised group can only be edited on the IP Office system on which it was created. Note that advertised groups are not saved as part of the configuration file when **File | Save Configuration As** is used.
  - Hunt groups that contain members from other IP Office systems are automatically advertised.
- **Call Waiting On:** *Default = Off. Software level = 3.0+.* For Manager 6.2 and above, this control has been combined with the **Ring Type** option *Collective*. See the **Ring Mode** setting *Collective Call Waiting* above.
- **User List** This is an ordered list of the users who are members of the hunt group. For *Sequential* and *Rotary* groups it also sets the order in which group members are used for call presentation.
  - Repeated numbers can be used, for example 201, 202, 201, 203, etc. Each extension will ring for the number of seconds defined by the **No Answer Time** before moving to the next extension in the list, dependent on the Hunt Type chosen.
  - The check box next to each member indicates the status of their membership. Checked boxes appear for members whose membership is enabled. The order of the users can be changed by dragging the existing entries to the required position.
  - To add entries select **Edit**. A new menu is displayed that shows available users on the left and current group members of the right. The lists can be sorted and filtered.
  - Users on remote systems in a Small Community Network can also be included. Groups containing remote members are automatically advertised within the SCN. For pre-IP Office Release 5 systems, this can only be done if the IP Office has an *Advanced Small Community Networking* license.
- **Overflow Group List** If a call cannot be answered by the extensions shown in the **User List**, it can be presented to available extensions in the groups listed in this list. Each group is used in turn in order from the top of the list. The call is presented to each overflow group member once, using the Ring Mode of the overflow group. If the call remains unanswered the next overflow group listed is used. If the call remains unanswered at the end of the list of overflow groups, it is presented to available members of the overflowing group again and then to those in its overflow list in a repeating loop.
  - If **Queuing** is off and all members of the hunt group are busy, a call presented to the group will overflow immediately, irrespective of the **Overflow Time**.
  - If **Queuing** is on and all members of the hunt group are busy, a call presented to the group may queue for up to the **Overflow Time** before overflowing.

- If the call is currently ringing a hunt group member when the **Overflow Time** expires, it will complete ringing using the group's **No Answer Time** before overflowing.
- If no **Overflow Time** is set, a call will overflow when it has rung each available hunt group member without being answered.
- The groups in the overflow list are only used to expand the set of users available to answer calls. The overflowing call still belongs to the group that is overflowing and uses the settings of that group. For example:
  - Calls that overflow use the announcement settings of the group from which they are overflowing.
  - Calls that overflow use the **Voicemail Answer Time** of the original group from which are are overflowing.
  - Calls that are overflowing are included in the overflowing group's **Queue Length** and **Calls In Queue Threshold**. They are not included in those values for the hunt group to which they overflow.
  - The only settings of the groups in the overflow list that are used is the **Ring Mode**.

- **Overflow Time**: *Default = Blank, Range = Off or 1 to 99999 seconds.*
  Hunt groups with an **Overflow Group List** set support the use of overflow to expand the group. When also using queueing 316 the **Overflow Time** can also be used to set how long calls queue before also overflowing.
- **Overflow Mode**: *Default = Group. Software level = 4.2+.*
  This option allows selection of whether the overflow of queued calls is determined on a call by call basis or applied to all calls for the hunt group once any one call overflows.
  - **Group**
    In this mode, once one call overflows all additional queued calls overflow immediately. This is equivalent to the overflow mode used by IP Office 4.0-4.1 systems.
  - **Call**
    In this mode, each individual call will follow the groups overflow time settings before it overflows. This is equivalent to the overflow mode used by pre-IP Office 4.0 system.

5.8.2 User List/Select Members

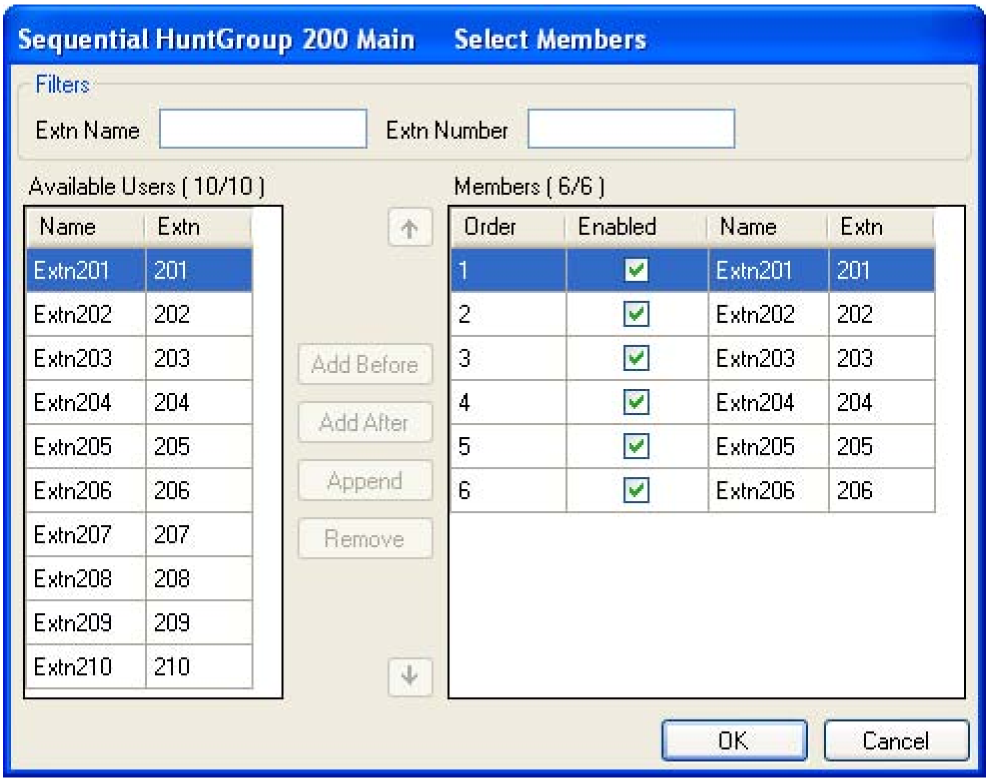
The hunt group Select Members form is used to add and remove users from the hunt group. For hunt group's with a Ring Mode of *Sequential* or *Rotary* it is also used to set the order of use for the members of the hunt group.

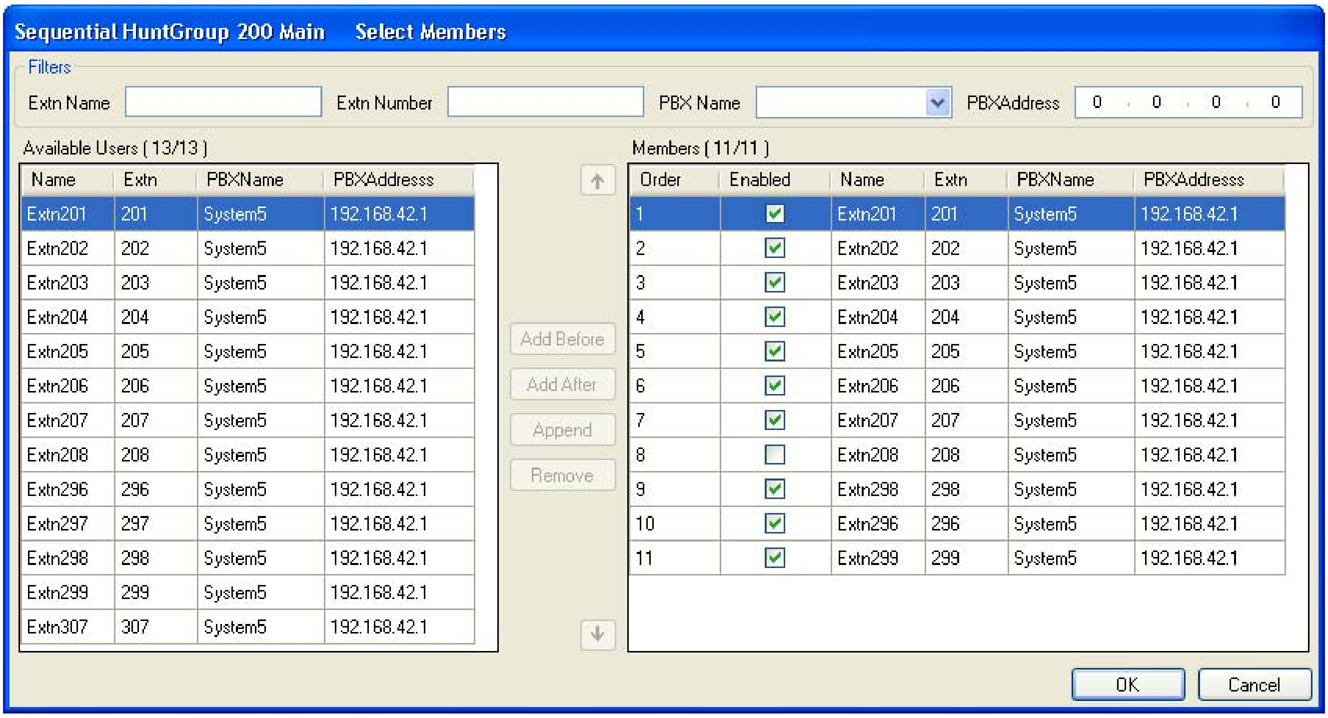
The filters section at the top of the form can be used to filter the users shown. Note for hunt groups set as a CCR Agent Group, only users set as CCR Agent are shown.

The controls and data on the form vary depending on the hunt group's Ring Mode setting and on whether the system is in a Small Community Network.

To sort either table, click on the column header that should be used for the sort the table. Sort changes the order of display only, it does not change the actual order of hunt group membership.

For *Sequential* and *Rotary* hunt groups, an Order column is shown. To change the order position of a hunt group member, select the member and then use the ↑ up and down ↓ arrow buttons.





During the actions below, the Shift and Ctrl keys can be used as normal to select multiple users. Note that that the list of members has been sorted, the sort is updated after adding or moving members.

- **Add Before**
  Using the Shift and/or Ctrl keys, select the users you want to add and then on the right select the existing member that you want to add them before.

- **Add After**
  Using the Shift and/or Ctrl keys, select the users you want to add and then on the left select the existing member after which you want them added.

- **Append**
  Add the selected users on the left to the hunt group members on the right as the last member in the group order.

- **Remove**
  Remove the selected users on the right from the list of hunt group members.

- **↑ ↓**
  Move the selected member on the right up or down the membership order of the group.

5.8.3 Voicemail

The system supports voicemail for hunt groups in addition to individual user voicemail mailboxes.

- **When is voicemail used?**
  If voicemail is available and enabled for a hunt group, it is used in the following scenarios.

  - **Voicemail Answer Time** *(IP Office 4.0+)*
    The default timeout is 30 seconds. When exceeded the call goes to voicemail (if available) regardless of any announcements, overflow, queuing or any other actions.

  - **Unanswered Calls**
    When a call has rung unanswered at all the available hunt group members. If overflow is being used that will include being unanswered by the available overflow group members.

  - **Queue Limit Reached**
    If queuing is being used, it overrides use of voicemail prior to expiry of the Voicemail Answer Time, unless the number of queued callers exceeds the set Queue Limit. By default there is no set limit.

  - **Night Service**
    When the hunt group is in night service with no Night Service Fallback Group set.

  - **Out of Service**
    When the hunt group is out of service with no Out of Service Fallback Group set.

  - **Automatic Call Recording**
    Incoming calls to a hunt group can be automatically recorded using the settings on the Hunt Group | Voice Recording 319 tab.

- **Which Mailbox is Used**

  - When a caller is directed to voicemail to leave a message, the system indicates the target user or hunt group mailbox.

    - Pre-IP Office 4.0: The mailbox of the user or hunt group whose settings have caused the call to go to voicemail is used.

    - For IP Office 4.0+: The mailbox of the originally targeted user or hunt group is used. This applies even if the call has been forwarded to another destination. It also includes scenarios where a hunt group call overflows or is in fallback to another group.

    - Voicemail Pro can be used to customize which mailbox is used separately from the mailbox indicated by the system.

- **Who Receives Message Waiting Indication?**
  By default no user is configured to receive message waiting indication when a hunt group voicemail mailbox contains new messages. Message waiting indication is configured by adding a H*groupname* entry to a user's SourceNumbers tab (User | Source Numbers 272).

- **Accessing Hunt Group Messages**
  By default no mechanism is provided for access to specific hunt group mailboxes. Access needs to be configured using either a short code, programmable button or source number.

  - **Phone Manager**
    User's with hunt group message waiting indication can access the hunt group mailbox through Phone Manager.

  - **Intuity Emulation Mailbox Mode**
    For systems using Intuity emulation mode mailboxes, the hunt group extension number and voicemail code can be used during normal mailbox access.

  - **IP Office Mailbox Mode**
    For IP Office mode mailbox access, short codes are required to access the mailbox directly.

- **Broadcast**
  The voicemail system (Voicemail Pro only) can be instructed to automatically forward messages to the individual mailboxes of the hunt group members. The messages are not stored in the hunt group mailbox.

Hunt Group | Voicemail	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- **Voicemail Code:** *Default = Blank, Range = 0 to 15 digits*
  A code (1-15 digits) used by the voicemail server to validate access to this mailbox. If remote access is attempted to a mailbox that has no voicemail code set, the prompt "Remote access is not configured on this mailbox" is played. Whether the caller will be prompted to enter this code varies as follows:
  - **Embedded Voicemail**
    The voicemail code is used if set.
  - **Voicemail Lite/Voicemail Pro in IP Office mode**
    The voicemail code is required when accessing the mailbox from a location that is not set as a trusted number in the user's Source Numbers 272 list. Also Voicemail Pro call flows containing an action where the action's PIN code set to $ will prompt the user for their voicemail code.
  - **Voicemail Pro in Intuity Emulation mode**
    By default the voicemail code is required for all mailbox access. The first time the mailbox is accessed the user will be prompted to change the password. Also if the voicemail code setting is left blank, the caller will be prompted to set a code when they next access the mailbox. The requirement to enter the voicemail code can be removed by adding a customized user or default collect call flow, refer to the Voicemail Pro manuals for full details. Also Voicemail Pro call flows containing an action where the action's PIN code set to $ will prompt the user for their voicemail code.
  - Codes set through the Voicemail Pro telephone user interface are restricted to valid sequences. For example, attempting to enter a code that matches the mailbox extension, repeat the same number (1111) or a sequence of numbers (1234) are not allowed. If these types of code are required they can be entered through Manager.
- **Voicemail On** *Default = On*
  When on, the mailbox is used by the system to answer the user's unanswered calls or calls when the user's extension returns busy. Note that selecting off does not disable use of the user's mailbox. Messages can still be forward to their mailbox and recordings can be placed in it. The mailbox can also still be accessed to collect messages.
  - When a caller is directed to voicemail to leave a message, the system indicates the target user or hunt group mailbox.
    - Pre-IP Office 4.0: The mailbox of the user or hunt group whose settings have caused the call to go to voicemail is used.
    - For IP Office 4.0+: The mailbox of the originally targeted user or hunt group is used. This applies even if the call has been forwarded to another destination. It also includes scenarios where a hunt group call overflows or is in fallback to another group.
    - Voicemail Pro can be used to customize which mailbox is used separately from the mailbox indicated by the system.
- **Voicemail Help** *Default = Off*
  For voicemail systems running IP Office mailbox mode, this option controls whether users retrieving messages are automatically given an additional prompt *"For help at any time press 8."* If switched off, users can still press 8 for help. For voicemail systems running in Intuity emulation mode, this option has no effect. On those systems the default access greeting always includes the prompt *"For help at any time, press *4"* (*H in the US locale).
- **Broadcast:** *Default = Off. Software level = 3.0+ (Voicemail Pro only).*
  If a voicemail message is left for the hunt group and Broadcast is enabled, copies of the message are forwarded to the mailboxes of the individual group members. The original message in the hunt group mailbox is deleted unless it occurred as the result of call recording.
- **UMS Web Services:** *Default = Off. Software level = 5.0+.*
  This option is used with Voicemail Pro. If enabled, the hunt group mailbox can be accessed using either an IMAP email client or a web browser. Note that the mailbox must have a voicemail code set in order to use either of the UMS interfaces. *UMS Web Service* licenses are required for the number of users and groups configured.
  - In the License 378 section, double-clicking on the **UMS Web Services** license display a menu that allows you to add and remove users and groups from the list of those enabled for UMS Web Services without having to open the settings of each individual user or group.
- **Voicemail Email:** *Default = Blank (No voicemail email features)*
  This field is used to set the user or group email address used by the voicemail server for voicemail email operation. When an address is entered, the additional Voicemail Email control below are selectable to configure the type of voicemail email service that should be provided.

- Use of voicemail email requires the Voicemail Pro server to have been configured to use either a local MAPI email client or an SMTP email server account. For embedded voicemail, voicemail email is supported with IP Office 4.2+ (except not on Small Office Edition) and uses the system's SMTP 177 settings.

- The use of voicemail email for the sending (automatic or manual) of email messages with wav files attached should be considered with care. A one-minute message creates a 1MB .wav file.

- Voicemail Email *Default = Off*
If an email address is entered for the user or group, the following options become selectable. These control the mode of automatic voicemail email operation provided by the voicemail server whenever the voicemail mailbox receives a new voicemail message.

    - Users can change their voicemail email mode using visual voice. If the voicemail server is set to IP Office mode, user can also change their voicemail email mode through the telephone prompts. The ability to change the voicemail email mode can also be provided in a call flow using a Play Configuration Menu action or a Generic action.

    - If the voicemail server is set to IP Office mode, users can manually forward a message to email.

  - *Off*
  If off, none of the options below are used for automatic voicemail email. Users can also select this mode by dialing *03 from their extension.

  - *Copy*
  If this mode is selected, each time a new voicemail message is received in the voicemail mailbox, a copy of the message is attached to an email and sent to the email address. There is no mailbox synchronization between the email and voicemail mailboxes. For example reading and deletion of the email message does not affect the message in the voicemail mailbox or the message waiting indication provided for that new message.

  - *Forward*
  If this mode is selected, each time a new voicemail message is received in the voicemail mailbox, that message is attached to an email and sent to the email address. No copy of the voicemail message is retained in the voicemail mailbox and their is no message waiting indication. As with Copy, their is no mailbox synchronization between the email and voicemail mailboxes. Users can also select this mode by dialing *01 from their extension.

    - UMS Exchange 2007 *(IP Office 5.0+)*
    With Voicemail Pro, the system supports voicemail email to an Exchange 2007 server email account. For users and groups also enabled for UMS Web Services this significantly changes their mailbox operation. The Exchange Server inbox is used as their voicemail message store and features such as message waiting indication are set by new messages in that location rather than the voicemail mailbox on the voicemail server. Telephone access to voicemail messages, including Visual Voice access, is redirected to the Exchange 2007 mailbox.

  - *Alert*
  If this mode is selected, each time a new voicemail message is received in the voicemail mailbox, a simple email message is sent to the email address. This is an email message announcing details of the voicemail message but with no copy of the voicemail message attached. Users can also select this mode by dialing **02* from their extension.

5.8.4 Fallback

Fallback settings can be used to make a hunt group unavailable and to set where the hunt group's calls should be redirected at such times. Hunt groups can be manually placed In Service, Out of Service or in Night Service. Additionally using a time profile, a group can be automatically placed in Night Service when outside the Time Profile settings.

Summary: Fallback redirects a hunt group's calls when the hunt group is not available, for example outside normal working hours. It can be triggered either manually or using an associated time profile.

- Hunt Group Service States
  A hunt group can be in one of three states; In Service, Out of Service and Night Service. When In service, calls are presented as normal. In any other state calls are redirected.
- Call Redirection During Fallback
  The following options are possible when a hunt group is either Out of Service or in Night Service.
  - Fallback Group
    If an Out of Service Fallback Group or Night Service Fallback Group has been set, calls are redirected to that group.
  - Voicemail
    If no fallback group has been set but voicemail is available, calls are redirected to voicemail.
  - Busy Tone
    If no fallback group has been set and voicemail is not available, busy tone is returned to calls.
- Manually Controlling the Service State
  Manager and or short codes can be used to change the service state of a hunt group. The short code actions can also be assigned to programmable buttons on phones.
  - The  icon is used for a hunt group manually set to Night Service mode.
  - The  icon is used for a hunt group manually set to Out of Service mode.
- Time Profile
  A time profile can be associated with the hunt group. When outside the time profile, the hunt group is automatically place into night service. When inside the time profile, the hunt group uses manually selected mode.
  - When outside the time profile and therefore in night service, manual night service controls cannot be used to override the night service. However the hunt group can be put into out of service.
  - When a hunt group is in Night Service due to a time profile, this is not indicated within Manager.
  - IP Office 4.0+: Time profile operation does not affect hunt groups set to Out of Service.

Hunt Group | Fallback	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Time Profile: *Default = <None> (No automatic night service)*
  This field allows selection of a previously created Time Profile 360. That profile then specifies the times at which it should use the manually selected Service Mode settings. Outside the period defined in the time profile, the hunt group behaves as if set to Night Service mode.
  - Please note that when a hunt group is in Night Service due to it associated time profile, this is not reflected by the Service Mode on this tab. Note also that the manual controls for changing a hunt group's service mode cannot be used to take a hunt group out of time profile night service.
- Out of Service Fallback Group: *Default = <None> (Voicemail or Busy Tone)*
  This field sets the alternate hunt group destination for calls when this hunt group is in Out of Service mode. If left blank, calls are redirected to voicemail if available or otherwise receive busy tone.
- Night Service Fallback Group: *Default = <None> (Voicemail or Busy Tone)*
  This field sets the alternate hunt group destination for calls when this hunt group is in Night Service mode. If left blank, calls are redirected to voicemail if available or otherwise receive busy tone.
- Service Mode: *Default = In Service*
  This field is used to manually select the current service mode for the hunt group.

- Out of Service
  When selected, calls are redirected using the Out of Service Fallback Group setting. This setting can also be manually controlled using the short code and button programming features Set Hunt Group Out of Service and Clear Hunt Group Out of Service.
- or : In Service
  When selected the hunt group is enabled. This is the default mode.
- Night Service
  When selected, calls are redirected using the Night Service Fallback Group setting. This setting can also be manually controlled using the short code and button programming features Set Hunt Group Night Service and Clear Hunt Group Night Service.

Hunt Group Fallback Controls

Hunt Group Fallback	
Manager	Hunt group fallback selection is done through the Hunt Group | Fallback 314 tab. A time profile if required is set through the Time Profile | Time Profile tab.
Controls	The following short code features/button programming actions can be used:

Feature/Action	Short Code	Default	Button
Set Hunt Group Night Service 493	✓	*20*N#	✓ - Toggles.
Clear Hunt Group Night Service 453	✓	*21*N#	✓
Set Hunt Group Out of Service 493	✗	✗	✓- Toggles.
Clear Hunt Group Out of Service 453	✗	✗	✓

Note that for a hunt group using a time profile, these controls only are only applied when the hunt group is within the specified time profile period. When outside its time profile, the hunt group is in night service mode and cannot be overridden.

Phone Manager	There are no specific controls for the operation of hunt group fallback.
SoftConsole	There are no specific controls for the operation of hunt group fallback.
Voicemail	There are no specific controls for the operation of hunt group fallback.

5.8.5 Queuing

When is a Call Queued

The definition of when a call is in a queue can vary:

- Pre-IP Office 4.0: Calls to a hunt group were only queued when the number of calls waiting exceeded the number of available hunt group members that could be ringing. Using that definition, calls that were actually ringing were not regarded as queued.
- IP Office 4.0+: Any calls waiting to be answered at a hunt group are regarded as being queued. The Normalise Queue Length control allows selection of whether features that are triggered by the queue length should include or exclude ringing calls.

- Additional Calls
  Once one call is queued, any further calls are also queued. When an available hunt group member becomes idle, the first call in the queue is presented.
- How Many Calls Can be Queued?
  Calls are added to the queue until the hunt group's Queue Limit, if set, is reached.
  - When the queue limit is reached, any further calls are redirected to the hunt group's voicemail if available.
  - If voicemail is not available excess calls receive busy tone. An exception to this are analog trunk and T1 CAS trunk calls which will remain queued regardless of the queue limit if no alternate destination is available.
  - If an existing queued call is displaced by a higher priority call, the displaced call will remain queued even if it now exceeds the queue limit.
- Queue Announcements
  The method of hunt group announcements depends on the system software level:
  - Pre-IP Office 4.0: Systems with Voicemail Pro or Voicemail Lite, announcements are applied to queued calls.
  - IP Office 4.0+: Hunt group announcements are separate from queuing. Announcements can be used even if queuing is turned off and are applied to ringing and queued calls. See Hunt Group | Announcements 320.
- Queue Monitoring
  There are several methods of displaying a hunt group queue.
  - Group Button
    On phones, with programmable buttons, the Group function can be assigned to monitor a specified group. The button indicates when there are calls ringing within the group and also when there are calls queued. The button can be used to answer the longest waiting call.
  - Phone Manager and SoftConsole
    Both these applications can display queue monitors for selected hunt groups, 2 using Phone Manager, 7 using SoftConsole. This requires the hunt group to have queuing enabled. These queues can be used to answer calls.
- What Happens When A Hunt Group Members Becomes Available
  When a hunt group member becomes available, the first call in the queue is presented to that member. If several members become available, the first call in the queue is simultaneously presented to all the free members.
- Overflow Calls
  Calls that overflow are counted in the queue of the original hunt group from which they overflow and not that of the hunt group to which they overflow. This affects the Queue Limit and Calls in Queue Threshold.

Settings

Hunt Group | Queuing	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Queuing On: *Default = On*
  This settings allows calls to this hunt group to be queued. The normal icon is replaced .
- Queue Limit: *Default = No Limit. Range = No Limit, 1 to 999 calls.*
  This setting can be used to limit the number of calls that can be queued. Calls exceeding this limit are passed to voicemail if available or otherwise receive busy tone. This value is affected by Normalize Queue Length setting.

- If voicemail is not available excess calls receive busy tone. An exception to this is analog trunk and T1 CAS trunk calls which will remain queued regardless of the queue limit if no alternate destination is available. This is due to the limited call status signalling supported by those trunks which would otherwise create scenarios where the caller has received ringing from the local line provider and then suddenly gets busy from the system, creating the impression that the call was answered and then hung up.
- If priority is being used with incoming call routes, high priority calls are place ahead of lower priority calls. If this would exceed the queue limit the limit is temporarily increased by 1.
- If an existing queued call is displaced by a higher priority call, the displaced call will remain queued even if it now exceeds the queue limit.

- **Normalize Queue Length**: *Default = Off. Software level = 4.0+.*
Pre-IP Office 4.0: The call queue only included calls waiting to ring and did not include calls actually ringing. IP Office 4.0+: Calls both waiting to ring and ringing are regarded as being queued. This therefore affects the use of the Queue Limit and Calls in Queue Alarm thresholds. If Normalize Queue Length is enabled, the number of hunt group members logged in and not on DND is added to those thresholds.
  - Example: A customer has two products that it is selling through a call center with 10 available agents; one product with a $10 margin and one with a $100 margin. Separate hunt groups with the same 10 members are created for each product.
    - The $100 product has a Queue Limit of 5 and Normalize Queue Length is on. The maximum number of $100 calls that can be waiting to be answered will be 15 (10 ringing/connected + 5 waiting to ring).
    - The $10 product has a Queue Limit of 5 and Normalize Queue Length is off. The maximum number of $10 calls that can be waiting to be answered is 5 (5 ringing/connected).

- **Queue Type**: *Default = Assign Call On Agent Answer. Software level = 4.2+.*
When queuing is being used, the call that the agent receives when they answer can be assigned in one of two ways:
  - **Assign Call On Agent Answer**
  In this mode the call answered by the hunt group member will always be the longest waiting call of the highest priority. The same call will be shown on all ringing phones in the group. At the moment of answering that may not necessarily be the same call as was shown by the call details at the start of ringing. This is the default mode for IP Office 4.0+.
  - **Assign Call on Agent Alert**
  In this mode, once a call has been presented to a hunt group member, that is the call they will answer if they go off hook. This is similar to the method used for IP Office 3.2 and earlier. This mode should be used when calls are being presented to applications which use the call details such as a fax server, CTI or TAPI.

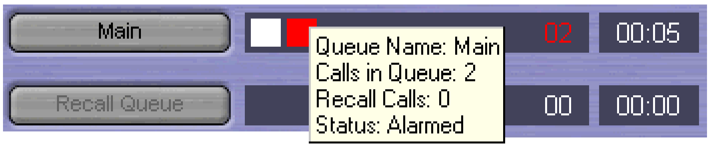
- **Queue Ring Time (secs)**: *Default = 10 seconds. Range = 0 to 99999 seconds. Software level = Up to 3.2 only.*
On systems with Voicemail Lite (pre-IP Office 5 only) or Voicemail Pro, the voicemail system can provide announcements to queued callers. This setting controls the time before the first queued announcement is played to a queued caller. For IP Office 4.0+ this has been replaced by the Hunt Group | Announcement 320 tab controls.

- **Calls In Queue Alarm**: *Software Level = 4.1+.*
The system can be set to send an alert to a specified extension when the number of calls queued for the hunt group reaches the specified threshold.
  - **Calls In Queue Threshold**: *Default = Off. Range = 1 to 99. Software level = 4.1+.*
  Alerting is triggered when the number of queued calls reaches this threshold. Alerting will stop only when the number of queued calls drops back below this threshold. This value is affected by Normalize Queue Length setting above.
  - **Analog Extension to Notify**: *Default = <None>. Software Level = 4.1+.*
  This should be set to the extension number of a user associated with an analog extension. The intention is that this analog extension port should be connected to a loud ringer or other alerting device and so is not used for making or receiving calls. The list will only shown analog extensions that are not members of any hunt group or the queuing alarm target for any other hunt group queue. The alert does not follow user settings such as forwarding, follow me, DND, call coverage, etc or receive ICLID information.

Hunt Group Queue Settings	
Manager	Hunt group queuing is enabled using the Queuing On option on the Hunt Group | Queuing 316 tab. When enabled, the icon is used for the hunt group.
Controls	The following short code features/button programming actions can be used:

Feature/Action	Short Code	Default	Button
Group	✗	✗	✓

Phone Manager	Phone manager Pro can be used to monitor up to two hunt group queues. This is configured by clicking and then on the Queue ID tab selecting the two hunt groups. During normal operation the Phone Manager user then has access to a Queue tab which is automatically given focus when calls become queued.

SoftConsole	SoftConsole can display up to 7 hunt group queues (an eight queue is reserved for recall calls). They are configured by clicking and selecting the Queue Mode tab. For each queue alarm threshold can be set based on number of queued calls and longest queued call time. Actions can then be selected for when a queue exceeds its alarm threshold; Automatically Restore SoftConsole, Ask me whether to restore SoftConsole or Ignore the Alarm.  Within the displayed queues, the number of queued calls is indicated and the time of the longest queued call is shown. Exceeding an alarm threshold is indicated by the queue icons changing from white to red. The longest waiting call in a queue can be answered by clicking on the adjacent button.

5.8.6 Voice Recording

This tab is used to configure automatic recording of external calls handled by hunt group members. IP Office 6.1: The recording of internal calls as well is also supported.

Call recording requires Voicemail Pro to be installed and running. Call recording also requires available conference resources similar to a 3-way conference.

- IP Office 4.0+ introduces the following changes to recording:
  - Calls to and from IP devices, including those using Direct media, can be recorded.
  - Calls parked or held pause recording until the unparked or taken off hold.
  - Recording is stopped if:
    - User recording stops if the call is transferred to another user.
    - User account code recording stops if the call is transferred to another user.
    - Hunt group recording stops if the call is transferred to another user who is not a member of the hunt group.
    - Incoming call route recording continues for the duration of the call on the system.
- IP Office 4.1+: A destination mailbox other than the hunt group's own mailbox can be specified as the destination for recordings.

Hunt Group | Voice Recording	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- **Record Inbound**: *Default = None*
  Select whether automatic recording of incoming calls is enabled. Options for recording are:
  - **None**: Do not automatically record calls.
  - **On**: Record the call if possible. If not possible to record, allow the call to continue.
  - **Mandatory**: Record the call if possible. If not possible to record, block the call and return busy tone.
  - **Percentages of calls**: Record a selected percentages of the calls.
- **Record Time Profile**: *Default = <None> (Any time)*
  Used to select a time profile 360 during which automatic call recording of incoming calls is applied. If no profile is selected, automatic recording of incoming calls is active at all times.
- **Recording (Auto)**: *Default = Mailbox*
  Sets the destination for automatically triggered recordings.
  - **Mailbox**
    This option sets the destination for the recording to be a selected user or hunt group mailbox. The adjacent drop down list is used to select the mailbox.
  - **Voice Recording Library**: *Software level = 3.0+.*
    This options set the destination for the recording to be a VRL folder on the voicemail server. The ContactStore application polls that folder and collects waiting recordings which it then places in its own archive. Recording is still done by the Voicemail Pro.
- **Auto Record Calls**: *Default = External. Software level = 6.1*
  This setting allows selection of whether *External* or *External & Internal* calls are subject to automatic call recording.

5.8.7 Announcements

This tab is new for IP Office 4.0 and higher. Unlike pre-4.0 systems, this method of using announcements is independent of hunt group queuing.

Announcements are played to callers waiting to be answered. This includes callers being presented to hunt group members, ie. ringing, and callers queued for presentation.

- The system supports announcements using Voicemail Pro, Voicemail Lite (pre-5.0 only) or Embedded Voicemail.
- If no voicemail channel is available for an announcement, the announcement is not played.
- In conjunction with Voicemail Pro, the system allows a number of voicemail channels to be reserved for announcements. See System | Voicemail 157.
- With Voicemail Pro, the announcement can be replaced by the action specified in a Queued (1st announcement) or Still Queued (2nd announcement) start point call flow. Refer to the Voicemail Pro Installation and Maintenance documentation for details.
- Calls can be answered during the announcement. If it is a mandatory requirement that announcements should be heard before a call is answered, then a Voicemail Pro call flow should be used before the call is presented.
  - **Warning: Call Billing and Logging**
    Note that a call becomes connected when the first announcement is played to it. That connected state is signaled to the call provider who may start billing at that point. The call will also be recorded as answered within the SMDR 898 output once the first announcement is played.
- If a call is rerouted to a hunt group's Night Service Group or Out of Service Fallback Group, the announcements of the new group are applied.
- If a call overflows, the announcements of the original group are still applied, not those of the overflow group.
- For announcements to be used effectively, the hunt group's **Voicemail Answer Time** must be extended or **Voicemail On** must be unselected.

Recording the Hunt Group Announcement
Voicemail Pro provides the default announcement *"I'm afraid all the operators are busy but please hold and you will be transferred when somebody becomes available"*. This default is used for announcement 1 and announcement 2 if no specific hunt group announcement has been recorded. Embedded Voicemail does not provide any default announcement. Voicemail Lite also provides the default announcements.

The maximum length for announcements is 10 minutes. New announcements can be recorded using the following methods:

- **Voicemail Lite**
  Access the hunt group mailbox and press 3. Then press either 3 to record the 1st announcement for the hunt group or 4 to record the 2nd announcement for the hunt group.
- **Voicemail Pro - IP Office Mode**
  Access the hunt group mailbox and press 3. Then press either 3 to record the 1st announcement for the hunt group or 4 to record the 2nd announcement for the hunt group.
- **Voicemail Pro - Intuity Emulation Mode**
  There is no mechanism within the Intuity telephony user interface (TUI) to record hunt group announcements. To provide custom announcements, hunt group queued and still queued start points must be configured with Voicemail Pro with the required prompts played by a generic action.
- **Embedded Voicemail**
  Embedded Voicemail does not include any default announcement or method for recording an announcement. The **Record Message** 486 short code feature is provided to allow the recording of announcements. The telephone number field of short codes using this feature requires the extension number followed by either ".1" for announcement 1 or ".2" for announcement 2. For example, for extension number 300, the short codes ***91N# | Record Message | N".1"*** and ***92N# | Record Message | N".2"*** could be used to allow recording of the announcements by dialing ***91300#*** and ***92300#***.

Hunt Group | Announcements	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	✓.

- Announcements On: *Default = Off.*
  This setting enables or disables announcements.

- Wait before 1st announcement: *Default = 10 seconds. Range = 0 to 9999 seconds.*
  This setting sets the time delay from the calls presentation, after which the first announcement should be played to the caller. If Synchronize Calls is selected, the actual wait may differ, see below.

- Flag call as answered: *Default = Off.*
  This setting is used by the CCC and CBC applications. By default they do not regarded a call as answered until it has been answered by a person or by a Voicemail Pro action with Flag call as answered selected. This setting allows calls to be marked as answered once the caller has heard the first announcement. This setting is not used by the Customer Call Reporter application.

  - Warning: Call Billing and Logging
    Note that a call becomes connected when the first announcement is played to it. That connected state is signaled to the call provider who may start billing at that point. The call will also be recorded as answered within the SMDR 898 output once the first announcement is played.

- Post announcement tone: *Default = Music on hold.*
  Following the first announcement, you can select whether the caller should hear *Music on Hold* 763, *Ringing* or *Silence* until answered or played another announcement.

- 2nd Announcement: *Default = On.*
  If selected, a second announcement can be played to the caller if they have still not been answered.

- Wait before 2nd announcement: *Default = 20 seconds. Range = 0 to 9999 seconds.*
  This setting sets the wait between the 1st and the 2nd announcement. If Synchronize Calls is selected, the actual wait may differ, see below.

- Repeat last announcement: *Default = On.*
  If selected, the last announcement played to the caller is repeated until they are answered or hang-up.

- Wait before repeat: *Default = 20 seconds. Range = 0 to 9999 seconds.*
  If Repeat last announcement is selected, this setting sets is applied between each repeat of the last announcement. If Synchronize Calls is selected, this value is grayed out and set to match the Wait before 2nd announcement setting.

- Synchronize calls: *Default = Off*
  This option can be used to restrict how many voicemail channels are required to provide the announcements.

  - *Off*
    When Synchronize calls is off, announcement are played individually for each call. This requires a separate voicemail channel each time an announcement is played to each caller. While this ensures accurate following of the wait settings selected, it does not make efficient use of voicemail channels.

  - *On*
    When Synchronize calls is on, if a required announcement is already being played to another caller, further callers wait until the announcement been completed and can be restarted. In addition, when a caller has waited for the set wait period and the announcement is started, any other callers waiting for the same announcement hear it even if they have not waited for the wait period. Using this setting, the maximum number of voicemail channels ever needed is 1 or 2 depending on the number of selected announcements.

    - Note: Interaction with Voicemail Pro Queued and Still Queued Start Points
      If either custom Queued or Still Queued start point call flows are being used for the announcements, when Synchronize Calls is enabled those call flows will support the playing of prompts only. Voicemail Pro actions such as Speak ETA, Speak Position, Menu, Leave Mail, Transfer and Assisted Transfer, etc. are not supported.

5.8.8 SIP

Each hunt group can be configured with its own SIP URI information. For calls received on a SIP line where any of the line's SIP URI fields are set to *Use Internal Data*, if the call is presented to the hunt group that data is taken from these settings.

This form is hidden if there are no system SCN lines in the configuration or no SIP lines with a URI set to *Use Internal Data*.

Hunt Group | Voice Recording	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	6.0+.
Mergeable	✓.

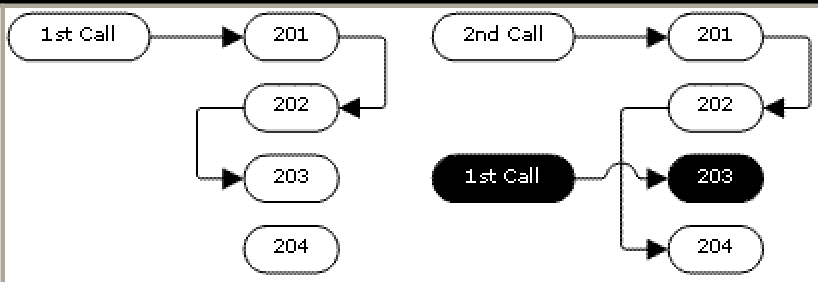
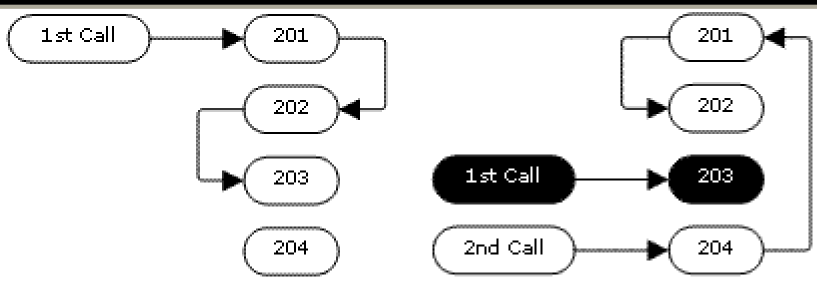
- SIP Name: *Default = User name.*
  The value from this field is used when the From field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- SIP Display Name (Alias): *Default = User name.*
  The value from this field is used when the Display Name field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- Contact: *Default = User name.*
  The value from this field is used when the Contact field of the SIP URI being used for a SIP call is set to *Use Internal Data*.
- Anonymous: *Default = Off.*
  If the From field in the SIP URI is set to *Use Internal Data*, selecting this option inserts *Anonymous* into that field rather than the SIP Name set above.

5.8.9 Hunt Group Operation

5.8.9.1 Hunt Types

At its most basic, a hunt groups settings consist of a hunt group name, an extension number, a list of hunt group members and a hunt type selection. It is the last two settings which determine the order in which incoming calls are presented to hunt group members.

The available hunt types are; Collective, Sequential, Rotary and Longest Waiting. These work are follows:

Collective Group (formerly Group Group)	
An incoming call is presented simultaneously to all the available hunt group members.	
Sequential Group (formerly Hunt or Linear Group)	
An incoming call is presented to the first available member in the list. If unanswered, it is presented to the next available member in the list. The next incoming call uses the same order. It is presented to the available members starting again from the top of the list.	
Rotary Hunt Type (formerly Circular Group)	
This hunt type operates similarly to Sequential. However the starting point for call presentation is the first available member after the last member to answer a call.	
Longest Waiting Hunt Type (formerly Idle or Most Idle)	
This hunt type does not present calls to hunt group members in the order that they are listed. It presents calls using the order of how long the available hunt group members have been idle. An incoming call is first presented to the available member who has been idle the longest. If unanswered it is presented to the next longest idle member. • IP Office 4.2+: Where hunt group calls are being presented to a twinned extension, the longest waiting status of the user can be reset by calls answered at either their master or twinned extension.	

5.8.9.2 Call Presentation

Summary: Calls are presented to each available hunt group member in turn. If having been presented to all the available members, none answers, the call is redirected to voicemail if available, otherwise it continues to be presented to the next available member.

In addition to the summary, options exist to have calls queued or to have calls redirected to overflow groups.

- First and Next Available Members
  The first available member to which a call is presented and the order of the next available members to which a call is presented are determined by the hunt group's Hunt Type 323 setting.

- Additional Calls
  When additional calls are waiting to be presented, additional available hunt group members are alerted using the hunt group type. The way additional calls are presented if there are available members depends on the system software level.

  - Pre-IP Office 4.0
    Additional calls ring around the group separately. This means that additional calls may be answered ahead of the first call.

  - IP Office 4.0 and Higher
    When any member answers a call it will be the first waiting call that is answered.

- No Available Members
  If the number of incoming calls exceeds the number of available members to which calls can be presented, the following actions are usable in order of precedence.

  - Queuing
    If queuing has been enabled for the hunt, it is applied to the excess calls up to the limits specified for the number of queued calls or length of time queued.

  - Voicemail
    If voicemail has been enabled for the hunt group, excess calls are directed to voicemail.

  - Busy Tone
    Busy tone is returned to the excess calls (except analog and T1 CAS calls which remain queued).

- No Answer Time
  This value is used to determine how long a call should ring at a hunt group member before being presented to the next available hunt group member. The System | Telephony | Telephony | No Answer Time setting is used unless a specific Hunt | Hunt Group | No Answer Time is set.

- Voicemail
  If voicemail is being used, if having been presented to all the available group members the call is still not answered then it goes to voicemail.

  - IP Office 4.0+: The call will also go to voicemail when the hunt group's Voicemail Answer Time is exceeded. the mailbox of the originally targeted hunt group is used even if the call has overflowed or gone to a night server hunt group.

  - Pre-IP Office 4.0: When voicemail was invoked, the mailbox of whichever hunt group was currently handling the call was used, for example the mailbox of the overflow or night service hunt group might be used if the call had gone from the original hunt group to an overflow or night server hunt group.

- Calls Not Being Answered Quick Enough - Overflow
  In addition to ringing at each available member for the No Answer Time, a separate Overflow Time can be set. When a calls total ring time against the group exceeds this, the call can be redirected to an overflow group or groups.

- No Available Member Answers
  If a call has been presented unanswered to all the available members, either of two actions can be applied. If voicemail is available, the call is redirected to voicemail. If otherwise, the call will continue being presented to hunt group members until answered or, if set, overflow is used.

- Call Waiting
  For hunt groups using the Group hunt type, call waiting can be used.

5.8.9.3 Member Availability

Summary: Details when a hunt group member is seen as being available to be presented a hunt group call.

The Hunt Group settings within Manager list those users who are members of the hunt group and therefore may receive calls directed to that hunt group. However there are a range of factors that can affect whether a particular hunt group member is available to take hunt group calls at any time.

- Existing Connected Call
  Users with an existing connected call are not available to further hunt group calls. This is regardless of the type of connected call, whether the user has available call appearance buttons or is using call waiting.
  - Hunt Group Call Waiting
    For Collective hunt groups call waiting can be enabled using the Ring Type of *Collective Call Waiting*.
- Logged In/Logged Out
  The system allows user's to log in and out extensions, a process known as 'hot desking'. Whilst a user is logged out they are not available to receive hunt group calls.
  - IP Office 4.2+: Mobile Twinning users with both Hunt group calls eligible for mobile twinning and Twin when logged out selected will still receive hunt group calls unless they switch off twinning.
- Membership Enabled/Disabled
  The system provides controls to temporarily disable a users' membership of a hunt group. Whilst disabled, the user is not available to receive calls directed to that hunt group.
- Do Not Disturb
  This function is used by users to indicate that they do not want to receive any calls. This includes hunt group calls. In call center environments this state is also known as 'Busy Not Available'. See Do Not Disturb 769.
- Busy on Held
  When a user has a held call, they can receive other calls including hunt group calls. The Busy on Held settings can be used to indicate that the user is not available to further calls when they have a held call.
- Forward Unconditional
  Users set to Forward Unconditional are by default not available to hunt group calls. The system allows the forwarding of hunt group calls to be selected as an option.
- Idle /Off Hook
  The hunt group member must be idle in order to receive hunt group call ringing.
- No Available Members
  If queuing has been enabled, calls will be queued. If queuing has not been enabled, calls will go to the overflow group if set, even if the overflow time is not set or is set to 0. If queuing is not enabled and no overflow is set, calls will go to voicemail. If voicemail is not available, external calls go to the incoming call routes fallback destination while internal calls receive busy indication.

Hunt Group Member Availability Settings

Manager	Forwarding and do not disturb controls for a user are found on the User | Forwarding [281] and User | DND [270] tabs. Enabling and disabling a users hunt group membership is done by ticking or unticking the user entry in the hunt group's extensions list on the Hunt Group | Hunt Group [306] tab.
Controls	The following short code features/button programming actions can be used:

Feature/Action	Short Code	Default	Button
Hunt Group Enable [480]	✓	✗	✓HGEna - Toggles.
Hunt Group Disable [480]	✓	✗	✓HGDis
Forward Hunt Group On [472]	✓	✓-*50	✓FwDH+ - Toggles
Forward Hunt Group Off [472]	✓	✓-*51	✓FwDH-
Busy on Held [441]	✓	✗	✓BusyH
Do Not Disturb On [466]	✓	✓-*08	✓DNDOn - Toggles
Do Not Disturb Off [466]	✓	✓-*09	✓DNDOf
Extn Login [468]	✓	✓-*35*N#	✓Login
Extn Logout [468]	✓	✓-*36	✓Logof

Phone Manager	DND, Forwarding and Busy on Held can all be controlled through Phone Manager. They are accessed by clicking and then selecting the Do Not Disturb, Forwarding or Telephone tabs respectively. Phone Manager Pro users can select agent mode by clicking , selecting the Agent Mode tab and selecting Agent Mode. In this mode, Phone Manager provides icons for Busy Wrap Up (Hunt group disable) and Busy Not Available (DND). It also allows individual selection of which group memberships are enabled. Phone Manager can also be used to log in and log out when the application is started or stopped.
SoftConsole	A SoftConsole user can view and edit a user's settings. Through the directory, select the required user. Their current status including DND, Logged In and hunt group membership states are shown and can be changed. Forwarding settings can be accessed by then selecting Forwarding.

5.8.9.4 Example Hunt Group

The follow are simple examples of how a department might use the facilities of a hunt group.

1. Basic Hunt Group	
Scenario	The Sales department want all sales related calls to be presented first to Jane, then Peter and finally Anne.
Actions	1. Create a hunt group named Sales and assign it an extension number. 2. Set the Hunt Type to Sequential. 3. Add Jane, Peter and Ann to the User List in that order. 4. Turn off queuing on the Queuing tab and voicemail on the Voicemail tab. 5. Route relevant calls to the Sales group by selecting it as the destination in the appropriate Incoming Call Routes.
Results	Any call received by the Sales hunt group is first presented to Jane if she is available. If Jane is not available or does not answer within 15 seconds the call is presented to Peter. If Peter is not available or does not answer within 15 seconds the call goes Anne. Since voicemail is not on, the call will continue to be presented around the group members in that order until it is answered or the callers hangs up.

2. Adding Voicemail Support	
Scenario	A voicemail server has now been added to the system. The Sales department wants to use it to take messages from unanswered callers. When messages are left, they want Jane to receive message waiting indication.
Actions	1. Open the Sales hunt group settings and select Voicemail On on the Voicemail tab. 2. Select the User settings for Jane. On the Source Numbers tab, add the entry HSales.
Results	Once a call to the Sales group has been presented to all the available members, if it is still unanswered then the call will be redirected to the group's voicemail mailbox to leave a message. When a message has been left, the message waiting indication lamp on Jane's phone is lit.

3. Using the Queuing Facility	
Scenario	The Sales department now wants calls queued when no one is available to answer. However if the number of queued calls exceeds 3 they then want any further callers directed to voicemail.
Actions	1. Open the Sales hunt group settings and select Queuing On on the Queuing tab. 2. Set the Queue Limit to 3.
Results	When the Sales group are all on calls or ringing, any further calls to the group are queued and receive queuing announcements from the voicemail server. When the number of queued calls exceeds 3, any further calls are routed to the group's voicemail mailbox.

4. Using Out of Service Fallback	
Scenario	During team meetings, the Sales department want their calls redirected to another group, for this example Support.
Actions	1. Open the Sales hunt group settings and select the Fallback tab. In the Out of Service Fallback Group field select the Support group. 2. Create a system short code *98 / 300 / Set Hunt Group Out of Service. 3. Create a system short code *99 / 300 / Clear Hunt Group Out of Service.
Results	Prior to team meetings, dialing *98 puts the Sales group into out of service mode. Its calls are then redirected to the Support group. Following the meeting, dialing *99 puts the Sales group back In Service.

5. Using a Night Service Time Profile	
Scenario	Outside their normal business hours the Sales department want their group calls automatically sent to voicemail. This can be done using a time profile and leaving the Night Service Fallback Group setting blank.
Actions	1. Create a Time Profile called Sales Hours and in it enter the time during which the Sales department are normally available. 2. Open the Sales hunt group settings and select the Fallback tab. 3. In the Time Profile field select Sales Hours.
Results	Outside the normal business hours set in the time profile, the Sales hunt group is automatically put into Night Service mode. Since no Night Service Fallback Group has been set, calls are redirected to voicemail.

5.8.9.5 CBC/CCC Agents and Hunt Groups

The use of and reporting on hunt groups is a key feature of call center operation. For Manager, reporting is provided through the Compact Business Center (CBC) or Compact Contact Center (CCC) applications.

In order for these applications to provide hunt group and hunt group user (agent) reports, the following rules apply:

- The hunt group names must be restricted to a maximum of 12 characters.
- The hunt group and user extension numbers should be a maximum of 4 digits.
- Hunt group members should be given a Login Code and set to Force Login.
- The agent state Busy Not Available is equivalent to Do Not Disturb. The agent state Busy Wrap Up is equivalent to hunt group disable.

5.9 Short Code

This form is used to create System Short Codes. System short codes can be dialed by all system users. However the system short code is ignored if the user dialing matches a user or user rights short code. For full details on short code usage and parameter see the section Short Codes 426.

Short Code | Short Code	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Short Code 427
  The dialing digits used to trigger the short code. Maximum length 31 characters.
- Telephone Number 427
  The number dialed by the short code or parameters for the short code feature. This field can contain numbers and characters. For example, it can contain Voicemail Pro start point names, user names, hunt group names and telephone numbers (including those with special characters). Maximum length 31 characters.
  - Dialing Complete
    The majority of North-American telephony services use 'en-bloc' dialing, ie. they expect to receive all the routing digits for a call as a single simultaneous set of digits. Therefore the use of a ; is recommended at the end of all dialing short codes that use an N. This is also recommended for all dialing where secondary dial tone short codes are being used.
- Line Group ID: *Default = 0*
  For short codes that result in the dialing of a number, that is short codes with a Dial feature, this field is used to enter the initially routing destination of the call. The drop down can be used to select the following from the displayed list:
  - Outgoing Group ID
    The Outgoing Group ID's current setup within the system configuration are listed. If an Outgoing Group ID is selected, the call will be routed to the first available line or channel within that group.
  - ARS *(IP Office 4.0+)*
    The ARS entries currently configured in the system are listed. If an ARS entry is selected, the call will be routed by the setting within that ARS entry. Refer to ARS Overview 407.
- Feature 439
  Select the action to be performed by the short code.
- Locale 844: *Default = Blank*
  For short codes that route calls to voicemail, this field can be used to set the prompts locale that should be used if available on the voicemail server.
  - When the system routes a call to the voicemail server it indicates the locale for which matching prompts should be provided if available. The locale sent to the voicemail server by the system is determined as follows:

Locale Source	Usage
Short Code	The short code locale, if set, is used if the call is routed to voicemail using the short code.
System	If no user or incoming call route locale is set system locale is used unless overridden by a short code locale.
Incoming Call Route	The incoming call route locale, if set, is used if caller is external.
User	The user locale, if set, is used if the caller is internal.

- Force Account Code: 375 *Default = Off.*
  For short codes that result in the dialing of a number, this field trigger the user being prompted to enter a valid account code before the call is allowed to continue.
- Force Authorization Code 404: *Default = Off*
  This option is only shown on systems where authorization codes have been enabled. If selected, then for short codes that result in the dialing of a number, the user is required to enter a valid authorization code in order to continue the call.

5.10 Service Settings

Services are used to configure the settings required when a user or device on the LAN needs to connect to a off-switch data service such as the Internet or another network. Services can be used when making data connections via trunk or WAN interfaces.

Once a service is created, it can be used as the destination for an IP Route entry. One service can also be set as the Default Service. That service will then be used for any data traffic received by the system for which no IP Route is specified.

The system supports three types of service:

-  Normal Service
  This type of service should be selected when for example, connecting to an ISP.

-  WAN Service
  This type of service is used when creating a WAN link. A User and RAS Service will also be created with the same name. These three entries are automatically linked and each open the same form. Note however, that this type of Service cannot be used if the Encrypted Password option is checked. In this case the RAS Service name must match the Account Name. Therefore either create each entry manually or create an Intranet Service.

-  Intranet Service
  This type of service can be selected to automatically create a User with the same name at the same time. These two entries are linked and will each open the same form. The User's password is entered in the Incoming Password field at the bottom on the Service tab. An Intranet Services shares the same configuration tabs as those available to the WAN Service.

5.10.1 Service

Service | Service	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Name
  The name of the service. It is recommended that only alphanumeric characters be used.
- Account Name
  The User Name that is used to authenticate the connection. This is provided by the ISP or remote system.
- Password: *Default = Blank*
  Enter the password that is used to authenticate the connection. This is provided by the ISP or remote system.
- Telephone Number: *Default = Blank*
  If the connection is to be made via ISDN enter the telephone number to be dialed. This is provided by the ISP or remote system.
- Firewall Profile: *Default = Internet01 if present, otherwise <None>*
  From the list box select the Firewall Profile that is used to allow/disallow protocols through this Service.
- Encrypted Password: *Default = Off*
  When enabled the password is authenticated via CHAP (this must also be supported at the remote end). If disabled, PAP is used as the authentication method.
- Default Route: *Default = Off*
  When enabled this Service is the default route for data packets unless a blank IP Route has been defined in the system IP Routes 367. A green arrow appears to the left of the Service in the Configuration Tree. Only one Service can be the default route. If disabled, a route must be created under IP Route.
- Incoming Password: *Default = Blank*
  Shown on WAN and Intranet services. Enter the password that will be used to authenticate the connection from the remote Control Unit. (If this field has appeared because you have created a Service and User of the same name, this is the password you entered in the User's Password field).

5.10.2 Bandwidth

These options give the ability to make ISDN calls between sites only when there is data to be sent or sufficient data to warrant an additional call. The calls are made automatically without the users being aware of when calls begin or end. Using ISDN it is possible to establish a data call and be passing data in less that a second. Note: the system will check Minimum Call Time first, then Idle Period, then the Active Idle Period.

Service | Bandwidth	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Minimum No of Channels: *Default = 1. Range = 1 to 30.*
  Defines the number of channels used to connect for an outgoing connection. The initial channel must be established and stable, before further calls are made.
- Maximum No of Channels: *Default = 1. Range = 1 to 30.*
  Defines the maximum number of channels to can be used. This field should contain a value equal to or greater than the Minimum Channels field.
- Extra BW Threshold: *Default = 50%. Range = 0 to 100%.*
  Defines the utilization threshold at which extra channels are connected. The value entered is a %. The % utilization is calculated over the total number of channels in use at any time, which may be one, two etc.
  - For example, if Minimum Channels set to 1, Maximum Channels set to 2 and Extra Bandwidth set to 50 - once 50% of first channel has been used the second channel is connected.
- Reduce BW Threshold: *Default = 10%. Range = 0 to 100%.*
  Defines the utilization threshold at which additional channels are disconnected. The value entered is a %. Additional calls are only dropped when the % utilization, calculated over the total number of channels in use, falls below the % value set for a time period defined by the Service-Idle Time. The last call (calls - if Minimum Calls is greater than 1) to the Service is only dropped if the % utilization falls to 0, for a time period defined by the Service-Idle Time. Only used when 2 or more channels are set above.
  - For example, if Minimum Channels set to 1, Maximum Channels set to 2 and Reduce Bandwidth is set to 10 - once the usage of the 2 channels drops to 10% the number of channels used is 1.
- Callback Telephone Number: *Default = Blank*
  The number that is given to the remote service, via BAP, which the remote Control Unit then dials to allow the bandwidth to be increased. Incoming Call routing and RAS Services must be appropriately configured.
- Idle Period (secs): *Default = 10 seconds. Range = 0 to 999999 seconds.*
  The time period, in seconds, required to expire after the line has gone idle. At this point the call is considered inactive and is completely closed.
  - For example, the 'Idle Period' is set to X seconds. X seconds before the 'Active Idle Period' timeouts the Control Unit checks the packets being transmitted/received, if there is nothing then at the end of the 'Active Idle Period' the session is closed & the line is dropped. If there are some packets being transmitted or received then the line stays up. After the 'Active Idle Period' has timed out the system performs the same check every X seconds, until there are no packets being transferred and the session is closed and the line dropped.
- Active Idle Period (secs): *Default = 180 seconds. Range = 0 to 999999 seconds.*
  Sets the time period during which time the line has gone idle but there are still active sessions in progress (for example an FTP is in process, but not actually passing data at the moment). Only after this timeout will call be dropped.
  - For example, you are downloading a file from your PC and for some reason the other end has stopped responding, (the remote site may have a problem etc.) the line is idle, not down, no data is being transmitted/ received but the file download session is still active. After the set time period of being in this state the line will drop and the sessions close. You may receive a remote server timeout error on your PC in the Browser/FTP client you were using.
- Minimum Call Time (secs): *Default = 60 seconds. Range = 0 to 999999 seconds.*
  Sets the minimum time that a call is held up after initial connection. This is useful if you pay a minimum call charge every time a call is made, no matter the actual length of the call. The minimum call time should be set to match that provided by the line provider.
- Extra BW Mode: *Default = Incoming Outgoing*
  Defines the mode of operation used to increases bandwidth to the initial call to the remote Service.
  - *Outgoing Only*
    Bandwidth is added by making outgoing calls.
  - *Incoming Only*
    Bandwidth is added by the remote service calling back on the BACP number (assuming that BACP is successfully negotiated).

- *Outgoing Incoming*
  Uses both methods but bandwidth is first added using outgoing calls.
- *Incoming Outgoing*
  Uses both methods but bandwidth is first added using incoming BACP calls.

5.10.3 IP

The fields in this tab are used to configure network addressing for the services you are running. Depending on how your network is configured, the use of Network Address Translation (NAT) 798 may be required.

Service | IP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- IP Address: *Default = 0.0.0.0 (address assigned by ISP)*
  An address should only be entered here if a specific IP address and mask have been provided by the Service Provider. Note that if the address is in a different domain from the system then NAT is automatically enabled.
- IP Mask: *Default = 0.0.0.0 (use NAT)*
  Enter the IP Mask associated with the IP Address if an address is entered.
- Primary Transfer IP Address: *Default = 0.0.0.0 (No transfer)*
  This address acts as a primary address for incoming IP traffic. All incoming IP packets without a session are translated to this address. This would normally be set to the local mail or web server address.
  - For control units supporting a LAN1 and LAN2, the primary transfer address for each LAN can be set through the System | LAN1 148 and System | LAN2 155 tabs.
- Request DNS: *Default = Off*
  When selected, DNS information is obtained from the service provider. To use this, the DNS Server addresses set in the system configuration (System | DNS 156) should be blank. The PC making the DNS request should have the system set as its DNS Server. For DHCP clients the system will provide its own address as the DNS server.
- Forward Multicast Messages: *Default = On*
  By default this option is on. Multicasting allows WAN bandwidth to be maximized through the reduction of traffic that needs to be passed between sites.
- RIP Mode: *Default = None*
  Routing Information Protocol (RIP) is a method by which network routers can exchange information about device locations and routes. RIP can be used within small networks to allow dynamic route configuration as opposed to static configuration using.
  - *None*
    The LAN does not listen to or send RIP messages.
  - *Listen Only (Passive)*
    Listen to RIP-1 and RIP-2 messages in order to learn RIP routes on the network.
  - *RIP1*
    Listen to RIP-1 and RIP-2 messages and send RIP-1 responses as a sub-network broadcast.
  - *RIP2 Broadcast (RIP1 Compatibility)*
    Listen to RIP-1 and RIP-2 messages and send RIP-2 responses as a sub-network broadcast.
  - *RIP2 Multicast*
    Listen to RIP-1 and RIP-2 messages and send RIP-2 responses to the RIP-2 multicast address.

5.10.4 Autoconnect

Fields in this tab enable you to set up automatic connections to the specified Service.

Service | Autoconnect	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Auto Connect Interval (mins): *Default = 0 (disabled). Range = 0 to 99999 minutes.*
  This field defines how often this Service will automatically be called ("polled"). For example setting 60 means the system will call this Service every hour in the absence of any normally generated call (this timer is reset for every call; therefore if the service is already connected, then no additional calls are made). This is ideal for SMTP Mail polling from Internet Service Providers.

- Auto Connect Time Profile: *Default = <None>*
  Allows the selection of any configured Time Profiles. The selected profile controls the time period during which automatic connections to the service are made. It does NOT mean that connection to that service is barred outside of these hours. For example, if a time profile called "Working Hours" is selected, where the profile is defined to be 9:00AM to 6:00PM Monday to Friday, then automatic connection to the service will not be made unless its within the defined profile. If there is an existing connection to the service at 9:00AM, then the connection will continue. If there is no connection, then an automatic connection will be made at 9:00AM.

5.10.5 Quota

Quotas are associated with outgoing calls, they place a time limit on calls to a particular IP Service. This avoids excessive call charges when perhaps something changes on your network and call frequency increases unintentionally.

Service | Quota	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Quota Time (mins): *Default = 240 minutes. Range = 0 to 99999 minutes.*
  Defines the number of minutes used in the quota. When the quota time is used up no further data can be passed to this service. This feature is useful to stop things like an internet game keeping a call to your ISP open for a long period.
  - Warning: Setting a value here without selecting a Quota period below will stop all further calls after the Quota Time has expired.
- Quota: *Default = Daily, Range = None, Daily, Weekly or Monthly*
  Sets the period during which the quota is applied. For example, if the Quota Time is 60 minutes and the Quota is set to *Daily*, then the maximum total connect time during any day is 60 minutes. Any time beyond this will cause the system to close the service and prevent any further calls to this service. To disable quotas select *None* and set a Quota Time of zero.
  - Note: The ClearQuota 454 feature can be used to create short codes to refresh the quota time.

5.10.6 Fallback

These options allow you to set up a fallback for the Service. For example, you may wish to connect to your ISP during working hours and at other times take advantage of varying call charges from an alternative carrier. You could therefore set up one Service to connect during peak times and another to act as fallback during the cheaper period.

You need to create an additional Service to be used during the cheaper period and select this service from the Fallback Service list box (open the Service form and select the Fallback tab).

If the original Service is to be used during specific hours and the Fallback Service to be used outside of these hours, a Time Profile can be created. Select this Time Profile from the Time Profile list box. At the set time the original Service goes into Fallback and the Fallback Service is used.

A Service can also be put into Fallback manually using short codes, for example:

- Put the service "Internet" into fallback:
  - Short Code: *85
  - Telephone Number: "Internet"
  - Line Group ID: 0
  - Feature: SetHuntGroupNightService

- Take the service "Internet" out of fallback:
  - Short Code: *86
  - Telephone Number: "Internet"
  - Line Group ID: 0
  - Feature: ClearHuntGroupNightService

Service | Fallback	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- **In Fallback:** *Default = Off*
  This option indicates whether the Service is in Fallback or not. A service can be set into fallback using this setting. Alternatively a service can be set into fallback using a time profile or short codes.

- **Time profile:** *Default = <None> (No automatic fallback)*
  Select the time profile you wish to use for the service. The time profile should be set up for the hours that you wish this service to be operational, out of these hours the Fallback Service is used.

- **Fallback Service:** *<None>*
  Select the service that is used when this service is in fallback.

5.10.7 DialIn

Only available for WAN and Intranet Services. This tab is used to define a WAN connection.

Service | Dial In	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

To define a WAN connection

1. Select Add.
2. Enter *WAN* if the service is being routed via a WAN port on a WAN3 expansion module.

5.11 RAS

A Remote Access Server (RAS) is a piece of computer hardware which sits on a corporate LAN and into which employees dial on the public switched telephone network to get access to their email and to software and data on the corporate LAN.

This form is used to create a RAS service that the system offers Dial In users. A RAS service is needed when configuring modem dial in access, digital (ISDN) dial in access and a WAN link. Some systems may only require one RAS service since the incoming call type can be automatically sensed.

RAS | RAS	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Name
  A textual name for this service. If Encrypted Password (see below) is used this name must match the Account Name entered in the Service Form.
- Extension
  Enter an extension number if this service is to be accessed internally.
- COM Port
  For future use.
- TA Enable: *Default = Off*
  Select to enable or disable - if enabled RAS will pass the call onto a TA port for external handling.
- Encrypted Password: *Default = Off*
  This option is used to define whether Dial In users are asked to use PAP or CHAP during their initial log in to the RAS Service. If the Encrypted Password box is checked then Dial In users are sent a CHAP challenge, if the box is unchecked PAP is used as the Dial In Authorization method.

5.11.1 PPP

PPP (Point-to-Point Protocol) is a Protocol for communication between two computers using a Serial interface, typically a personal computer connected by phone line to a server.

RAS | PPP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- CHAP Challenge Interval (secs): *Default = 0 (disabled). Range = 0 to 99999 seconds.*
  The period between successive CHAP challenges. Blank or 0 disables repeated challenges. Some software, for example Windows 95 DUN, does not support repeated CHAP challenges.
- Header Compression: *Default = Off*
  Enables the negotiation and use of IP Header Compression as per RFC2507, RFC2508 and RFC2509.
- PPP Compression Mode: *Default = MPPC*
  This option is used to negotiate compression (or not) using CCP. If set to MPPC or StacLZS the system will try to negotiate this mode with the remote Control Unit. If set to Disable CCP is not negotiated.
  - *Disable*
    Do not use or attempt to use compression.
  - *StacLZS*
    Attempt to use and negotiate STAC compression (the standard, Mode 3)
  - *MPPC*
    Attempt to use and negotiate MPPC (Microsoft) compression. Useful for dialing into NT Servers.
- PPP Callback Mode: *Default = Disable*
  - *Disable*:
    Callback is not enabled
  - *LCP: (Link Control Protocol)*
    After authentication the incoming call is dropped and an outgoing call to the number configured in the Service will be made to reestablish the link.
  - *Callback CP: (Microsoft's Callback Control Protocol)*
    After acceptance from both ends the incoming call is dropped and an outgoing call to the number configured in the Service is made to reestablish the link.
  - *Extended CBCP: (Extended Callback Control Protocol)*
    Similar to Callback CP however the Microsoft application at the remote end will prompt for a telephone number. An outgoing call will then be made to that number to reestablish the link.
- *Data Pkt. Size: Default = 0, Range = 0 to 2048.*
  This is the number of data bytes contained in a Data Packet.
- BACP: *Default = Off*
  Allows negotiation of the BACP/BCP protocols. These are used to control the addition of additional B channels to simultaneously improve data throughput.
- Multilink: *Default = Off*
  When enabled the system attempts to negotiate the use of the Multilink protocol (MPPC) on the link(s) into this Service. Multilink must be enabled if the more than one channel is allowed to be Bundled/Multilinked to this RAS Service.

5.12 Incoming Call Route Settings

Incoming call routes are used to determine the destination of voice and data calls received by the system. On systems where a large number incoming call routes need to be setup for DID numbers, the MSN/DID Configuration 132 tool can be used. Select **Tools | MSN Configuration**.

- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.

Determining which incoming call route is used is based on the call matching a number of possible criteria. In order of highest priority first, these are:

1. The **Bearer Capability** indicated, if any, with the call. For example voice, data, video.
2. The **Line Group ID** of the trunk or trunk channel on which the call was received.
3. The **Incoming Number** received with the call.
4. The **Incoming Sub Address** received with the call.
5. The **Incoming CLI** of the caller.

Multiple Matches

If there is a match between more than one incoming call route entry, the one added to the configuration first is used.

Incoming Call Route Destinations

Each incoming route can include a fallback destination for when the primary destination is busy. It can also include a time profile which control when the primary destination is used. Outside the time profile calls are redirected to a night service destination.

- **IP Office 4.1+:** Multiple time profiles can be associated with an incoming call route. Each time profile used has its own destination and fallback destination specified.

Incoming Call Routing Examples

Example 1

For this example, the customer has subscribes to receive two 2-digit DID numbers. They want calls on one routed to a Sales hunt group and calls on the other to a Services hunt group. Other calls should use the normal default route to hunt group Main. The following incoming call routes were added to the configuration to achieve this:

Line Group	Incoming Number	Destination
0	77	Sales
0	88	Services
0	blank	Main

Note that the incoming numbers could have been entered as the full dialed number, for example 7325551177 and 7325551188 respectively. The result would still remain the same as incoming number matching is done from right-to-left.

Line Group	Incoming Number	Destination
0	7325551177	Sales
0	7325551188	Services
0	blank	Main

Example 2

In the example below the incoming number digits 77 are received. The incoming call route entries 677 and 77 have the same number of matching digit place and no non-matching places so both a potential matches. In this scenario the systemwill use the incoming call route with the Incoming Number specified for matching.

Line Group	Incoming Number	Destination
0	677	Support
0	77	Sales
0	7	Services

0	blank	Main

Example 3

In the following example, the 677 entry is used as the match for 77 as it has more matching digits than the 7 entry and no non-matching digits.

Line Group	Incoming Number	Destination
0	677	Support
0	7	Services
0	blank	Main

Example 4

In this example the digits 777 are received. The 677 entry had a non-matching digit, so it is not a match. The entry 7 is used as it has one matching digit and no non-matching digits.

Line Group	Incoming Number	Destination
0	677	Support
0	7	Services
0	blank	Main

Example 5

In this example the digits 77 are received. Both the additional incoming call routes are potential matches. In this case the route with the shorter Incoming Number specified for matching is used and the call is routed to *Services*.

Line Group	Incoming Number	Destination
0	98XXX	Support
0	8XXX	Services
0	blank	Main

Example 6

In this example two incoming call routes have been added, one for incoming number 6XXX and one for incoming number 8XXX. In this case, any three digit incoming numbers will potential match both routes. When this occurs, potential match that was added to the system configuration first is used. If 4 or more digits were received then an exact matching or non-matching would occur.

Line Group	Incoming Number	Destination
0	6XXX	Support
0	8XXX	Services
0	blank	Main

5.12.1 Standard

Incoming call routes are used to match call received with destinations. Routes can be based on the incoming line group, the type of call, incoming digits or the caller's ICLID. If a range of MSN/DID numbers has been issued, this form can be populated using the MSN Configuration tool (see MSN Configuration 132).

- **Default Blank Call Routes**
  By default the configuration contains two incoming calls routes; one set for **Any Voice** calls (including analog modem) and one for **Any Data** calls. While the destination of these default routes can be changed, it is strongly recommended that the default routes are not deleted.
  - Deleting the default call routes, may cause busy tone to be returned to any incoming external call that does not match any incoming call route.
  - Setting any route to a blank destination field, may cause the incoming number to be checked against system short codes for a match. This may lead to the call being rerouted off-switch.
- Calls received on IP, S_0 and QSIG trunks do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.
- If there is no matching incoming call route for a call, matching is attempted against system short codes and finally against voicemail nodes before the call is dropped.
- **SIP Calls**
  For SIP calls, the following fields are used for call matching:
  - **Line Group ID**
    This field is matched against the **Incoming Group** settings of the SIP URI (Line | SIP URI 233). This must be an exact match.
  - **Incoming Number**
    This field can be used to match the called details (TO) in the SIP header of incoming calls. It can contain a number, SIP URI or Tel URI. For SIP URI's the domain part of the URI is removed before matching by incoming call routing occurs. For example, for the SIP URI mysip@example.com , only the user part of the URI, ie. mysip, is used for matching.
    - IP Office 6.0: The Call Routing Method 233 setting of the SIP line can be used to select whether the value used for incoming number matching is taken from the *To Header* or the *Request URI* information provided with incoming calls on that line.
  - **Incoming CLI**
    This field can be used to match the calling details (FROM) in the SDP header of incoming SIP calls. It can contain a number, SIP URI, Tel URI or IP address received with SIP calls. For all types of incoming CLI except IP addresses a partial entry can be used to achieve the match, entries being read from left to right. For IP addresses only full entry matching is supported.

Incoming Call Route | Standard	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

Incoming Call Matching Fields

The following fields are used to determine if the Incoming Call Route is a potential match for the incoming call. By default the fields are used for matching in the order shown starting with Bearer Capability.

- Bearer Capability: *Default = Any Voice*
  The type of call selected from the list of standard bearer capabilities: *Any*, *Any Voice*, *Any Data*, *Speech*, *Audio 3K1*, *Data 56K*, *Data 64K*, *Data V110*, *Video*.
- Line Group ID: *Default = 0, Range = 0 to 99999.*
  Matches against the Incoming Line Group to which the trunk receiving the call belongs.
- Incoming Number: *Default = Blank (Match any unspecified)*
  Matches to the digits presented by the line provider. A blank entry matches all calls that do not match other entries. By default this is a right-to-left matching.
  - * = Incoming CLI Matching Takes Precedence
  - – = Left-to-Right Exact Length Matching
    Using a - in front of the number causes a left-to-right match. When left-to-right matching is used, the number match must be the same length. For example -96XXX will match a DID of 96000 but not 9600 or 960000.
  - X = Single Digit Wildcard
    Use *X*'s to enter a single digit wild card character. For example *91XXXXXXXX* will only match DID numbers of at least 10 digits and starting with 91, *-91XXXXXXXX* would only match numbers of exactly 10 digits starting with 91. Other wildcard such as *N*, *n* and *?* cannot be used.
    - Where the incoming number potentially matches two incoming call routes with X wildcards and the number of incoming number digits is shorter than the number of wildcards, the one with the shorter overall Incoming Number specified for matching is used.
  - *i = ISDN Calling Party Number 'National'*
    The *i* character does not affect the incoming number matching. It is used for Outgoing Caller ID Matching, see notes below.
- Incoming Sub Address: *Default = Blank (Match all)*
  Matches any sub address component sent with the incoming call. If this field is left blank, it matches all calls.
- Incoming CLI: *Default = Blank (Match all)*
  Enter a number to match the caller's ICLID provided with the call. This field is matched left-to-right. Number options are:
  - Full telephone number.
  - Partial telephone number, for example just the area code.
  - ! : Matches calls where the ICLID was withheld.
  - ? : for number unavailable.
  - Blank for all.

Call Setting Fields

For calls routed using this Incoming Call Route, the settings of the following fields are applied to the call regardless of the destination.

- Locale: *Default = Blank (Use system setting)*
  This option specifies the language prompts, if available, that voicemail should use for the call if it is directed to voicemail.

  - When the system routes a call to the voicemail server it indicates the locale for which matching prompts should be provided if available. The locale sent to the voicemail server by the system is determined as follows:

Locale Source	Usage
Short Code	The short code locale, if set, is used if the call is routed to voicemail using the short code.
System	If no user or incoming call route locale is set system locale is used unless overridden by a short code locale.
Incoming Call Route	The incoming call route locale, if set, is used if caller is external.
User	The user locale, if set, is used if the caller is internal.

- Priority: *Default = 1-Low, Range = 1-Low to 3-High.*
  This setting allows incoming calls to be assigned a priority. Other calls such as internal calls are assigned priority *1-Low*.

  - In situations where calls are queued, high priority calls are placed before calls of a lower priority. This has a number of effects:

    - Mixing calls of different priority is not recommended for destinations where Voicemail Pro is being used to provided queue ETA and queue position messages to callers since those values will no longer be accurate when a higher priority call is placed into the queue. Note also that Voicemail Pro will not allow a value already announced to an existing caller to increase.

    - If the addition of a higher priority call causes the queue length to exceed the hunt group's Queue Length Limit 316, the limit is temporarily raised by 1. This means that calls already queued are not rerouted by the addition of a higher priority call into the queue.

  - IP Office 4.2+: A timer can be used to increase the priority of queued calls, see System | Telephony | Telephony | Call Priority Promotion Time 160.

  - IP Office 4.2+: The current priority of a call can be changed through the use of the p short code character 427 in a short code used to transfer the call.

- Tag: *Default = Blank (No tag). Software level = 4.1+.*
  Allows a text tag to be associated with calls routed by this incoming call route. This tag is displayed with the call within applications and on phone displays. See Call Tagging 756.

- Hold Music Source: *Default = System source. Software level = 4.2+ (Not Small Office Edition).*
  The system can support up to 4 music on hold source; the *System Source* (either an internal file or the external source port or tones) plus up to 3 additional internal wav files, see System | Telephony | Tones & Music 163. If the system has several hold music sources available, this field allows selection of the source to associate with calls routed by this incoming call route. The new source selection will then apply even if the call is forwarded or transferred away from the Incoming Call Route destination. If the call is routed to another IP Office 4.2+ system in a Small Community Network, the matching source is used if available. The hold music source associated with a call can also be changed by a hunt group's Hold Music Source 306 setting.

- Destination: *Default = Blank, Software Level = Up to 4.0 only.*
  For IP Office 4.1+, this option has moved to the Incoming Call Route | Destinations 349 tab. Select the destination for the call from the drop-down list box which contains all available extensions, users, groups, RAS services and voicemail. System short codes and dialing numbers can be entered manually. Once the incoming call is matched the call is passed to that destination.

  - Drop-Down List Options
    The following options appear in the drop-down in the following order:

    - Voicemail allows remote mailbox access with Embedded Voicemail, Voicemail Lite or Voicemail Pro. Callers are asked to enter the extension ID of the mailbox required and then the mailbox access code.

    - User Names

    - Hunt Groups Names

    - AA:Name directs calls to an Embedded Voicemail auto-attendant services.

  - Manually Entered Options
    The following options can be entered manually into the field.

    - VM:Name Directs calls to the matching start point in Voicemail Pro.

- A . matches the Incoming Number field. This can be used even when X wildcards are being used in the Incoming Number field.
- A # matches all X wildcards in the Incoming Number field. For example, if the Incoming Number was - 91XXXXXXXXXXX, the Destination would be XXXXXXXXXXX.
- Text and number strings entered here are passed through to system short codes, for example to direct calls into a conference. Note that not all short code features are supported.

- Fallback Extension: *Default = Blank (No fallback). Software level = Up to 4.0 only.*
  For IP Office 4.1+, this option has moved to the Incoming Call Route | Destinations 349 tab. Defines an alternate destination which should be used when the current destination, set in the Destination or Night Service Destination field, cannot be obtained. For example if the primary destination is a hunt group returning busy and without queuing or voicemail.
- Night Service Profile: *Default = <None> (No night service). Software level = Up to 4.0 only.*
  A time profile during which the Night Service Destination should be used rather than the Destination.
- Night Service Destination: *Default = Blank. Software level = Up to 4.0 only.*
  Set the destination to be used during periods defined by the Night Service Profile. The same range of values can be used as for the Destination field.

Outgoing Caller ID Matching

In cases where a particular Incoming Number is routed to a specific individual user, the system will attempt to use that Incoming Number as the user's caller ID when they make outgoing calls if no other number is specified. This requires that the Incoming Number is a full number suitable for user as outgoing caller ID and acceptable to the line provider.

When this is the case, the character *i* can also be added to the Incoming Number field. This character does not affect the incoming call routing. However when the same Incoming Number is used for an outgoing caller ID, the calling party number plan is set to ISDN and the type is set to National. This option may be required by some network providers.

5.12.2 Voice Recording

This tab is used to activate the automatic recording of incoming calls that match the incoming call route.

Call recording requires Voicemail Pro to be installed and running. Call recording also requires available conference resources similar to a 3-way conference.

- IP Office 4.0+ introduces the following changes to recording:
  - Calls to and from IP devices, including those using Direct media, can be recorded.
  - Calls parked or held pause recording until the unparked or taken off hold.
  - Recording is stopped if:
    - User recording stops if the call is transferred to another user.
    - User account code recording stops if the call is transferred to another user.
    - Hunt group recording stops if the call is transferred to another user who is not a member of the hunt group.
    - Incoming call route recording continues for the duration of the call on the system.

Incoming Call Route | Voice Recording	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	✓.

- **Record Outbound:** *Default = None*
  Select whether automatic recording of outgoing calls is enabled. The **Auto Record Calls** option sets whether just external calls or external and internal calls are included. Options for recording are:
  - **None:** Do not automatically record calls.
  - **On:** Record the call if possible. If not possible to record, allow the call to continue.
  - **Mandatory:** Record the call if possible. If not possible to record, block the call and return busy tone.
  - **Percentages of calls:** Record a selected percentages of the calls.
- **Recording (Auto):** *Default = Mailbox*
  Sets the destination for automatically triggered recordings.
  - **Mailbox**
    This option sets the destination for the recording to be a selected user or hunt group mailbox. The adjacent drop down list is used to select the mailbox.
  - **Voice Recording Library:** *Software level = 3.0+.*
    This options set the destination for the recording to be a VRL folder on the voicemail server. The ContactStore application polls that folder and collects waiting recordings which it then places in its own archive. Recording is still done by the Voicemail Pro.

5.12.3 Destinations

IP Office 4.1+: The system allows multiple time profiles to be associated with an incoming call route. For each time profile, a separate Destination and Fallback Extension can be specified.

When multiple entries are added, they are resolved from the bottom up. The entry used will be the first one, working from the bottom of the list upwards, that is currently 'true', ie. the current day and time or date and time match those specified by the Time Profile. If no match occurs the Default Value options are used.

Once a match is found, the system does not use any other destination set even if the intended Destination and Fallback Extension destinations are busy or not available.

Incoming Call Route | Standard	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.1+.
Mergeable	✓.

- Time Profile
  This column is used to specify the time profiles 360 used by the incoming call routes. The entry displays a drop-down list of existing time profiles from which a selection can be made. To remove an existing entry, select it by clicking on the button on the left of the row, then right-click on the row and select Delete.
  - Default Value
    This entry is fixed and is used if no match to a time profile below occurs.
- Destination: *Default = Blank*
  Either enter the destination manually or select the destination for the call from the drop-down list. The dr box which contains all available extensions, users, groups, RAS services and voicemail. System short codes and dialing numbers can be entered manually. Once the incoming call is matched the call is passed to that destination.
  - Drop-Down List Options
    The following options appear in the drop-down list:
    - Voicemail allows remote mailbox access with Embedded Voicemail, Voicemail Lite or Voicemail Pro. Callers are asked to enter the extension ID of the mailbox required and then the mailbox access code.
    - Local user names.
    - Local hunt groups names.
    - AA:*Name* directs calls to an Embedded Voicemail auto-attendant services.
  - Manually Entered Options
    In addition to short codes, extension and external numbers, the following options can be also be entered manually:
    - *VM:Name* Directs calls to the matching start point in Voicemail Pro.
    - A . matches the Incoming Number field. This can be used even when X wildcards are being used in the Incoming Number field.
    - A # matches all X wildcards in the Incoming Number field. For example, if the Incoming Number was - 91XXXXXXXXXXX, the Destination of "#" would match XXXXXXXXXXX. Note that the # should be enclosed in " quotation marks.
    - Text and number strings entered here are passed through to system short codes, for example to direct calls into a conference. Note that not all short code features are supported.
- Fallback Extension: *Default = Blank (No fallback)*
  Defines an alternate destination which should be used when the current destination, set in the Destination field cannot be obtained. For example if the primary destination is a hunt group returning busy and without queuing or voicemail.

5.13 WAN Port Settings

These entries are used to configure the operation of system WAN ports and services.

Physical WAN ports are the 37-way D-type WAN ports found on the rear of IP403, IP406, IP406v2 and IP412 control unit. Additional WAN ports can be added by the installation of up to two WAN3 expansion modules, each module providing 3 additional WAN ports. On the Small Office Edition control unit, a single WAN port can be added by the installation of a WAN trunk card at the rear of the unit. Physical WAN ports are not supported with IP500 and IP500v2 systems. For full details of installing additional WAN ports, refer to the Manager Installation Manual.

Creating a Virtual WAN Port

WAN services can be run over a T1 PRI trunk connection. This requires creation of a virtual WAN port. For full details refer to Using a Dedicated T1/PRI ISP Link 801 in Appendix A.

1. Select WAN Port.

2. Click and select PPP.

3. In the **Name** field, enter either *LINEx.y* where:

   - *LINE* must be in uppercase.

   - *x* is the line number. For a PRI/T1 module in Slot A, this will be 1. For a PRI/T1 module in Slot B, this will be 5.

   - *y* is the lowest numbered channel number to be used by the WAN link minus 1. For example, if the lowest channel to be used is channel 1 then y = 1 - 1 = 0.

4. In the **Speed** field, enter the total combined speed of the maximum number of channels sets in the Service. In this example, 12 channels x 64000 bits = 76800.

   - Note: The maximum number of channels that can be used will be limited by the number of data channels supported by the system Control Unit and not already in use.

5. In the **RAS Name** field, select the RAS name created when the new Service of that name was created.

6. Click OK.

5.13.1 WAN Port

Use this form to configure the leased line connected to the WAN port on the Control Unit. Normally this connection is automatically detected by the control unit. If a WAN Port is not displayed, connect the WAN cable, reboot the Control Unit and receive the configuration. The WAN Port configuration form should now be added.

WAN Port | WAN Port	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- **Name**
  The physical ID of the Extension port,. This parameter is not configurable; it is allocated by the system.
- **Speed**
  The operational speed of this port. For example for a 128K connection, enter 128000. This should be set to the actual speed of the leased line as this value is used in the calculation of bandwidth utilization. If set incorrectly, additional calls may be made to increase Bandwidth erroneously.
- **Mode**: *Default = SyncPPP*
  Select the protocol required:
  - *SyncPPP*
    For a data link.
  - *SyncFrameRelay*
    For a link supporting Frame Relay.
- **RAS Name**
  If the Mode is *SyncPPP*, selects the RAS service to associate with the port. If the Mode is *SyncFrameRelay*, the RAS Name is set through the DCLIs tab.

5.13.2 Frame Relay

This tab is only available for Frame Relay entries. These show *SyncFrameRelay* as the Mode on the WAN Port tab.

WAN Port | Frame Relay	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Frame Management Type
  This must match the management type expected by the network provider. Selecting *AutoLearn* allows the system to automatically determine the management type based on the first few management frames received. If a fixed option is required the following are supported: *Q933 AnnexA 0393*, *Ansi AnnexD* , *FRFLMI* and *None*.
- Frame Learn Mode
  This parameter allows the DLCIs that exist on the given WAN port to be provisioned in a number of different ways.
  - *None*
    No automatic learning of DLCIs. DLCIs must be entered and configured manually.
  - *Mgmt*
    Use LMI to learn what DLCIs are available on this WAN.
  - *Network*
    Listen for DLCIs arriving at the network. This presumes that a network provider will only send DLCIs that are configured for this particular WAN port.
  - *NetworkMgmt*
    Do both management and network listening to perform DLCI learning and creation.
- Max Frame Length
  Maximum frame size that is allowed to traverse the frame relay network.
- Fragmentation Method
  Options are *RFC1490* or *RFC1490+FRF12*.

5.13.3 DLCIs

This tab is only available for Frame Relay entries. These show *SyncFrameRelay* as the Mode on the WAN Port 351 tab.

The tab lists the DLCIs created for the connection. These can be edited using the Add, Edit and Remove buttons.

WAN Port | DLCIs	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Frame Link Type: *Default = PPP*
  Data transfer encapsulation method. Set to the same value at both ends of the PVC (Permanent Virtual Channel).
  - *None*
  - *PPP*
    Using PPP offers features such as out of sequence traffic reception, compression and link level connection management.
  - *RFC 1490*
    RFC 1490 encapsulation offers performance and ease of configuration and more inter-working with third party CPE.
  - *RFC1490 + FRF12*
    Alternate encapsulation to PPP for VoIP over Frame Relay. When selected all parameters on the Service | PPP tab being used are overridden.
- DLCI: *Default = 100*
  This is the Data Link Connection Identifier, a unique number assigned to a PVC end point that has local significance only. Identifies a particular PVC endpoint within a user's physical access channel in a frame relay.
- RAS Name
  Select the RAS Service you wish to use.
- Tc: *Default = 10*
  This is the Time Constant in milliseconds. This is used for measurement of data traffic rates. The Tc used by the system can be shorter than that used by the network provider.
- CIR: *(Committed Information Rate) Default = 64000 bps*
  This is the Committed Information Rate setting. It is the maximum data rate that the WAN network provider has agreed to transfer. The committed burst size (Bc) can be calculated from the set Tc and CIR as Bc = CIR x Tc. For links carrying VoIP traffic, the Bc should be sufficient to carry a full VoIP packet including all its required headers. See the example below.
- EIR: *(Excess Information Rate) Default = 0 bps*
  This is the maximum amount of data in excess of the CIR that a frame relay network may attempt to transfer during the given time interval. This traffic is normally marked as De (discard eligible). Delivery of De packets depends on the network provider and is not guaranteed and therefore they are not suitable for UDP and VoIP traffic. The excess burst size (Be) can be calculated as Be = EIR x Tc.

Example: Adjusting the Tc Setting

G.729 VoIP creates a 20 byte packet every 20ms. Adding typical WAN PPP headers results in a 33 byte packet every 20ms.

For a Committed Information Rate (CIR) of 14Kbps, with the Time Constant (Tc) set to 10ms; we can calculate the Committed Burst size:

Bc = CIR x Tc = 14,000 x 0.01 = 140 bits = 17.5 bytes.

Using 10ms as the Tc, a full G.729 VoIP packet (33 bytes) cannot be sent without exceeding the Bc. The most likely result is lost packets and jitter.

If the Tc is increased to 20ms:

Bc = CIR x Tc = 14,000 x 0.02 = 280 bits = 35 bytes.

The Bc is now sufficient to carry a full G.729 VoIP packet.

Notes

1. Backup over Frame Relay is not supported when the Frame Link Type is set to RFC1490.
2. When multiple DLCIs are configured, the WAN link LED is switched off if any of those DLCIs is made inactive, regardless of the state of the other DLCIs. Note also that the WAN link LED is switched on following a reboot even if one of the DLCIs is inactive. Therefore when multiple DLCIs are used, the WAN link LED cannot be used to determine the current state of all DLCIs.
3. When the Frame Link Type is set to RFC1490, the WAN link LED is switched on when the WAN cable is attached regardless other whether being connected to a frame relay network.

5.13.4 Advanced

The settings on this tab are used for Frame Relay connections.

WAN Port | Advanced	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Address Length
  The address length used by the frame relay network. The network provider will indicate if lengths other than two bytes are to be used.
- N391: *Full Status Polling Counter*
  Polling cycles count used by the CPE and the network provider equipment when bidirectional procedures are in operation. This is a count of the number of link integrity verification polls (T391) that are performed (that is Status Inquiry messages) prior to a Full Status Inquiry message being issued.
- N392: *Error Threshold Counter*
  Error counter used by both the CPE and network provider equipment. This value is incremented for every LMI error that occurs on the given WAN interface. The DLCIs attached to the given WAN interface are disabled if the number of LMI errors exceeds this value when N393 events have occurred. If the given WAN interface is in an error condition then that error condition is cleared when N392 consecutive clear events occur.
- N393: *Monitored Events Counter*
  Events counter measure used by both the CPE and network provider equipment. This counter is used to count the total number of management events that have occurred in order to measure error thresholds and clearing thresholds.
- T391: *Link Integrity Verification Polling Timer*
  The link integrity verification polling timer normally applies to the user equipment and to the network equipment when bidirectional procedures are in operation. It is the time between transmissions of Status Inquiry messages.
- T392: *Polling Verification Timer*
  The polling verification timer only applies to the user equipment when bidirectional procedures are in operation. It is the timeout value within which to receive a Status Inquiry message from the network in response to transmitting a Status message. If the timeout lapses an error is recorded (N392 incremented).

5.13.5 PPP

Fields in this tab enable you to configure Point to Point Protocol (PPP) in relation to this particular service. PPP is a protocol for communication between two computers using a Serial interface.

Service | PPP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 , IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Chap Challenge Interval (secs): *Default = 0 (disabled). Range = 0 to 99999 seconds.*
  The period between CHAP challenges. Blank or 0 disables repeated challenges. Some software such as Windows 95 DUN does not support repeated CHAP challenges.
- Bi-Directional Chap: *Default =Off.*
- Header Compression: *Default = None selected*
  Enables the negotiation and use of IP Header Compression. Supported modes are IPHC and VJ. IPHC should be used on WAN links.
- PPP Compression Mode: *Default = MPPC*
  Enables the negotiate and use of compression. Do not use on VoIP WAN links.
  - *Disable*
    Do not use or attempt to use compression.
  - *StacLZS*
    Attempt to use STAC compression (Mode 3, sequence check mode).
  - *MPPC*
    Attempt to use MPPC compression. Useful for NT Servers.
- PPP Callback Mode: *Default = Disable*
  - *Disable*
    Callback is not enabled
  - *LCP: (Link Control Protocol)*
    After authentication the incoming call is dropped and an outgoing call to the number configured in the Service is made to re-establish the link.
  - *Callback CP: (Microsoft's Callback Control Protocol)*
    After acceptance from both ends the incoming call is dropped and an outgoing call to the number configured in the Service is made to re-establish the link.
  - *Extended CBCP: (Extended Callback Control Protocol)*
    Similar to Callback CP except the Microsoft application at the remote end prompts for a telephone number. An outgoing call is then made to that number to re-establish the link.
- PPP Access Mode: *Default = Digital64*
  Sets the protocol, line speed and connection request type used when making outgoing calls. Incoming calls are automatically handled (see RAS services).
  - *Digital64*
    Protocol set to Sync PPP, rate 64000 bps, call presented to local exchange as a "Data Call".
  - *Digital56*
    As above but rate 56000 bps.
  - *Voice56*
    As above but call is presented to local exchange as a "Voice Call".
  - *V120*
    Protocol set to Async PPP, rate V.120, call presented to local exchange as a "Data Call". This mode runs at up to 64K per channel but has a higher Protocol overhead than pure 64K operation. Used for some bulletin board systems as it allows the destination end to run at a different asynchronous speed to the calling end.
  - *V110*
    Protocol is set to Async PPP, rate V.110. This runs at 9600 bps, call is presented to local exchange as a "Data Call". It is ideal for some bulletin boards.
  - *Modem*
    Allows Asynchronous PPP to run over an auto-adapting Modem to a service provider (requires a Modem2 card in the main unit)

- Data Pkt. Size: *Default = 0, Range = 0 to 2048.*
  Sets the size limit for the Maximum Transmissible Unit.

- BACP: *Default = Off*
  Enables the negotiation and use of BACP/BCP protocols. These are used to control the addition of B channels to increase bandwidth.

- Incoming traffic does not keep link up: *Default = On*
  When enabled, the link is not kept up for incoming traffic only.

- Multilink/QoS: *Default = Off*
  Enables the negotiation and use of Multilink protocol (MPPC) on links into this Service. Multilink must be enabled if there is more than one channel that is allowed to be Bundled/Multilinked to this RAS Service.

5.14 Directory

This section is used to edit directory records that are stored in the system's configuration. These directory records can be manually imported or export using a CSV file 61. The system can also use Directory Services 166 to automatically import directory entries from an LDAP server at regular intervals.

IP Office 5.0+: The systemcan also automatically import directory entries from another system. Automatically imported entries are used as part of the system directory but are not part of the editable configuration. Automatically imported entries cannot override manually entered entries.

Directory entries are used for two types of function:

- Directory Dialing
  Directory numbers are displayed by user applications such as Phone Manager and SoftConsole. Directory numbers are viewable through the Dir 615 function on many Avaya phones (Contacts or History). They allow the user to select the number to dial by name. The directory will also contain the names and numbers of users and hunt groups on the system.
  - The Dir function groups directory entries shown to the phone user into the following categories. Depending on the phone, the user may be able to select the category currently displayed. In some scenarios, the categories displayed may be limited to those supported for the function being performed by the user:
    - External
      Directory entries from the system configuration. IP Office 5.0+: This includes HTTP and LDAP imported entries.
    - Groups
      Groups on the system. If the system is in a Small Community Network it will also include groups on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - Users or Index
      Users on the system. If the system is in a Small Community Network it will also include users on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - Personal
      Available for T3 phones, T3 IP phones, 1400, 1600 and 9600 Series phones. These are the user's personal directory entries stored within the system configuration.
- Name Matching
  Directory entries are also used to associate a name with the dialled number on outgoing calls or the received CLI on incoming calls. When name matching is being done, a match in the users personal directory overrides any match in the system directory. Note that some user applications also have their own user directory.
  - The IP Office Phone Manager and SoftConsole applications have their own user directories which are also used by the applications name matching. Matches in the application directory may lead to the application displaying a different name from that shown on the phone.
  - Name matching is not performed when a name is supplied with the incoming call, for example QSIG trunks.
  - Directory name matching is not supported for DECT handsets.

A maximum of 2500 directory entries (1000 in pre-IP Office 5 systems) are supported in the system configuration. Using a 1400, 1600 or 9600 Series phone, system phone users can also edit the configuration directory records.

System	Number of Directory Records			Total Number of Directory Records
	Configuration	LDAP Import	HTTP Import	
IP500 / IP500v2	2500	5000	5000	5000
IP412	2500	2500	2500	2500
IP406 V2	2500	2500	2500	2500

Directory | Directory Entry	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- **Name**
  Enter the text, to be used to identify the number. Names should not begin with numbers.

- **Number**
  Enter the number to be matched with the above name. Any brackets or - characters used in the number string are ignored. The directory number match is done on reading from the left-hand side of the number string. Note that if the system has been configured to use an external dialing prefix, that prefix should be added to directory numbers.

The following characters *(IP Office Release 5.0+)* are supported in directory entries. They are supported in both system configuration entries and in HTTP/LDAP imported entries.

- **? = Any Digit**
  Directory entries containing a ? are only used for name matching against the dialed or received digits on outgoing or incoming. They are not included in the directory of numbers to dial available to users through their phones or applications. The wildcard can be used in any position but typically would be used at the end of the number.
  - In the following example, any calls where the dialed or received number is 10 digits long and starts 732555 will have the display name Homdel associated with them.
    - **Name**: Holmdel
    - **Number**: 9732555????
- **(and) brackets = Optional Digits**
  These brackets are frequently used to enclose an optional portion of a number, typically the area code. Only one pair of brackets are supported in a number. Entries containing digits inside ( ) brackets are used for both name matching or user dialling. When used for name matching, the dialed or received digits are compared to the directory number with and without the ( ) enclosed digits. When used for dialling from a phone or application directory, the full string is dialed with the ( ) brackets removed.
  - The following example is a local number. When dialed by users they are likely to dial just the local number. However on incoming calls, for the CLI the telephony provider includes the full area code. Using the ( ) to enclose the area code digits, it is possible for the single directory entry to be used for both incoming and outgoing calls.
    - **Name**: Raj Garden
    - **Number**: 9(01707)373386
- **Space and - Characters**
  Directory entries can also contain spaces and - characters. These will be ignored during name matching and dialing from the directory.

5.15 Time Profile

Time Profiles are used by different services to change their operation when required. In most areas where time profiles can be used, not setting a time profile is taken as meaning 24-hour operation.

- Time profiles consist of recurring weekly patterns of days and times when the time profile is in effect.
- Time profiles *(IP Office 4.1+)* can include time periods on specified calendar days when the time profile is in effect. Calendar entries can be entered for the current and following calendar year.

Time profiles are used by the following entry types:

Hunt Group 304
A hunt group can use time profiles in the following ways:

- A time profile can be used to determine when a hunt group is put into night service mode 314. Calls then go to an alternate Night Service Fallback group if set, otherwise to voicemail if available or busy tone if not.
- For automatic voice recording 319, a time profile can be used to set when voice recording is used.

Service 331
A Service can use time profiles in the following ways:

- A time profile can be used to set when a data service is available. Outside its time profile, the service is either not available or uses an alternate fallback service if set.
- For services using auto connect, a time profile can be used to set when that function is used. See Service | Autoconnect 336.

User 262
A user can use time profiles in the following ways:

- Users being used for Dial In 283 data services such as RAS can have an associated time profile that defines when they can be used for that service.
- Users can be associated with a working hours and an out of hours user rights. A time profile can then be used to determine which user rights is used at any moment.
- For automatic voice recording 284, a time profile can be used to set when that voice recording is used.
- For mobile twinning 296, a time profile can be used to define when twinning should be used.

Incoming Call Route 342
Incoming call routes can also use time profiles to specify when calls should be recorded. Multiple time profiles can be associate with an incoming call route, each profile specifying a destination and fall back destination.

Least Cost Route
A Least Cost Route uses time profiles to determine when the routes should be used.

ARS 407
ARS forms use time profile to determine when the ARS form should be used or calls rerouted to an out of hours route.

Account Code 375
Account Codes can use automatic voice recording triggered by calls with particular account codes. A time profile can be used to set when this function is used.

Auto Attendant 400
Embedded voicemail auto attendants can use time profiles to control the different greetings played to callers.

For a time profile with multiple entries, for example a week pattern and some calendar entries, the profile is valid when any entry is valid.

Time Profile | Time Profile	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✓.

- Name: *Range = Up to 31 characters*
  This name is used to select the time profile from within other tabs.

- Time Entry List
  This list shows the current periods during which the time profile is active. Clicking on an existing entry will display the existing settings and allows them to be edited if required. To remove an entry, selecting it and then click on Remove or right-click and select Delete.

- Recurrence Pattern (Weekly Time Pattern)
  When a new time entry is required, click Add Recurring and then enter the settings for the entry using the fields displayed. Alternately right-click and select Add Recurring Time Entry. This type of entry specific a time period and the days on which it occurs, for example 9:00 - 12:00, Monday to Friday. A time entry cannot span over two days. For example you cannot have a time profile starting at 18:00 and ending 8:00. If this time period is required two Time Entries should be created - one starting at 18:00 and ending 11:59, the other starting at 00:00 and ending 8:00.

  - Start Time
    The time at which the time period starts.

  - End Time
    The time at which the time period ends.

  - Days of Week
    The days of the week to which the time period applies.

- Recurrence Pattern (Calendar Date): *Software Level = 4.1+.*
  When a new calendar date entry is required, click Add Date and then enter the settings required. Alternately right-click and select Add Calendar Time Entry. Calendar entries can be set for up to the end of the next calendar year.

  - Start Time
    The time at which the time period starts.

  - End Time
    The time at which the time period ends.

  - Year
    Select either the current year or the next calendar year.

  - Date
    To select or de-select a particular day, double-click on the date. Selected days are shown with a dark gray background. Click and drag the cursor to select or de-select a range of days.

5.16 Firewall Profile Settings

The system can act as a firewall, allowing only specific types of data traffic to start a session across the firewall and controlling in which direction such sessions can be started.

- Static NAT 366 *(IP Office 4.2+)*
  The system supports Static NAT address translation by a firewall profiles. If the Firewall Profile contains any Static NAT entries, all packets received by the firewall must match one of those static NAT entries to not be blocked.

System firewall profiles can be applied in the following areas of operation:

- System 155
  A firewall profile can be selected to be applied to traffic between LAN1 and LAN2.

- User 283
  Users can be used as the destination of incoming RAS calls. For those users a firewall profile can be selected on the user's Dial In 283 tab.

- Service 332
  Services are used as the destination for IP routes connection to off-switch data services such as the internet. A Firewall Profile can be selected for use with a service.

- Logical LAN 385
  Where a logical LAN is created for use as an IP Route destination, a Firewall Profile can be selected for use with the logical LAN.

If Network Address Translation (NAT) is used with the firewall (which it typically is), then you must also configure a Primary Incoming Translation Address (see IP 335 tab of the Service configuration form) if you wish sessions to be started into your site (typically for SMTP) from the Internet.

5.16.1 Standard

By default, any protocol not listed in the standard firewall list is dropped unless a custom firewall entry 364 is configured for that protocol.

Firewall | Standard	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Name: *Range = Up to 31 characters*
  Enter the name to identify this profile.
- Protocol Control
  For each of the listed protocols, the options *Drop*, *In* (Incoming traffic can start a session), *Out* (Outgoing traffic can start a session) and *Both Directions* can be selected. Once a session is started, return traffic for that session is also able to cross the firewall.

Protocol	Default	Description
TELNET	Out	Remote terminal log in.
FTP	Out	File Transfer Protocol.
SMTP	Out	Simple Mail Transfer Protocol.
TIME	Out	Time update protocol.
DNS	Out	Domain Name System.
GOPHER	Drop	Internet menu system.
FINGER	Drop	Remote user information protocol.
RSVP	Drop	Resource Reservation Protocol.
HTTP	Out	Hypertext Transfer Protocol.
POP3	Out	Post Office Protocol.
NNTP	Out	Network News Transfer Protocol.
SNMP	Drop	Simple Network Management Protocol.
IRC	Out	Internet Relay Chat.
PPTP	Drop	Point to Point Tunneling Protocol.
IGMP	Drop	Internet Group Membership Protocol.

- Service Control: *Software level = 4.0+.*
  For each of the listed services, the options *Drop*, *In*, *Out* and *Both Directions* can be selected. Once a session is started, return traffic for that session is also able to cross the firewall.

Protocol	Default	Description
SSI	In	System Status Application access.
SEC	Drop	TCP security settings access.
CFG	Drop	TCP configuration settings access.
TSPI	In	TSPI service access.

5.16.2 Custom

The tab lists custom firewall settings added to the firewall profile. The Add, Edit and Remove controls can be used to amend the settings in the list.

Firewall | Custom	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- Notes
  For information only. Enter text to remind you of the purpose of the custom firewall entry.
- Remote IP Address
  The IP address of the system at the far end of the link. Blank allows all IP addresses.
- Remote IP Mask
  The mask to use when checking the Remote IP Address. When left blank no mask is set, equivalent to 255.255.255.255 - allow all.
- Local IP Address
  The address of devices local to this network (pre-translated). Blank allows all IP addresses.
- Local IP Mask
  The mask to use when checking the Local IP Address. When left blank no mask is set, equivalent to 255.255.255.255 - allow all.
- IP Protocol
  The value entered here corresponds to the IP Protocol which is to be processed by this Firewall profile: 1 for ICMP, 6 for TCP, 17 for UDP or 47 for GRE. This information can be obtained from the "pcol" parameter in a Monitor trace.
- Match Offset
  The offset into the packet (0 = first byte of IP packet) where checking commences for either a specific port number, a range of port numbers, or data.
- Match Length
  The number of bytes to check in the packet, from the Match Offset point, that are checked against the Match Data and Match Mask settings.
- Match Data
  The values the data must equal once masked with the Match Mask. This information can be obtained from "TCP Dst" parameter in a Monitor trace (the firewall uses hex so a port number of 80 is 50 in hex)
- Match Mask
  This is the byte pattern, which is logically ANDed with the data in the packet from the offset point. The result of this process is then compared against the contents of the "Match Data" field.
- Direction
  The direction that data may take if matching this filter.

Drop	All matching traffic is dropped.
In	Incoming traffic can start a session.
Out	Outgoing traffic can start a session.
Both Directions	Both incoming and outgoing traffic can start sessions.

Example Custom Firewall Entries

Example: Dropping NetBIOS searches on an ISPs DNS

We suggest that the following filter is always added to the firewall facing the Internet to avoid costly but otherwise typically pointless requests from Windows machines making DNS searches on the DNS server at your ISP.

- Direction: Drop
- IP Protocol: 6 (TCP)
- Match Offset: 20
- Match Length: 4
- Match Data: 00890035
- Match Mask: FFFFFFFF

Example: Browsing Non-Standard Port Numbers

The radio button for HTTP permits ports 80 and 443 through the firewall. Some hosts use non-standard ports for HTTP traffic, for example 8080, 8000, 8001, 8002, etc. You can add individual filters for these ports as you find them.

You wish to access a web page but you cannot because it uses TCP port 8000 instead of the more usual port 80, use the entry below.

- Direction: Out
- IP Protocol: 6 (TCP)
- Match Offset: 22
- Match Length: 2
- Match Data: 1F40
- Match Mask: FFFF

A more general additional entry given below allows all TCP ports out.

- Direction: Out
- IP Protocol: 6 (TCP)
- Match Offset: 0
- Match Length: 0
- Match Data: 00000000000000000000000000000000
- Match Mask: 00000000000000000000000000000000

Example: Routing All Internet Traffic through a WinProxy

If you wish to put WinProxy in front of all Internet traffic via the Control Unit. The following firewall allows only the WinProxy server to contact the Internet : -

1. Create a new Firewall profile and select *Drop* for all protocols
2. Under Custom create a new Firewall Entry
3. In Notes enter the name of the server allowed. Then use the default settings except in Local IP Address enter the IP address of the WinProxy Server, in Local IP Mask enter 255.255.255.255 and in Direction select Both Directions.

Stopping PINGs

You wish to stop pings - this is ICMP Filtering. Using the data below can create a firewall filter that performs the following; Trap Pings; Trap Ping Replies; Trap Both.

- Trap Pings: Protocol = 1, offset = 20, data = 08, mask = FF
- Trap Ping Replies: Protocol = 1, offset = 20, data = 00, mask = FF
- Trap Both: Protocol = 1, offset = 20, data = 00, mask = F7, Traps Both.

5.16.3 Static NAT

The Static NAT table allows the firewall to perform address translation between selected internal and external IP addresses. Up to 64 internal and external IP address pairs can be added to the Static NAT section of a Firewall Profile.

This feature is intended for incoming maintenance access using applications such as PC-Anywhere, Manager and the Voicemail Pro Client. The address translation is used for destinations such a Voicemail Pro server or the system's own LAN1 address.

- If there are any entries in the Static NAT settings of a Firewall Profile, each packet attempting to pass through the firewall must match one of the static NAT pairs or else the packet will be dropped.
- The destination address of incoming packets is checked for a matching External IP Address. If a match is found, the target destination address is changed to the corresponding Internal IP Address.
- The source address of outgoing packets is checked for a matching Internal IP Address. If a match is found, the source address is changed to the corresponding External IP Address.
- Even when a static NAT address match occurs, the other settings on the Firewall Profile Standard 363 and Custom 364 tabs are still applied and may block the packet.

5.17 IP Route Settings

The system acts as the default gateway for its DHCP clients. It can also be specified as the default gateway for devices with static IP addresses on the same subnet as the system. When devices on LAN1 and LAN2 want to send data to IP addresses on a different subnet, they will send that data to their default gateway for onward routing.

The IP Route table is used by the system to determine where data traffic should be forwarded. This is done by matching details of the destination IP address to IP Route entries and then using the Destination specified by the matching IP route. These are referred to as 'static routes'.

- Automatic Routing (RIP)
  The system can support RIP (Routing Information Protocol) on LAN1 and or LAN2. This is a method through which the system can automatically learn routes for data traffic from other routers that also support matching RIP options, see RIP 369. These are referred to as 'dynamic routes'.

- Dynamic versus Static Routes
  By default, static routes entered into the system override any dynamic routes it learns by the use of RIP. This behavior is controlled by the Favor RIP Routes over static routes option on the System | System 143 tab.

- Static IP Route Destinations
  The system allows the following to be used as the destinations for IP routes:

  - *LAN1* 148
    Direct the traffic to the system's LAN1.

  - *LAN2* 155
    On Small Office Edition, IP412, IP500 and IP500v2 systems, traffic can be directed to LAN2. For Manager 4.1+, LAN port 8 on IP406 V2 control units can be enabled as LAN2.

  - *Service* 331
    Traffic can be directed to a service. The service defines the details necessary to connect to a remote data service.

  - *Logical LAN* 385
    Traffic can be directed to a logical LAN already added to the configuration.

  - *Tunnel* 380
    Traffic can be directed to an IPSec or L2TP tunnel.

- Default Route
  The system provides two methods of defining a default route for IP traffic that does not match any other specified routes. Use either of the following methods:

  - *Default Service*
    Within the settings for services, one service can be set as the Default Route (Service | Service 332).

  - *Default IP Route*
    Create an IP Route entry with a blank IP Address and blank IP Mask set to the required destination for default traffic.

5.17.1 IP Route

This tab is used to setup static IP routes from the system. These are in addition to RIP if RIP is enabled on LAN1 and or LAN2. Up to 100 routes are supported.

IP Route | IP Route	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- IP Address
  The IP address to match for ongoing routing. Any packets meeting the IP Address and IP Mask settings are routed to the entry configured in the Destination field. When left blank then an IP Address of 255.255.255.255 (all) is used.
- IP Mask
  The Subnet Mask used to mask the IP Address for ongoing route matching. If blank the mask used is 255.255.255.255 (all).
  - A *0.0.0.0* entry in the IP Address and IP Mask fields routes all packets for which there is no other specific IP Route available. The Default Route option with Services 332 can be used to do this if a blank IP route is not added.
- Gateway IP Address: *Default = Blank*
  The address of the gateway where packets for the above address are to be sent. If this field is set to *0.0.0.0* or is left blank then all packets are just sent down to the Destination specified, not to a specific IP Address. This is normally only used to forward packets onto another Router on the local LAN.
- Destination
  Allows selection of LAN1, LAN2 (if supported) and any configured Service, Logical LAN or Tunnel (L2TP only).
- Metric: *Default = 0*
  The number of "hops" this route counts as.
- Proxy ARP: *Default = Off*
  This allows the system to respond on behalf of this IP address when receiving an ARP request.

5.17.2 RIP

Routing Information Protocol (RIP) is a protocol which allows routers within a network to exchange routes of which they are aware approximately every 30 seconds. Through this process, each router adds devices and routes in the network to its routing table.

Each router to router link is called a 'hop' and routes of up to 15 hops are created in the routing tables. When more than one route to a destination exists, the route with the lowest metric (number of hops) is added to the routing table.

When an existing route becomes unavailable, after 5 minutes it is marked as requiring 'infinite' (16 hops). It is then advertised as such to other routers for the next few updates before being removed from the routing table. The system also uses 'split horizon' and 'poison reverse'.

RIP is a simple method for automatic route sharing and updating within small homogeneous networks. It allows alternate routes to be advertised when an existing route fails. Within a large network the exchange of routing information every 30 seconds can create excessive traffic. In addition the routing table held by each system is limited to 100 routes (including static and internal routes).

RIP is supported with system's from Level 2.0 upwards. The normal default is for RIP to be disabled. It can be enabled on LAN1, LAN2 and individual services.

- Listen Only (Passive):
  The system listens to RIP1 and RIP2 messages and uses these to update its routing table. However the system does not respond.

- RIP1:
  The system listens to RIP1 and RIP2 messages. It advertises its own routes in a RIP1 sub-network broadcast.

- RIP2 Broadcast (RIP1 Compatibility):
  The system listens to RIP1 and RIP2 messages. It advertises its own routes in a RIP2 sub-network broadcast. This method is compatible with RIP1 routers.

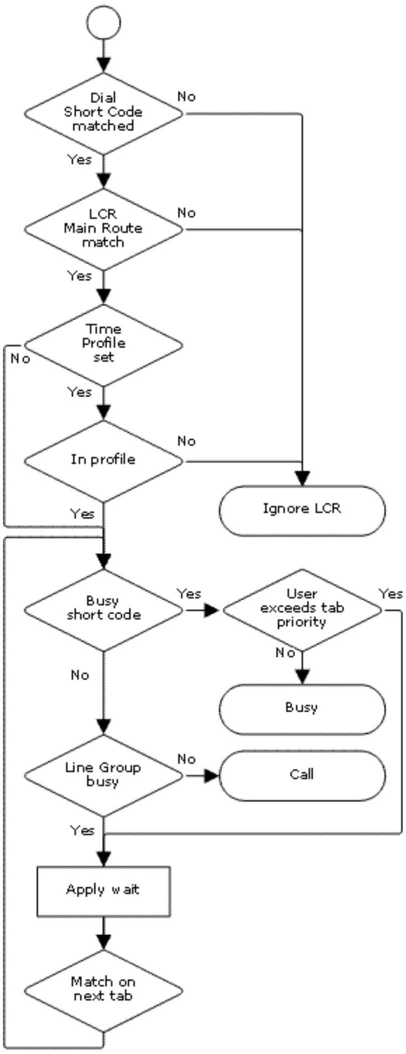
- RIP2 Multicast:
  The system listens to RIP1 and RIP2 messages. It advertises its own routes to the RIP2 multicast address (249.0.0.0). This method is not compatible with RIP1 routers.

Broadcast and multicast routes (those with addresses such as 255.255.255.255 and 224.0.0.0) are not included in RIP broadcasts. Static routes (those in the IP Route table) take precedence over a RIP route when the two routes have the same metric.

5.18 Least Cost Routing Settings

Summary: Least cost routes allow short code matching on the number being dialed from the system rather than the number originally dialed by the user or application.

For IP Office 4.0, Least Cost Routes have been replaced by ARS 407. When a system is upgraded to IP Office 4.0, the LCR entries are automatically replaced by ARS entries and appropriate short codes.



When a line, user, user rights or system short code results in a number to be dialed off-switch, the resulting telephone number to be dialed can be further processed by matching to Least Cost Route (LCR) short codes.

LCR short codes are grouped in sets. Within each set, the short codes are grouped into tabs called Main Route, Alternate Route 1 and Alternate Route 2. Each tab also has a priority and a timeout setting.

- **Using a Time Profile**
  Each LCR set can have an associated time profile. LCR sets without a time profile are active all the time. LCR sets with a time profile are only active within the times defined by that profile.
- **Which Number is Used For Matching**
  The telephone number output by the original matched short code is checked against the Main Route tab short codes of the active LCR sets.
  - If a match is found, that set is used for processing.
  - If no match is found, the calls is dialed without LCR.
- **Returning Busy**
  If the LCR short code match is set to the Busy feature:
  - If the user's priority is higher than the LCR tabs, the system will immediately look for a matching short code on the next tab and use that short code if found.
  - Otherwise the user receives busy tone.
- **Switching Outgoing Line Groups**
  If the LCR short code match is a dial feature, the system will attempt to seize a line from the outgoing line group specified by the LCR short code.
  - If a line cannot be seized within the time specified on the LCR tab, the system will look in the next tab for an alternate LCR short code match. If an alternate match is found it is used.

Least Cost Routing Example

Site A has two outgoing line groups. Outgoing line group 0 contains external lines to the public telephone network. Outgoing line group 1 contains private lines to Site B.

Requirements

- Scenario 1
  The external public number for Site B is 123456. The internal speed dial number is 600. When a user dials 600, the administrator want the call to be routed by the private lines if possible.
- Scenario 2
  The sales hot line at Site B has the public number 654321. The administrator only want high priority users at Site A to be able to dial that number to test its performance.

Settings

- System Short Code 1: 600/123456/Dial/0.
- System Short Code 2: 654321/N/Dial/0.
- User 1: Priority 2. User 2: Priority 4.

Least Cost Route "SiteB"	Main Route	Alternate Route 1
Timeout	10	30
Priority	3	5
Short Codes	123456/N/Dial/1	123456/N/Dial/0
	654321/N/Busy	654321/N/Dial/0

Effects

Scenario 1
When a user dials 123456, it matches system short code 1. That short code specifies dialing Site B via the public lines (Outgoing line group 0).

The number to be dialed is checked against the least cost routing Main Route tabs for any match. In this example a match occurs in the SiteB least cost route. The short code there specifies dialing the number using the private lines (Outgoing line group 1).

If the system cannot seize a line for the call from that group within 10 seconds, it looks for an alternate short code match in Alternate Route 1 tab of the Site B least cost route. In this example that match changes the call to using the public lines (Outgoing line group 0).

Scenario 2
When a user dials 654321, it matches system short code 2. That short code specifies dialing the Site B sale hot line number via the public lines (Outgoing line group 0).

Since this short code is set to a Dial feature, the number to be dialed is checked against the least cost routing Main Route tabs for any match. In this example a match occurs in the SiteB least cost route. The short code there specifies Busy and so returns busy to callers.

User 1 has a priority of 2. They will receive busy tone when they dial 654321.

User 2 has a priority of 4 which is higher than the Main Route tab in the Site B least cost route. Therefore the system will immediately check for a further match in the Alternate Route 1 tab. In this example the short code match for 654321 in the Alternate 1 tab allows the number to be dialed to the public lines.

5.18.1 LCR

Least Cost Routing | LCR	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✗, IP500v2 ✗.
Software Level	1.0 to 3.2 only.
Mergeable	✓.

- Name
  The name to identify the LCR set.

- Time Profile: *Default = Blank*
  Selects a time profile 360 that is used to define when this least cost route can be used. If no profile is selected the route settings apply at all times.

5.18.2 Main Route

This tab is used for the initial short code matching. The match is performed on the telephone number outputs by a Line, User, User Rights or System short code that resulted in a number to dial. If a match is found it is used, otherwise the call is dialed using the original short code.

Least Cost Routing | Main Route	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✗, IP500v2 ✗.
Software Level	1.0 to 3.2 only.
Mergeable	✓.

- Timeout (secs): *Default = 30 seconds. Range = 1 to 99999 seconds or 0 (No IP trunk fallback).*
  If an LCR short code match is found, the system will attempt to seize a line in the outgoing line group specified by that short code. If, after this timeout period has expired, the system still cannot seize a line, it will look for a alternate short code match in the Alternate Route 1 373 tab.

- Priority: *Default = 5, Range 1 (lowest) to 5 (highest).*
  Normally, if the LCR short code match is a short code set to Busy, the user will receive busy tone. For users whose own Priority 264 setting (User | User | Priority) is higher than the Main Route tab's, the system will look for an alternate short code match in the Alternate Route 1 373 tab.

- Allow Bump: *Default = Off*
  When a line indicated by the LCR short code match cannot be seized because lines in that outgoing line group are being used for a multilink PPP data call, this options allows a line to be seized from the data call.

- Short Code List
  These are the short codes used for matching against the telephone number output Line, User, User Right or System dial short code.

  - The only short code features that should be used in a Least Cost Route short code are: *Dial*, *Dial3K1*, *Dial56K*, *Dial64K*, *DialEmergency*, *DialSpeech*, *DialV110*, *DialV120*, *DialVideo* and *Busy*.

  - The ; character and [ ] characters cannot be used.

  - Short codes can be added and edited using the Add, Remove and Edit buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

5.18.3 Alternate Route 1

This tab of a Least Cost Route is used in the following cases:

- The short code match on the Main Route 372 tab is set to Busy but the user has a higher priority than that tab.
- The Main Route 372 tab timeout has expired while trying to seize a line from the outgoing line group specified by the short code match on that tab.

In either case the system will look for an alternate short code match on this tab. The match is performed on the telephone number outputs by a Line, User, User Rights or System short code that resulted in a number to dial. If a match is found it is used, otherwise the call receives busy.

Least Cost Routing | Alternate Route 1	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✗, IP500v2 ✗.
Software Level	1.0 to 3.2 only.
Mergeable	✓.

- Timeout (secs): *Default = 30 seconds. Range = 1 to 99999 seconds or 0 (No IP trunk fallback).*
  If an LCR short code match is found, the system will attempt to seize a line in the outgoing line group specified by that short code. If, after this timeout period has expired, the system still cannot seize a line, it will look for a alternate short code match in the Alternate Route 2 374 tab.
- Priority: *Default = 5, Range 1 (lowest) to 5 (highest).*
  Normally, if the LCR short code match is a short code set to Busy, the user will receive busy tone. For users whose own Priority 264 setting (User | User | Priority) is higher than the Main Route tab's, the system will look for an alternate short code match in the Alternate Route 2 374 tab.
- Allow Bump: *Default = Off*
  When a line indicated by the LCR short code match cannot be seized because lines in that outgoing line group are being used for a multilink PPP data call, this options allows a line to be seized from the data call.
- Short Code List
  These are the short codes used for matching against the telephone number output Line, User, User Right or System dial short code.
  - The only short code features that should be used in a Least Cost Route short code are: *Dial*, *Dial3K1*, *Dial56K*, *Dial64K*, *DialEmergency*, *DialSpeech*, *DialV110*, *DialV120*, *DialVideo* and *Busy*.
  - The *;* character and *[ ]* characters cannot be used.
  - Short codes can be added and edited using the Add, Remove and Edit buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

5.18.4 Alternate Route 2

This tab of a Least Cost Route is used in the following cases:

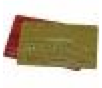
- The short code match on the Alternate Route 1 373 tab is set to Busy but the user has a higher priority than that tab.
- The Alternate Route 1 373 tab timeout has expired while trying to seize a line from the outgoing line group specified by the short code match on that tab.

In either case the system will look for an alternate short code match on this tab. The match is performed on the telephone number outputs by a Line, User, User Rights or System short code that resulted in a number to dial. If a match is found it is used, otherwise the call receives busy.

Least Cost Routing | Alternate Route 2	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✗, IP500v2 ✗.
Software Level	1.0 to 3.2 only.
Mergeable	✓.

- **Timeout (secs)**
  Not used. This is the last tab within a Least Cost Route.
- **Priority**
  Not used. This is the last tab within a Least Cost Route.
- **Allow Bump:** *Default = Off*
  When a line indicated by the LCR short code match cannot be seized because lines in that outgoing line group are being used for a multilink PPP data call, this options allows a line to be seized from the data call.
- **Short Code List**
  These are the short codes used for matching against the telephone number output Line, User, User Right or System dial short code.
  - The only short code features that should be used in a Least Cost Route short code are: Dial, *Dial3K1*, *Dial56K*, *Dial64K*, *DialEmergency*, *DialSpeech*, *DialV110*, *DialV120*, *DialVideo* and Busy.
  - The ; character and [ ] characters cannot be used.
  - Short codes can be added and edited using the Add, Remove and Edit buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

5.19 Account Code Settings



Account codes are commonly used to control cost allocation and out-going call restriction. The account code used on a call is included in the call information output by the system's call log. Incoming calls can also trigger account codes automatically by matching the Caller ID stored with the account code.

Once a call has been completed using an account code, the account code information is removed from the user's call information. This means that redial functions will not re-enter the account code. The maximum recommended number of accounts codes is 1000.

Setting a User to Forced Account Code

1. Receive the system configuration if one is not opened.
2. In the left-hand panel, click  **User**. The list of existing user is shown in the right-hand panel.
3. Double-click the required user.
4. Select the **Telephony** tab.
5. Tick the **Force Account Code** option.
6. Click OK.
7. Merge the configuration.

Forcing Account Code Entry for Specific Numbers
Account code can be set a being required for any dialing that matches a particular short code. This is done by ticking the Force Account Code option found in the short code settings. Note that the account code request happens when the short code match occurs. Potentially this can be in the middle of dialing the external number, therefore the use of X wildcards in the short code to ensure full number dialing is recommended.

Entering Account Codes
The method for entering account codes depends on the type of phone being used. Refer to the relevant telephone User's Guide for details.

- **Account Code Button**
  The Account Code Entry 575 action (*User | Button Programming | Emulation | Account Code Entry*) and Set Account Code 492 action (*User | Button Programming | Advanced | Set | Set Account Code*) can be assigned to a programmable button on some phones. They both operate the same. The button can be preset with a specific account code or left blank to request account code entry when pressed. The button can then be used to specify an account code before a call or during a call.

- **Phone Manager**
  The Phone Manager application can be used to enter account codes before or during calls. For full details refer to the Phone Manager documentation.

  - To enter an account code before making a call or during a call select **Actions | Account Code**. A valid account code can then be selected from the Account Code drop down.
  - The **Account Codes** tab can be used to create icons to speed dial specific numbers and account codes that are regularly used.

- **Setting an Account Code using Short Codes**
  The **Set Account Code** feature allows short codes to be created that specify an account code before making a call.

- **Show Account Code Setting** 160 (System | Telephony | Telephony 160)
  This setting controls the display and listing of system account codes:

  - When on:
    - When entering account codes through Phone Manager, users can select from a drop-down list of available account codes.
    - When entering account codes through a phone, the account code digits are shown while being dialed.
  - When off:
    - Within Phone Manager the drop-down list of available account codes is not useable, instead account codes must be entered using the Phone Managers PIN Code features.
    - When entering account codes through a phone, the account code digits are replaced by s characters on the display.

5.19.1 Account Code

This tab is used to define an individual account code.

Account Code | Account Code	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- **Account Code**
  Enter the account code required. The code can include alphabetic characters for users dialing via Phone Manager. It can also include wildcards; *?* matches a single digit and *** matches any digits.
- **Caller ID**
  A caller ID can be entered and used to automatically assign an account code to calls made to or received from caller ID.

5.19.2 Voice Recording

This tab is used to activate the automatic recording of external calls when the account code is entered at the start of the call or automatically assigned by call ID matching when the call is received.

Call recording requires Voicemail Pro to be installed and running. Call recording also requires available conference resources similar to a 3-way conference.

- IP Office 4.0+ introduces the following changes to recording:
  - Calls to and from IP devices, including those using Direct media, can be recorded.
  - Calls parked or held pause recording until the unparked or taken off hold.
  - Recording is stopped if:
    - User recording stops if the call is transferred to another user.
    - User account code recording stops if the call is transferred to another user.
    - Hunt group recording stops if the call is transferred to another user who is not a member of the hunt group.
    - Incoming call route recording continues for the duration of the call on the system.
- IP Office 4.1+: The destination mailbox for the recording can be specified.

Account Code | Voice Recording	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- **Record Outbound:** *Default = None*
  Select whether automatic recording of outgoing calls is enabled. The **Auto Record Calls** option sets whether just external calls or external and internal calls are included. Options for recording are:
  - **None:** Do not automatically record calls.
  - **On:** Record the call if possible. If not possible to record, allow the call to continue.
  - **Mandatory:** Record the call if possible. If not possible to record, block the call and return busy tone.
  - **Percentages of calls:** Record a selected percentages of the calls.
- **Record Inbound:** *Default = None*
  Select whether automatic recording of incoming calls is enabled. Options for recording are:
  - **None:** Do not automatically record calls.
  - **On:** Record the call if possible. If not possible to record, allow the call to continue.

- Mandatory: Record the call if possible. If not possible to record, block the call and return busy tone.
- Percentages of calls: Record a selected percentages of the calls.

- Record Time Profile: *Default = <None> (Any time)*
Used to select a time profile 360 during which automatic call recording of incoming calls is applied. If no profile is selected, automatic recording of incoming calls is active at all times.
- Recording (Auto): *Default = Mailbox*
Sets the destination for automatically triggered recordings.
  - Mailbox
  This option sets the destination for the recording to be a selected user or hunt group mailbox. The adjacent drop down list is used to select the mailbox.
  - Voice Recording Library: *Software level = 3.0+.*
  This options set the destination for the recording to be a VRL folder on the voicemail server. The ContactStore application polls that folder and collects waiting recordings which it then places in its own archive. Recording is still done by the Voicemail Pro.
- Auto Record Calls: *Default = External. Software level = 6.1*
This setting allows selection of whether *External* or *External & Internal* calls are subject to automatic call recording.

5.20 License

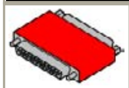
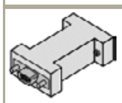
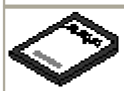
This form is used to display the function, value and status of license keys entered into the system configuration. License keys are 32 character strings uniquely based on the feature they active and the serial number of a Feature Key dongle being used by the control unit.

The serial number is printed on the feature key dongle and prefixed with SN (FK for IP500v2 SD card dongles). It can also be viewed in the system configuration by selecting System | System | Dongle Serial Number.

Feature Key dongles are available in several types. Each system only supports license validation against a single dongle and vice versa. The licenses in the systems configuration must match the serial number of the Feature Key dongle. Depending on the dongle type, it is installed either directly on the control unit or onto a PC running the Feature Key Server application.

- Each will only support license validation against one feature key dongle.
- If being used, a Feature Key Server PC will only validate licenses for the first system to which it connects after starting.

- For parallel and USB feature keys, the address of the PC hosting the dongle and running the Feature Key Server software is set by the License Server IP Address setting on the System | System [143] tab. For serial key dongles, the address is set to 0.0.0.0.
- Note that for IP500 and IP500v2 control units, a Feature Key dongle must be present even if no licensed features are being used.
- The serial number and whether the control unit has an local (serial or smart card) or remote (parallel or USB) Feature Key dongle are indicated by the Dongle Serial Number field on the System | System [143] tab.
- For IP Office 6.0and higher, feature key dongles using the Feature Key server application are not supported.

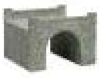
Feature Key Dongle		Server PC	License Server IP Address	SOE	IP403	IP406 V1	IP406 V2	IP412	IP500	IP500 v2
	Parallel This type of feature key dongle is plugged into the parallel port of a PC running the Feature Key Server software.	✓	255.255.255.255 or server PC address.	✓	✓	✓	✓	✓	–	–
	USB This type of feature key dongle is plugged into the USB port of a PC running the Feature Key Server software.	✓	255.255.255.255 or server PC address.	✓	✓	✓	✓	✓	–	–
	Serial This type of feature key dongle is plugged into the 9-pin serial port on the back of Small Office Edition and IP406 V2 control units. No separate PC running Feature Key software is required.	✗	0.0.0.0	✓	–	–	✓	✓	–	–
	Smart Card This type of feature key dongle, a credit-card sized memory card, is plugged into IP500 control units.	✗	Not used.	–	–	–	–	–	✓	–
	SD Memory Card This type of feature key dongle is plugged into IP500v2 control units.	✗	Not used.	–	–	–	–	–	–	✓

Importing License Keys
It is recommended that licenses are cut and pasted electronically. This removes the chances of errors due to mistyping and misinterpretation of characters fonts. Where multiple licenses need to be added, the CSV import option can be used (File | Import/Export | Import). Licenses imported this way may be listed as invalid until the configuration is saved and then reloaded

License | License	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✓.

- License Key
  This field is used to enter the 32-character license key.
- License Type
  For information only. If the key is recognized, its function will be listed here. Invalid indicates that the License Key has not been recognized as enabling any feature, regardless of the Feature Key dongle serial number. For a list of licenses and their purpose refer to the Manager Installation manual.
- License Status
  For information only. This field indicates the current validation status of the license key against the serial number of the Feature Key dongle being used by the system.
  - *Unknown* is shown for newly entered licenses until the configuration is sent to the system and then retrieved again.
  - *Valid* is shown if the license key matches the Feature Key dongle serial number.
  - *Invalid* is shown if the license key does not match the Feature Key dongle serial number.
  - *Dormant* is shown if the license key is valid but is conditional on another license that is not present.
  - *Obsolete* is shown if the license key is valid but the license is no longer used by the version of software installed in the system.
  - *Expired* is shown if the license has passed its expiry date.
- Instances
  For information only. Some licenses enable a number of port, channels or users. When that is the case, the number of such is indicated here. Multiple licenses for the same feature are usually cumulative.
- Expiry Date
  For information only. License can be set to expire within a set period from their issue by Avaya. The expiry date is shown here.

5.21 Tunnel Settings



Tunneling allows additional security to be applied to IP data traffic. This is useful when sites across an unsecure network such as the public internet. The system supports two methods of tunneling, L2TP and IPSec. Once a tunnel is created, it can be used as the destination for selected IP traffic in the IP Route 368 table.

Two types of tunneling are supported:

- L2TP - Layer 2 Tunneling Protocol 381
  PPP (Point to Point Protocol) authentication normally takes place between directly connected routing devices. For example when connecting to the internet, authentication is between the customer router and the internet service provider's equipment. L2TP allows additional authentication to be performed between the routers at each end of the connection regardless of any intermediate network routers. The use of L2TP does not require a license.

- IPSec 383
  IPSec allows data between two locations to be secured using various methods of sender authentication and or data encryption. The use of IPSec requires entry of an IPSec Tunneling license into the system at each end.

5.21.1 L2TP Tunnel

5.21.1.1 Tunnel

Tunnel | Tunnel (L2TP)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- Name: *Default = Blank.*
  A unique name for the tunnel. Once the tunnel is created, the name can be selected as a destination in the IP Route table.
- Local Configuration
  The account name and password is used to set the PPP authentication parameters.
  - Local Account Name
    The local user name used in outgoing authentication.
  - Local Account Password/Confirm Password
    The local user password. Used during authentication.
  - Local IP Address
    The source IP address to use when originating an L2TP tunnel. By default (un-configured), the system uses the IP address of the interface on which the tunnel is to be established as the source address of tunnel.
- Remote Configuration
  The account name and password is used to set the PPP authentication parameters.
  - Remote Account Name
    The remote user name that is expected for the authentication of the peer.
  - Remote Account Password/Confirm Password
    The password for the remote user. Used during authentication.
  - Remote IP Address
    The IP address of the remote L2TP peer or the local VPN line IP address or the WAN IP address.
- Minimum Call Time (Mins): *Default = 60 minutes, Range = 1 to 999.*
  The minimum time that the tunnel will remain active.
- Forward Multicast Messages: *Default = On*
  Allow the tunnel to carry multicast messages when enabled.
- Encrypted Password: *Default = Off*
  When enabled, the CHAP protocol is used to authenticate the incoming peer.

5.21.1.2 L2TP

Tunnel | L2TP	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- Shared Secret/Confirm Password
  User setting used for authentication. Must be matched at both ends of the tunnel. This password is separate from the PPP authentication parameters defined on the L2TP|Tunnel 381 tab.
- Total Control Retransmission Interval: *Default = 0, Range = 0 to 65535.*
  Time delay before retransmission.
- Receive Window Size: *Default = 4, Range = 0 to 65535.*
  The number of unacknowledged packets allowed.
- Sequence numbers on Data Channel: *Default = On*
  When on, adds sequence numbers to L2TP packets.
- Add checksum on UDP packets: *Default = On.*
  When on, uses checksums to verify L2TP packets.
- Use Hiding: *Default = Off*
  When on, encrypts the tunnel's control channel.

5.21.1.3 PPP

Tunnel | PPP (L2TP)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- CHAP Challenge Interval (secs): *Default = 0 (Disabled), Range = 0 to 99999 seconds.*
  Sets the period between CHAP challenges. Blank or 0 disables repeated challenges. Some software (such as Windows 95 DUN) does not support repeated challenges.
- Header Compression: *Default = None*
  Select header compression. Options are: IPHC and/or VJ.
- PPP Compression Mode: *Default = MPPC*
  Select the compression mode for the tunnel connection. Options are: Disable, StacLZS or MPPC.
- Multilink / QoS: *Default = Off*
  Enable the use of Multilink protocol (MPPC) on the link.
- Incoming traffic does not keep link up: *Default = On*
  When enabled, the link is not kept up when the only traffic is incoming traffic.
- LCP Echo Timeout (secs): *Default = 6, Range = 0 to 99999 seconds.*
  When a PPP link is established, it is normal for each end to send echo packets to verify that the link is still connected. This field defines the time between LCP echo packets. Four missed responses in a row will cause the link to terminate.

5.21.2 IP Security Tunnel

5.21.2.1 Main

Tunnel | Main (IPSec)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- Name: *Default = Blank.*
  A unique name for the tunnel. Once the tunnel is created, the name can be selected as a destination for traffic in the IP Route 368 table.
- Local Configuration
  The IP Address and IP Mask are used in conjunction with each other to configure and set the conditions for this Security Association (SA) with regard to inbound and outbound IP packets.
  - IP Address
    The IP address or sub-net for the start of the tunnel.
  - IP Mask
    The IP mask for the above address.
  - Tunnel Endpoint IP Address
    The local IP address to be used to establish the SA to the remote peer. If left un-configured, the system will use the IP address of the local interface on which the tunnel is to be configured.
- Remote Configuration
  The IP Address and IP Mask are used in conjunction with each other to configure and set the conditions for this Security Association (SA) with regard to inbound and outbound IP packets.
  - IP Address
    The IP address or sub-net for the end of the tunnel.
  - IP Mask
    The IP mask for the above address.
  - Tunnel Endpoint IP Address
    The IP address of the peer to which a SA must be established before the specified local and remote addresses can be forwarded.

5.21.2.2 IKE Policies

Tunnel | LKE Policies (IPSec)	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- Shared Secret/Confirm Password
  The password used for authentication. This must be matched at both ends of the tunnel.
- Exchange Type: *Default = ID Prot*
  Aggressive provides faster security setup but does not hide the ID's of the communicating devices. *ID Prot* is slower but hides the ID's of the communicating devices.
- Encryption: *Default = DES CBC*
  Select the encryption method used by the tunnel. The options are: *DES CBC*, *3DES* or *Any*.
- Authentication: *Default = MD5*
  The method of password authentication. Options are: MD5, SHA or Any.
- DH Group: *Default = Group 1*
- Life Type: *Default = KBytes*
  Sets whether Life (below) is measured in seconds or kilobytes.
- Life: *Range = 0 to 99999999.*
  Determines the period of time or the number of bytes after which the SA key is refreshed or re-calculated.

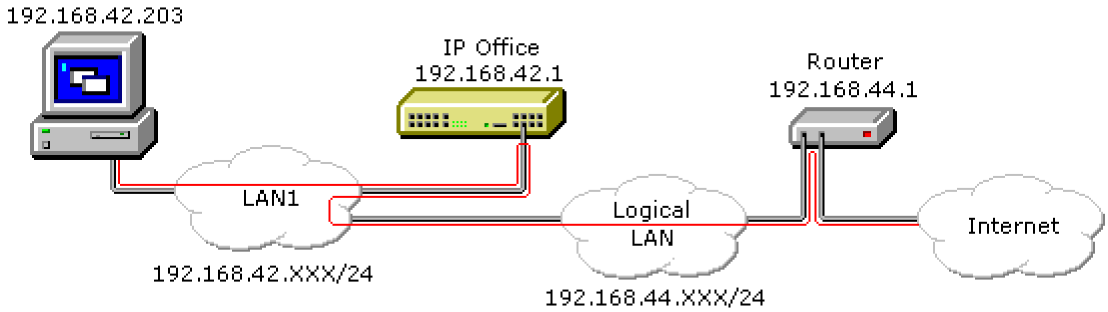
5.21.2.3 IPSec Policies

Tunnel | IPSec Policies	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	✗.

- Protocol: *Default = ESP*
  ESP (Encapsulated Security Payload) or *AH* (Authentication Header, no encryption).
- Encryption: *Default = DES*
  Select the encryption method used by the tunnel. The options are: *DES CBC*, *3DES* or *Any*.
- Authentication: *Default = HMAC M*D5
  The method of password authentication. Options are: HMAC MD5, *HMAC SHA* or *Any*.
- Life Type: *Default = KBytes*
  Sets whether Life (below) is measured in seconds or kilobytes.
- Life
  Determines the period of time or the number of bytes after which the SA key is refreshed or re-calculated.

5.22 Logical LAN

A logical LAN can be used on control units with only a single LAN (LAN1) . A logical LAN allows these systems to support a second separately addressed LAN on the same interface. Traffic between the system LAN1 and the logical LAN can then be controlled by the system's IP route table and firewalls. Logical LAN settings are not supported in IP Office 6.1+.



Logical LAN | Logical LAN	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	2.0 to 6.0.
Mergeable	✗.

- Name: *Default = Blank. Range = Up to 31 characters.*
  A unique name for the logical LAN. This name becomes selectable as a destination in the IP Route table.
- IP Address: *Default = 0.0.0.0*
  The IP address provided by the internet service provider for the logical LAN.
- IP Mask: *Default = 0.0.0.0*
  The IP address mask provided by the internet service provider for the logical LAN.
- Gateway IP Address: *Default = 0.0.0.0*
  The IP address of the router on the logical LAN.
- Gateway Mac Address: *Default = 00:00:00:00:00:00*
  The MAC address of the router. If the MAC address isn't known, from a PC that can ping the router's IP address, use the command arp -a <ip address>.
- Firewall Profile: *Default = Blank*
  This field allows selection of an existing firewall profile that should be applied to traffic to and from the logical LAN.
- Enable NAT: *Default = On (Grayed out).*
  NAT is applied to all traffic from the system LAN to the logical LAN. The use of NAT is not compatible with H.323 VoIP operation, therefore a VPN tunnel should also be applied to traffic being routed

5.23 Wireless Settings

The Small Office Edition control unit can act as an 802.11b wireless access point. To do this requires the insertion of an Avaya supplied wireless card into one of the control unit's PCMCIA slots and entry of a Small Office Edition WiFi license into the configuration. The Manager Wireless settings can then be configured.

In order to connect to the Manager LAN, wireless devices must be configured to match the Manager Wireless settings. Additionally the wireless device must match the control unit's LAN1 or LAN2 network settings unless using DHCP.

5.23.1 SSID

This tab is used to set the general identity of the wireless connection to the system LAN.

Wireless | SSID	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✗, IP412 ✗, IP500 ✗, IP500v2 ✗.
Software Level	2.0+.
Mergeable	✗.

- Network Name: *Default = IP Office Wireless.Net*
  A unique name used to identify and distinguish the system wireless LAN from other wireless LAN's. This is the wireless LAN's Service Set Identifier (SSID).
- Wireless Mac Address
  Displays a list of the MAC addresses of the devices currently connected to the wireless LAN.
- Frequency/Channel: *Default = 6*
  The 802.11b wireless frequency band is sub-divided into a number of channels. In locations where there are multiple wireless LAN's or multiple access points to the same wireless LAN, each access point should use a separate channel. Devices connecting to a wireless LAN will automatically connect to the channel providing the strongest signal.
  - The number of channels available is country specific. In the US channels 1 to 11 are available. In most of Europe, channels 1 to 13 are available. In Japan only channel 14 is available.
  - The channel frequencies overlap. For instance, channel 2 shares part of the same frequency band as channels 1 and 3. In areas with multiple access points or LAN, use widely spaced channels. For example uses channels 1, 6 and 11 on different access points.
- Accept Any: *Default = Off*
  If on, allows any wireless device to connect to the wireless LAN without having to have a matching wireless network name (SSID) set. When off, only devices configured with a matching wireless network name can connect to the wireless LAN.

5.23.2 Security

This tab allows for additional security through the use of WEP wireless encryption keys. If enabled, in addition to encrypting the wireless traffic, only devices using a matching encryption key can connect to the wireless LAN.

Wireless | Security	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✗, IP412 ✗, IP500 ✗, IP500v2 ✗.
Software Level	2.0+.
Mergeable	✗.

- Encryption: *Default = Disabled*
  Allows selection of 50/64 bit or 128 bit security. Note: 50/64 bit encryption is also know as 40/64 encryption in some locales.
- Alpha/Hex: *Default = Hex*
  Switch key entry between hexadecimal and alphabetic entry modes.
- Key 1/4
  Allows entry of the security key and selection of which key is the current key to use.

5.24 User Restrictions Settings

User Restrictions are only available in pre-IP Office 3.2 configurations. For IP Office 3.2 and higher they have been replaced by User Rights 389.

Within Manager, users can be grouped by the types of numbers they are allowed to dial or not allowed to dial. For example, those who are allowed to dial 1900 or international numbers.

The User Restriction form allows named groups of dialing short codes/restrictions to be created. These short codes can then be applied to a user by associating them with the User Restriction name rather than having to recreate the short codes for each user.

To set up a restriction within the User Restriction form

1. Click User Restriction form within the Configuration Tree.
2. Enter a name for the restriction.
3. Click the Short Code List tab and create a short code.
4. Merge the configuration.

To apply a User Restriction to a specific user

1. Click the User form within the Configuration Tree.
2. Double-click the user for whom you want this restriction applied.
3. Within the User tab, click the Restriction drop down box and select the User Restriction you want applied to this user.
4. Merge the configuration.

5.24.1 User Restrictions

User Restrictions are only available in pre-IP Office 3.2 configurations. For IP Office 3.2 and higher, they have been replaced by User Rights 389.

User Restrictions | Restrictions	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✗, IP500v2 ✗.
Software Level	2.0 to 3.1 only.
Mergeable	✓.

- Name: *Default = Blank*
  A name used to identify the set of user restrictions and allow its selection through the Restrictions field in each individual user's User settings.
- Priority: *Default = 5 (highest), Range 0 to 5*
  The priority that should be applied to user calls if routed via a Least Cost Route 370. This overrides the priority of the individual user.
- Outgoing Call Bar: *Default = Off.*
  When on, bars users making external calls.

5.24.2 Short Codes

User Restrictions are only available in pre-IP Office 3.2 configurations. For IP Office 3.2 and higher they have been replaced by User Rights 389.

- WARNING
  User dialing of emergency numbers must not be blocked by the addition of short codes. If short codes are added, the users ability to dial emergency numbers must be tested and maintained.

Allows entry of short codes for dialing by associated users. These short codes override any match system short codes but not individual user short codes.

User Restrictions | Short Codes	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✗, IP500v2 ✗.
Software Level	2.0 to 3.1 only.
Mergeable	✓.

5.25 User Rights Settings

User Rights act as templates for users, locking selected user settings to the template value. For most of the settings within the user rights tabs, the following options can be selected from an adjacent drop down list. Note that some settings are grouped and are set and locked as a group.

- Apply User Rights Value
  Apply the value set in the user rights to all associated users.
  - The matching user setting is grayed out and displays a lock symbol.
  - Users attempting to change the settings using short codes receive inaccessible tone.
  - Within the user's Phone Manager the associated fields are grayed out or hidden.
- Not Part of User Rights
  Ignore the setting.

Adding User Rights

1. Select User Rights.
2. Click and select User Rights.
3. Enter a name.
4. Configure the user rights as required.
5. Click OK.

Creating User Rights Based on an Existing User

1. Select User Rights.
2. In the group pane, right-click and select New User Rights from User.
3. Select the user and click OK.

Associating User Rights to a User

1. Select User Rights or User.
2. In the group pane, right-click and select Apply User Rights to Users.
3. Select the user rights to be applied.
4. On the Members of this User Rights sub tab select the users to which the user rights should be applied as their Working Hours User Rights.
5. On the Members when out of hours sub tab select which users should use the selected user rights as their out of hours user rights.
6. Click OK.

or

1. Select the required user to display their settings in the details pane.
2. Select the User tab.
3. Use Working Hours User Rights drop-down to select the user rights required.
4. If required a Working Hours Time Profile and Out of Hours User Rights can be selected.
5. Click OK.

Copy User Rights Settings over a User's Settings

This process replaces a user's current settings with those that are part of the selected user rights. It does not associate the user with the user rights.

1. Select User Rights or User.
2. In the group pane, right-click and select Copy user rights values to users.
3. Select the user rights to be applied.
4. Click OK.

Default User Rights

For defaulted systems, the following user rights are created as a part of the default configuration. Fields not listed are not part of the user rights.

✓ = Set to On. ✗ = Set to Off. – = Not part of the user rights.

User Rights	Call Center Agent	Boss	Application	Default	IP Hard Phone	Mailbox	Paging	T3
Priority	✓ 5	✓ 5	✓ 5	✓ 5	✓ 5	✓ 5	✓ 5	✓ 5
Voicemail	✓	–	–	–	–	✓	–	–
Voicemail Ringback	✗	✗	✗	✗	✗	✗	–	✗
Outgoing Call Bar	✗	✗	✗	✗	✗	✗	✗	✗
No Answer Time	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0
Transfer Return Time	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0	✓ 0
Individual Coverage Time	✓ 10	✓ 10	✓ 10	✓ 10	✓ 10	✓ 10	✓ 10	✓ 10
Busy on Held	✓	✗	✓	✗	✗	–	–	✗
Call Waiting	✗	✗	✓	✗	✗	✗	✗	✓
Can be Intruded	✗	✗	✗	✗	✗	✗	✗	✗
Cannot be Intruded	✗	✗	✓	✓	✓	✗	✗	✗
Force Login	✓	–	–	–	–	–	–	–
Force Account Code	✗	✗	✗	✗	✗	✗	✗	✗
Button Programming	1: a= 2: b= 4: HGEna 5: DNDOn 6: Busy	1: a= 2: b= 3: c= 6: DNDOn 7: Dial *17	–	1: a= 2: b= 3: c=	1: a= 2: b= 3: c= 6: Dial *17	–	–	–

5.25.1 User

This tab is used to set and lock various user settings.

User Rights | User	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- Name
  The name for the user rights . This must be set in order to allow the user rights to be selected within the User Rights drop down list on the User | User 264 tab of individual users.
- Locale: *Default = Blank*
  Sets and locks the language used for voicemail prompts to the user, assuming the language is available on the voicemail server. On a digital extension it also controls the display language used for messages from the system to the phone. See Supported Country and Locale Settings 844.
- Priority: *Default = 5, Range 1 (Lowest) to 5 (Highest)*
  Sets and locks the user's priority setting for least cost routing.
- Do Not Disturb: *Default = Off*
  Sets and locks the user's DND status setting.
- Voicemail On: *Default = On*
  Moved to the Voicemail 399 tab.
- Voicemail Ringback: *Default = Off*
  Moved to the Voicemail 399 tab.

5.25.2 Short Codes

This tab is used to set and lock the user's short code set. The tab operates in the same way as the User | Short Codes tab. Where the same short code exists in both the User | Short Codes tab and the associated User Rights | Short Codes tab, the system will use the user short code.

User and User Rights short codes are only applied to numbers dialed by that user. For example they are not applied to calls forwarded via the user.

- WARNING
  User dialing of emergency numbers must not be blocked by the addition of short codes. If short codes are added, the users ability to dial emergency numbers must be tested and maintained.

User Rights | Short Codes	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

Short codes can be added and edited using the Add, Remove and Edit buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

5.25.3 Telephony

This tab allows various user telephony settings to be set and locked. These match settings found on the User | Telephony 274 tab.

User Rights | Telephony	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

These settings are divided in to a number of sub-tabs:

- Call Settings 392
- Supervisor Settings 393
- Multi-line Options 394
- Call Log 394

5.25.3.1 Call Settings

User Rights | Telephony	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- No Answer Time: *Default = Blank (Use system setting), Range = 6 to 99999 seconds.*
  Sets how long a call rings the user before following forwarded on no answer if set or going to voicemail. Leave blank to use the system default setting.
- Transfer return Time (secs): *Default = Blank (Off), Range 1 to 99999 seconds.*
  Sets the delay after which any call transferred by the user, which remains unanswered, should return to the user if possible.
- Wrap up Time (secs): *Default = 2 seconds, Range 0 to 99999 seconds.*
  Specifies the amount of time after ending one call before another call can ring. You may wish to increase this in a "call center" environment where users may need time to log call details before taking the next call. It is recommended that this option is not set to less than the default of 2 seconds. 0 is used for immediate ringing.
- Call waiting on/Enable call waiting: *Default = Off*
  For users on phones without appearance buttons, if the user is on a call and a second call arrives for them, an audio tone can be given in the speech path to indicate a waiting call (the call waiting tone varies according to locale). The waiting caller hears ringing rather than receiving busy. There can only be one waiting call, any further calls receive normal busy treatment. If the call waiting is not answered within the no answer time, it follows forward on no answer or goes to voicemail as appropriate. User call waiting is not used for users on phones with multiple call appearance buttons. Call waiting can also be applied to hunt group calls, see Hunt Group | Hunt Group | Call Waiting 306.
- Busy on held/Enable busy on Held: *Default = On*
  If on, when the user has a call on hold, new calls receive busy tone (ringing for incoming analog call) or are diverted to voicemail if enabled, rather than ringing the user. Note this overrides call waiting when the user has a call on hold.

5.25.3.2 Supervisor Settings

User Rights | Telephony	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- Can Intrude: *Default = Off*
  Check this option if the User can interrupt other user's calls. This setting and the setting below are used to control the use of the following short code and button features: Call Intrude, Call Listen, Call Steal and Dial Inclusion.

- Cannot be Intruded: *Default = On*
  If checked, this user's calls cannot be interrupted or acquired. In addition to the features listed above, this setting also affects whether other users can use their appearance buttons to bridge into a call to which this user has been the longest present user.

- Force Login: *Default = Off*
  If checked, the user must log in using their Login Code to use an extension. For example, if Force Login is ticked for User A and user B has logged into A's phone, after B logs off A must log back. If Force Login was not ticked, A would be automatically logged back in.

- Force Account Code: *Default = Off*
  If checked, the user must enter a valid account code to make an external call.

- Inhibit Off-Switch Forward/Transfer: *Default = Off*
  When enabled, this setting stops the user from transferring or forwarding calls externally. Note that all user can be barred from forwarding or transferring calls externally by the System | Telephony | Telephony | Inhibit Off-Switch Forward/Transfers 160 setting.

- CCR Agent: *Default = Off. Software level = 4.2+.*
  This field is used by the CCR application to indicate which users are Agents monitored by that application. It also indicate to the system those users who can use other CCR features within the system configuration. If a user is set as an CCR Agent, Forced Login is enabled and greyed out from being changed and a warning is given if the user does not have a log in code set.

  - WARNING
    This setting should not be enabled/disabled for a user by using User Rights associated with a Time Profile. Do so will cause invalid data to be recorded in the Customer Call Reporter applications database.

  - The number of simultaneous logged in CCR Agents supported by the system is controlled by licenses entered into the configuration. If all CCR Agent licenses on a system have been used, additional agents are prevented from logging in.

- After Call Work Time: *Default = System Default, Range = 10 to 999 seconds. Software level = 4.2+.*
  CCR Agents (see above) can be automatically put into *After Call Work* (ACW) state after ending a hunt group call. During ACW state, hunt group calls are not presented to the user. If set to *System Default*, the value set in Default After Call Work 184 (System | CCR) 184 is used.

- Automatic After Call Work: *Default = Off. Software level = 4.2+.*
  For CCR Agents with Automatic After Call Work enabled, this value sets the duration of the ACW period.

- Outgoing Call Bar: *Default = Off*
  When set, bars the user from making external calls.

- Coverage Group: *Default = <None>. Software level = 5.0+*
  If a group is selected, the system will not use voicemail to answer the users unanswered calls. Instead the call will continue ringing until either answered or the caller disconnects. For external calls, after the users no answer time, the call is also presented to the users who are members of the selected Coverage Group. For further details refer to Coverage Groups 748.

5.25.3.3 Multi-line Options

User Rights | Telephony	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- Individual Coverage Time (secs): *Default = 10 seconds, Range 1 to 99999 seconds.*
  This function sets how long the phone will ring at your extension before also alerting at any call coverage users. This time setting should not be equal to or greater than the No Answer Time.

5.25.3.4 Call Log

User Rights | Telephony	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✓.

- Centralized Call Log: *Default = System Default (On)*
  This setting allows the use of centralized call logging to be enabled or disabled on a per user basis. The default is to match the system setting Default Centralized Call Log On 165 (System | Telephony | Call Log 165). The other options are *On* or *Off* for the individual user. If off is selected, the call log shown on the users phone is the local call log stored by the phone.
- Delete entries after (hours:minutes): *Default = 00:00 (Never). Software level = 6.1+.*
  If a time period is set, entries in the user's call log are automatically deleted after this period.
- Groups: *Default = System Default (On).*
  This section contains a list of hunt groups on the system. If the system setting Log Missed Huntgroup Calls 160 (System | Telephony | Call Log) 160 has been enabled, then missed calls for those groups selected are shown as part of the users call log. The missed calls are any missed calls for the hunt group, not just group calls presented to the user and not answered by them.

5.25.4 Button Programming

This tab is used to set and lock the user's programmable button set. When locked, the user cannot use Admin or Admin1 buttons on their phone to override any button set by their user rights.

Buttons not set through the user rights can be set through the user's own settings.

When Apply user rights value is selected, the tab operates in the same manner as the User | Button Programming 288 tab.

User Rights | Button Programming	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

Adding Blank Buttons
There are scenarios where users are able to program their own buttons but you may want to force certain buttons to be blank. This can be done through the user's associated User Rights as follows:

1. Assign the action Emulation | Inspect to the button. Enter some spaces as the button label.

2. When pressed by the user, this button will not perform any action. However it cannot be overridden by the user.

5.25.5 Menu Programming

This tab is used to set and lock the user's programmable button set.

When Apply User Rights value is selected, the tab operates in the same manner as the User | Menu Programming 289 tab.

User Rights | Menu Programming	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2-4.2.
Mergeable	✓.

5.25.6 Phone Manager

This tab is used to set and lock which parts of Phone Manager the associated users can use or adjust.

User Rights | Phone Manager	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- Allow user to modify Phone Manager settings: *Default = On*
  This setting is used with the Phone Manager Status Options, Screen Pop Options and Hide Options. It controls whether those options are applied every time the user starts Phone Manager or only the first time the user starts Phone Manager.
  - If this setting is enabled, then the system configuration setting of those options are only applied the first time a user starts Phone Manager on a PC. Those settings become part of the user's Phone Manager profile on that PC. They can be changed by the user through Phone Manager. On subsequent Phone Manager starts the Manager settings are ignored.
  - If this setting is not enabled, the system configuration settings are applied every time the user starts Phone Manager and cannot be overridden by the user.
- Agent Mode: *Default = Off*
  This option controls the setting of the Agent Mode option on the Configure Preferences | Agent Mode tab within Phone Manager Pro. When enabled, the user has additional toolbar controls for Busy Wrap Up, Busy Not Available and Select Group. Note that the options on the Phone Manager Pro Agent Mode tab can be grayed out from user changes by the Agent Mode setting in Configuration Options below.
- Phone Manager Type: *Default = Lite*
  Determines the mode in which the user's copy of the Phone Manager application operates. This setting cannot be changed by the user.
  - Lite
    Basic Phone Manager mode. This mode does not require any licenses.
  - Pro
    Advanced Phone Manager mode that enables a range of additional functions. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.
  - Phone Manager PC Softphone
    This is the VoIP IP phone mode of Phone Manager Pro. This mode requires both an available Phone Manager Pro license and a Phone Manager Pro IP Audio Enabled license. The user must be associated with an VoIP extension within the system configuration.
  - Pro Telecommuter: *Software level = 4.1+.*
    This version of Phone Manager Pro is supported with Phone Manager 4.1+. It allows the user to make and receive calls via an external phone specified at Phone Manager log in. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.
- Enable VoIP: *Default = Off*
  This option only appears if the selected Phone Manager Type is Phone Manager PC Softphone. It enables or disables the matching setting on the user's Phone Manager PC Softphone.
- Configuration Options
  These options allow the user access to the indicated configure preferences tabs within Phone Manager.
  - The controllable tabs for Phone Manager Lite are Telephone and Do Not Disturb.
  - IP Office 4.0+: The Mobile Twinning option on the Forwarding tab is also controllable (the twinning number remains editable even if the Mobile Twinning option has been restricted).
  - The additional controllable tabs for Phone Manager Pro and Phone Manager PC Softphone are Screen Pop, Compact Mode, Agent Mode, Voicemail (Voicemail and Voicemail Ringback controls only).
- Screen Pop Options
  These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone screen pop options Ringing, Answering, Internal, External and Outlook. The Allow user to modify Phone Manager settings option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.
- Phone Manager Status Options
  These options allow selection of the tabs to show within the call history area of the user's Phone Manager. The tabs selectable for Phone Manager are All, Missed, Status and Messages. The additional tabs selectable for Phone Manager Pro and PC Softphone are Incoming, Outgoing and Account Code. The Allow user to modify Phone Manager settings option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.

- Hide Options
  These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone options Hide on close and Hide on no calls. The Allow user to modify Phone Manager settings option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.

5.25.7 Twinning

This tab is used to set and lock the following settings relating to the use of mobile twinning. Use of mobile twinning requires entry of a mobile twinning license. This tab is no long available for IP Office 4.2+.

User Rights | Twinning	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2 to 4.1 only.
Mergeable	✓.

- **Mobile Dial Delay**
  Sets and locks the dial delay applied to calls eligible for mobile twinning.
- **Hunt group calls eligible for mobile twinning**
  Sets whether mobile twinning is applied to hunt group calls.
- **Forwarded calls eligible for mobile twinning**
  Sets whether mobile twinning is applied to forwarded calls.

5.25.8 User Rights Membership

The tabs display the users associated with the user rights. and allows these to be changed.

User Rights | User Rights Membership	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- **Members of this User Rights**
  This tab indicates those users associated with the user rights. If the user has an associated Working hours time profile, their association to the user rights applies only during the periods defined by the time profile. If the user does not have an associated Working hours time profile, they are associated with the user rights at all times.
- **Members when out of service**
  This tab indicates those users associated with the user rights outside the time periods defined by their Working hours time profile. The Members when out of service tab is not populated unless there are time profiles available within the configuration.

5.25.9 Voicemail

The tabs display the users associated with the user rights. and allows these to be changed.

User Rights | User Rights Membership	
Control Unit	SOE ✗, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	5.0+.
Mergeable	✓.

- Voicemail On *Default = On*
  When on, the mailbox is used by the system to answer the user's unanswered calls or calls when the user's extension returns busy. Note that selecting off does not disable use of the user's mailbox. Messages can still be forward to their mailbox and recordings can be placed in it. The mailbox can also still be accessed to collect messages.

- Voicemail Ringback: *Default = Off*
  When enabled and a new message has been received, the voicemail server calls the user's extension to attempt to deliver the message each time the telephone is put down. Voicemail will not ring the extension more than once every 30 seconds.

- DTMF Breakout
  When a caller is directed to voicemail to leave a message, they can be given the option to be transferred to a different extension. The greeting message needs to be recorded telling the caller the options available. The extension numbers that they can be transferred to are entered in the fields below. IP Office 5.0+: These system default values can be set for these numbers and are used unless a different number is set within these user settings.

  - Reception / Breakout (DTMF 0)
    The number to which a caller is transferred if they press *0* while listening to the mailbox greeting rather than leaving a message (**0* on embedded voicemail).

    - For systems set to Intuity emulation mode, the mailbox user can also access this option when collecting their messages by dialing **0*.

    - If the mailbox has been reached through a call flow containing a Leave Mail action, the option provided when *0* is pressed are:

      - For IP Office mode, the call follows the Leave Mail action's *Failure* or *Success* results connections depending on whether the caller pressed *0* before or after the record tone.

      - For Intuity mode, pressing *0* always follows the Reception / Breakout (DTMF 0) setting.

  - Breakout (DTMF 2)
    The number to which a caller is transferred if they press 2 while listening to the mailbox greeting rather than leaving a message (**2* on embedded voicemail). Pre-IP Office 5 this option is not support for Voicemail Pro running in IP Office mailbox mode.

  - Breakout (DTMF 3)
    The number to which a caller is transferred if they press *3* while listening to the mailbox greeting rather than leaving a message (**3* on embedded voicemail). Pre-IP Office 5 this option is not support for Voicemail Pro running in IP Office mailbox mode.

5.26 Auto Attendant Settings

The Small Office Edition, IP406 V2, IP500 and IP500v2 control units support embedded voicemail. This is setup by adding an Avaya embedded voicemail memory card to the control unit and then selecting *Embedded Voicemail* as the Voicemail Type on the System | Voicemail 157 tab.

- This tab and its settings are hidden unless the system has been configured to use embedded voicemail on the System | Voicemail 157 tab.
- For full details on configuration and operation of Embedded Voicemail auto-attendants refer to the IP Office Embedded Voicemail Installation Manual.

Up to 40 auto-attendant services can be configured. For pre-IP Office 4.1 systems, only 4 auto-attendant services are supported.

Embedded voicemail services include auto-attendant, callers accessing mailboxes to leave or collect messages and announcements to callers waiting to be answered.

- The IP500/IP500v2 supports 2 simultaneous embedded voicemail calls by default but can be licensed for up to 6. The licensed limit applies to total number of callers leaving messages, collecting messages and or using an auto attendant.
- The IP406 V2 and IP500 support up to 4 simultaneous calls to embedded voicemail services.
- The Small Office Edition supports up to 10 simultaneous calls to embedded voicemail depending on available voice compression channels. A call from an IP device to voicemail uses two voice compression channels on the Small Office Edition.

In addition to basic mailbox functionality, embedded voicemail can also provide auto-attendant operation. Each auto attendant can use existing time profiles to select the greeting given to callers and then provide follow on actions relating to the key presses 0 to 9, * and #.

- Time Profiles
  Each auto attendant can use up to three existing time profiles 360, on each for Morning, Afternoon and Evening. These are used to decide which greeting is played to callers. They do not change the actions selectable by callers within the auto attendant. If the time profiles overlap or create gaps, then the order of precedence used is morning, afternoon, evening.
- Greetings
  Four different greetings are used for each auto attendant. One for each time profile period. This is then always followed by the greeting for the auto-attendant actions. By default a number of system short codes are automatically created to allow the recording of these greetings from a system extension. See below.
- Actions
  Separate actions can be defined for the DTMF keys 0 to 9, * and #. Actions include transfer to a specified destination, transfer to another auto-attendant transfer to a user extension specified by the caller (dial by number) and replaying the greetings.
  - IP Office 4.0+: The Fax action can be used to reroute fax calls when fax tone is detected by the auto-attendant.
  - IP Office 5.0+: The Dial by Name action can be used to let callers specify the transfer destination.
- Short Codes
  Adding an auto attendant automatically adds a number of system short codes. These use the Auto Attendant short code feature. These short codes are used to provide dialing access to record the auto attendant greetings.
  - For pre-IP Office 4.1
    Four system short codes are automatically added for each auto attendant. These use a telephone number of the form AA:*Name*.*Y* where *Name* is replaced by the Auto Attendant name and *Y* is 1, 2, 3 or 4 for the morning, afternoon, evening or menu option greeting.
    - You can manually delete the short codes or add additional short codes as required.
    - To create a short code to access an auto attendant, for example to allow internal calls to an auto attendant, omit the *.Y* part of the short code telephone number.
  - For IP Office 4.1+
    Four system short codes (**81XX*, **82XX*, **83XX* and **84XX*) are automatically added for use with all auto attendants, for the morning, afternoon, evening and menu options greetings respectively. These use a telephone number of the form "AA:"*N*".*Y*" where *N* is the replaced with the auto attendant number dialed and *Y* is 1, 2, 3 or 4 for the morning, afternoon, evening or menu option greeting.
    - An additional short code of the form (for example) **80XX / Auto Attendant / "AA:"N* can be added manual if internal dialed access to auto attendants is required.
    - For IP Office 4.1+, to add a short code to access a specific auto attendant, the name method as used for pre-IP Office 4.1 should be used.

- Routing Calls to the Auto Attendant
  The telephone number format *AA:Name* can be used to route callers to an auto attendant. It can be used in the destination field of incoming call routes and telephone number field of short codes set to the Auto Attend feature.

5.26.1 Auto Attendant

This tab is used to define the name of the auto attendant service and the time profiles that should control which auto attendant greetings are played.

Auto Attendant | Auto Attendant	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✗, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✓.

- Name: *Range = Up to 12 characters*
  This field sets the name for the auto-attendant service. External calls can be routed to the auto attendant by entering AA:Name in the destination field of an Incoming Call Route.

- Maximum Inactivity: *Default = 8 seconds; Range = 1 to 20 seconds. Software level = 3.0+.*
  This field sets how long after playing the prompts the Auto Attendant should wait for a valid key press. If exceeded, the caller is either transferred to the Fallback Extension set within the Incoming Call Route used for their call or else the caller is disconnected.

- Enable Local Recording: *Default = On. Software level = 4.0+ (Q2 2007 Maintenance release).*
  When off, use of short codes to record auto-attendant prompts is blocked. The short codes can still be used to playback the greetings.

- Direct Dial-By-Number: *Default = Off. Software level = 6.0+.*
  This setting affects the operation of any key presses in the auto attendant menu set to use the *Dial By Number* action.

  - If selected, the key press for the action is included in any following digits dialed by the caller for system extension matching. For example, if 2 is set in the actions to *Dial by Number*, a caller can dial 201 for extension 201.

  - If not selected, the key press for the action is not included in any following digits dialed by the caller for system extension matching. For example, if 2 is set in the actions to *Dial by Number*, a caller must dial 2 and then 201 for extension 201.

- Dial by Name Match Order: *Default = First Name/Last Name. Software level = 5.0+.*
  Determines the name order used for the Embedded Voicemail Dial by Name function. The options are *First then Last* or *Last then First*.

- AA Number: *Software level = 4.1+.*
  This number is assigned by the system and cannot be changed. It is used in conjunction with short codes to access the auto attendant service or to record auto attendant greetings.

- Morning/Afternoon/Evening/Menu Options:
  Each auto-attendant can consist of three distinct time periods, defined by associated time profiles. A greeting can be recorded for each period. The appropriate greeting is played to callers and followed by the Menu Options greeting which should list the available actions.

  - Time Profile
    The time profile that defines each period of auto-attendant operation. When there are overlaps or gaps between time profiles, precedence is given in the order morning, afternoon and then evening.

  - Short code
    These fields indicate the system short codes automatically created to allow recording of the time profile greetings and the menu options prompt.

  - Recording Name: *Default = Blank. Range = Up to 31 characters. Software level = 4.0+ (Q2 2007 Maintenance release).*
    This field appears next to the short code used for manually recording auto-attendant prompts. It is only used is using pre-recorded wav files as greeting rather than manually recording greetings using the indicated short codes. If used, note that the field is case sensitive and uses the name embedded within the wav file file header rather than the actual file name.

    - IP Office 4.1+: This field can be used with all systems supporting embedded voicemail. The utility for converting . wav files to the correct format is provided with Manager and can be launched via File | Advanced | LVM Greeting Utility 124. Files then need to be manually transferred to the embedded voicemail memory card. For full details refer to the IP Office Embedded Voicemail Installation manual.

5.26.2 Actions

This tab defines the actions available to callers dependant on which DTMF key they press. To change an action, select the appropriate row and click Edit. When the key is configured as required click OK.

Auto Attendant | Actions	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✗, IP500 ✓, IP500v2 ✓.
Software Level	2.0+.
Mergeable	Pre-3.2 ✗, 3.2+ ✓.

- Key
  The standard telephone dial pad keys, 0 to 9 plus * and #.
  - IP Office 4.0+: The option Fax can be used for a blind transfer to the required fax destination and will then be triggered by fax tone detection. If left as Not Defined, fax calls will follow the incoming call routes fallback settings once the auto-attendant Maximum Inactivity Time set on the Auto Attendant | Auto Attendant 402 tab is reached.
- Action
  The following actions can be assigned to each key.
  - *Blind Transfer: Software level = 4.0+.*
    Transfer the call to the selected destination. This is an unsupervised transfer, if the caller is not answered they will be handled as per a direct call to that number.
  - *Dial by Name: Software level = 5.0+.*
    Callers are asked to dial the name of the user they require and then press #. The recorded name prompts of matching users are then played back for the caller to make a selection. The name order used is set by the Dial by Name Match Order setting on the Auto Attendant 402 tab. Note the name used is the user's Full Name if set, otherwise their User Name is used. Users without a recorded name prompt or set to Ex Directory are not included. Users can record their name by accessing their mailbox and dialing *05.
  - *Dial By Number: Software level = 4.0+.*
    This option allows callers with DTMF phones to dial the extension number of the user they require. No destination is set for this option. The prompt for using this option should be included in the auto attendant Menu Options greeting. A uniform length of extension number is required for all users and hunt group numbers. IP Office 6.0: The operation of this action is affected by the auto attendant's Direct Dial-by-Number 402 setting.
  - *Normal Transfer*
    Transfer the caller to the selected destination. This is an unsupervised transfer, if the caller is not answered they will be handled as per a direct call to that number. If no destination is set, the caller can dial the user extension number that they require.
  - *Not Defined*
    The corresponding key takes no action.
  - *Replay Menu Greeting*
    Replay the auto-attendant greetings again.
  - *Transfer to Attendant: Software level = 4.0+.*
    This action can be used to transfer calls to another existing auto attendant.
  - *Transfer to Operator: Software level = Up to 3.2 only.*
    Transfer the caller to the selected destination. Operates the same as the Normal Transfer option below.
- Destination
  Sets the destination for the action:
  - Destination can be a user, a hunt group or a short code.
  - If the destination field is left blank, callers can dial the user extension number that they require. Note however that no prompt is provided for this option so it should be included in the auto attendant Menu Options greeting.

5.27 Authorization Codes

Authorization codes are not shown by default. Manager must be modified in order to support authorization codes. Similarly in order to record authorization codes used with calls in the system SMDR, the Delta Server software must be modified.

Each authorization code is associated with a particular user or user rights set. The user or users associated with the user rights, can then dial numbers which are set to trigger forced authorization code entry. Once a code is entered, the short code settings of the user or user rights with which the code is associated are used to completed the call.

This can be used to allow authorized users to make otherwise restricted calls from any extension without first having to log in to that extension and then log out after the call. Authorization code usage can be recorded along with the call details by the Delta Server in its SMDR output, including valid/invalid code entry and the code used.

Overview

- A user dials a number that matches a short code set to Force Authorization Code.
- The user is prompted to enter an authorization code.
- They dial their authorization code. If a matching entry is found in Authorization Codes entries the system checks the corresponding user or user right (in that order). Note that the user or user right checked does not necessarily need to be connected with the user dialing or the user whose extension is being used to make the call.
  - The dial string is checked against the short codes with the matching user or user rights.
  - If it matches a dial short code or no short code the call is allowed, otherwise it is blocked. Note that the short code is not processed, it is just checked for a match.
  - If multi-tier authorization codes are required there must be blocking (busy) short codes (or a wild card '?')

Example
A restaurant has a number of phones in publicly accessible areas and so want to control what calls can be made. They want the phones to anyone to make allow local calls, restrict restaurant staff to locale and cell phone (044...) numbers while the manager can dial local, cell phone and long distance (01...) numbers).

<table>
<tr><th>ARS Table</th><th>Authorization Codes</th></tr>
<tr><td>In the Main (50) ARS table, they add the following short codes:
• 044XXXXXXXXXX / 044N / Dial / Force Auth Code
• 01XXXXXXXXXX / 01N / Dial / Force Auth Code</td><td>They have two authorization codes configured.
• 2008 is associated with a set of User Rights called Cell.
• 1983 is associate with a set of User Rights called LDandCell.</td></tr>
<tr><th colspan="2">User Rights</th></tr>
<tr><th>Cell</th><th>LDandCell</th></tr>
<tr><td>• 044N / . / Dial
Allows calls to cell phone numbers.
• 01N / . / Busy
Blocks calls to long distance numbers.
• ? / . / Busy
Block any other calls that require an authorization code.</td><td>• 044N / . / Dial
Allows calls to cell phone numbers.
• 01N / . / Dial
Allows calls to long distance numbers.
• ? / . / Busy
Block any other calls that require an authorization code.</td></tr>
</table>

It is recommended to use short codes that use X characters to match the full number of characters to be dialed. That ensures that authorization code entry is not triggered until the full number has been dialed rather than mid-dialing. For example 09 numbers are premium rate in the UK, so you would create a 09XXXXXXXXX / N / Dial short code set to Forced Authorization. In the associated user or user right short code it is recommended to use 09N type short codes.

For IP Office 4.0+ systems, system short codes that route to ARS will not have their Force Authorization Code setting used. However short codes within an ARS table will have their Force Authorization Code setting used.

- WARNING: Changing PC Registry Settings
  Avaya accepts no liability for any issues arising from the editing of a PC's registry settings. If you are in any doubt about how to perform this process you should not proceed. It is your responsibility to ensure that the registry is correctly backed up before any changes are made.

Enabling Authorization Codes in Manager

To enable support for authorization codes within Manager requires a change to the Manager PC registry settings. Once this change is made, various authorization code related features are visible when Manager is restarted and a configuration from a 3.2 or higher system is loaded.

1. Close Manager.
2. Locate the registry key *HKEY_CURRENT_USER\Software\Avaya\IP400\Manager\EnableAuthorisationCodes* and change its value from *0* to *1*.
3. Restart Manager and load a configuration from an IP Office 3.2 or higher system.

Enabling Authorization Codes in Delta Server

The use of authorization codes can be included in the SMDR output logged by the Delta Server application. Again this requires changes to the registry of the PC running the Delta Server application.

1. Open the registry and locate the *HKEY_LOCAL_MACHINE\Software\Avaya\CCCServer\Setup* registry keys.
2. Add two new DWORD registry keys and set their values to 1. They are:
   - AllowAuthorization
   - ShowAllowAuthorization
3. Open the browser to the Delta Server configuration screens.
4. Select SMDR. An Add Authorization Fields to SMDR option should now be available. Select this to enable logging of authorization codes to the SMDR log file.

Authorization codes are only logged to the SMDR log file. Two new fields are added to the end of each call log record in the SMDR log file. The first new field is the authorization code used or n/a if no authorization code was used. The second field is 1 for valid authorization or 0 for invalid authorization.

Forcing Authorization Codes

There are two methods to force a user to enter an authorization code in order to complete dialing an external call.

- **To Force Authorization Codes on All External Calls**
  A user can be required to enter an authorization code for all external calls. This is done by selecting Force Authorization Code 276 (User | Telephony | Supervisor Settings 276).
- **To Force Authorization Codes on Specific Calls**
  To require entry of an authorization code on a particular call or call type, the Force Authorization Code option should be selected in the short code settings. This can be used in user, user rights or system short codes in order to apply its effect to a user, group of users or all users respectively. You need to ensure that the user cannot dial the same number by any other method that would by pass the short code, for example with a different prefix.

Entering an Authorization Code

Where possible, when an authorization code is required, the user can enter it through their phones display. However this is not possible for all type of phone, for example it is not possible with analog phones and Avaya XX01 or XX02 phones. The users of these device must either enter the authorization code using Phone Manager or by using a short code set to the Set Authorization Code feature immediately before making the call.

When entry of an authorization code is triggered, the user can enter any authorization code with which they are either directly associated or associated through their current user rights.

Note

1. If account code entry is setup for a particular number, calls forwarded or transferred to that number will also trigger account code entry.
2. On systems using line appearances to BRI trunk channels to make outgoing calls, account code entry may not be triggered. This can be resolved by adding a short code such as [9]XN;/Dial/XN/0 (adjust the prefix and line group as necessary).

Authorization Code | Authorization Code	
Control Unit	SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	3.2+.
Mergeable	✓.

- Authorization Code: *Range = Up to 12 digits.*
  The digits used for the authorization code. Each code must be unique. Wildcards are not useable with authorization codes.
  - User Rights
    This field is used to select the user right with which the authorization code is associated. The authorization code can then be used to authorize calls made by users currently associated with that set of user rights.
  - User
    This field is used to select a user with which the authorization code is associated. The authorization code can then be used to authorize calls made by that user.

5.28 ARS

ARS (Alternate Route Selection) is used by IP Office 4.0+. It replaces LCR (Least Cost Routing) used by previous releases of IP Office. It also replaces the need to keep outgoing call routing short codes in the system short codes.

When a dialed number matches a short code that specifies that the number should be dialled, there are two methods by which the routing of the outgoing call can be controlled.

- **Routing Calls Directly to a Line**
  Every line and channel belongs has an Outgoing Group ID setting. Several lines and channels can have belong to the same Outgoing Group ID. Within short codes that should be routed via a line within that group, the required Outgoing Group ID is specified in the short code's Line Group ID setting.

- **Routing Calls via ARS**
  The short code for a number can specify an ARS form as the destination. The final routing of the call is then controlled by the setting available within that ARS form.

ARS Features

- **Secondary Dial Tone**
  The first ARS form to which a call is routed can specify whether the caller should receive secondary dial tone.

- **Out of Service Routing**
  ARS forms can be taken out of service, rerouting any calls to an alternate ARS form while out of service. This can be done through the configuration or using short codes.

- **Out of Hours Routing**
  ARS forms can reroute calls to an alternate ARS form outside the hours defined by an associated time profile.

- **Priority Routing**
  Alternate routes can be made available to users with sufficient priority if the initial routes specified in an ARS form are not available. For users with insufficient priority, a delay is applied before the alternate routes become available.

- **Line Types**
  ARS can be used with all line types except Small Community Network (SCN) trunks.

  - IP Office 4.2+: A SIP line is treated as busy and can follow alternate routes based on the SIP line setting Call Initiation Timeout 233. Previously a SIP line was only seen as busy if all the configured channels were in use.

  - IP lines use the NoUser Source Number setting *H323SetupTimerNoLCR* to determine how long to wait for successful connection before treating the line as busy and following ARS alternate routing. For IP Office 4.2+, this is now set through the IP line option Call Initiation Timeout 221.

- **Small Community Network**
  Calls to SCN users are always routed using the appropriate SCN trunk. ARS can be configured for SCN numbers but will only be used if the SCN call fails due to congestion or network failure. However if the SCN becomes available before ARS finds an alternate route then SCN is used. This also applies to calls made across the SCN using the Break Out 441 feature.

- **Main Route**
  The ARS form 50, named "Main" cannot be deleted. For defaulted systems it is used as a default route for outgoing calls.

Routing Calls to ARS

1. Create the ARS form.

2. Create the required system, user or user rights short code to match the user dialing.

   2.1. In the Telephone Number field define the digits that will be used to match a short code in the ARS form.

   2.2. Use the Line Group ID field drop-down to select the ARS form required for routing the call.

Example ARS Operation

The simplest example for ARS operation are the settings applied to a defaulted system. These vary between MU-Law systems and A-LAW systems.

A-Law Systems

This set of defaults is applied to A-Law systems, typically supplied to locales other than North America. The defaults allow any dialing that does not match an internal number to be routed off-switch as follows:

1. **System Short Code - ? / . / Dial / 50:Main**
   The default system short code ? will match any dialing for which no other user, user rights or system short code match is found. This short code is set to route all the digits dialed to ARS form 50.

2. **ARS Form - 50:Main**
   This form contains just a single short code.

   - **? / . / Dial3K1 / 0**
     This short code matches any digits passed to the ARS form. It then dials the digits out on the first available line within line group 0 (the default outgoing line group for all lines).

MU-Law Systems

This set of defaults is applied to MU-Law systems, typically supplied to locales in North America. The defaults route any dialing prefixed with a 9 to the ARS and secondary dial tone.

1. **System Short Code - 9N / N / Dial / 50:Main**
   The default system short code 9N is used to match any dialing that is prefixed with a 9. It passes any digits following the 9 to ARS form 50.

2. **ARS Form - 50:Main**
   This form has secondary dial tone enabled. It contains a number of short codes which all pass any matching calls to the first available line within line group 0 (the default outgoing line group for all lines). Whilst all these short code route calls to the same destination, having them as separate items allows customization if required. The short codes are:

   - **11 / 911 / Dial Emergency / 0**
     This short code matches an user dialing 911 for emergency services.

   - **911 / 911 / Dial Emergency / 0**
     This short code matches an user dialing 9911 for emergency services.

   - **0N; / 0N / Dial 3K1 / 0**
     This short code matches any international calls.

   - **1N; / 1N / Dial 3K1 / 0**
     This short code matches any national calls.

   - **XN; / N / Dial 3K1 / 0**
     This short code matches 7 digit local numbers.

   - **XXXXXXXXXX / N / Dial 3K1 / 0**
     This short code matches 10 digit local numbers.

5.28.1 ARS

Each ARS form contains short codes which are used to match the result of the short code that triggered use of the ARS form, ie. the Telephone Number resulting from the short code is used rather than the original number dialed by the user.

ARS | ARS	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	4.0+.
Mergeable	✓.

- ARS Route ID
  This value is automatically assigned and cannot be edited.
- Route Name: *Default = Blank, Range = Up to 15 characters.*
  The name is used for reference and is displayed in other areas when selecting which ARS to use.
- Dial Delay Time: *Default = System. Range = 1 to 30 seconds.*
  This settings defines how long ARS should wait for further dialing digits before assuming that dialing is complete and looking for a short code match against the ARS form short codes. When set to *System*, the system's Dial Delay Time 160 (System | Telephony | Telephony 160) value is used.
- Secondary Dial Tone: *Defaults = Off.*
  When on, this setting instructs the system to play secondary dial tone to the user. The tone used is set by the field below.
  - The tone used is set as either *System Tone* (normal dial tone) or *Network Tone* (secondary dial tone). Both tone types are generated by the system in accordance with the system specific locale setting 844. Note that in some locales normal dial tone and secondary dial tone are the same.
  - When Secondary Dial Tone is selected, the ARS form will return tone until it receives digits with which it can begin short code matching. Those digits can be the result of user dialing or digits passed by the short code which invoked the ARS form. For example with the following system short codes:
    - In this example, the 9 is stripped from the dialed number and is not part of the telephone number passed to the ARS form. So in this case secondary dial tone is given until the user dials another digit or dialing times out.
      - Code: 9N
      - Telephone Number: N
      - Line Group ID: 50 Main
    - In this example, the dialed 9 is included in the telephone number passed to the ARS form. This will inhibit the use of secondary dial tone even if secondary dial tone is selected on the ARS form.
      - Code: 9N
      - Telephone Number: 9N
      - Line Group ID: 50 Main
- Check User Call Barring: *Default = Off*
  If enabled, the dialing user's Outgoing Call Bar setting and any user short codes set to the function Barred are checked to see whether they are appropriate and should be used to bar the call.
- In Service: *Default = On*
  This field is used to indicate whether the ARS form is in or out of service. When out of service, calls are rerouted to the ARS form selected in the Out of Service Route field.
  - Short codes can be used to take an ARS form in and out of service. This is done using the short code features Disable ARS Form 463 and Enable ARS Form 466 and entering the ARS Route ID as the short code Telephone Number value.
- Out of Service Route: *Default = None.*
  This is the alternate ARS form used to route calls when this ARS form is not in service.
- Time Profile: *Default = None.*
  Use of a ARS form can be controlled by an associate time profile. Outside the hours defined within the time profile, calls are rerouted to an alternate ARS form specified in the Out of Hours Route drop-down. Note that the Time Profile field cannot be set until an Out of Hours Route is selected.
- Out of Hours Route: *Default = None.*
  This is the alternate ARS form used to route calls outside the hours defined within the Time Profile 360 selected above.
- Short Codes
  Short codes 426 within the ARS form are matched against the "Telephone Number" output by the short code that routed the call to ARS. The system then looks for another match using the short codes with the ARS form.

- Only short codes using the following features are supported within ARS: *Dial*, *Dial Emergency*, *Dial Speech*, *Dial 56K*, *Dial64K*, *Dial3K1*, *DialVideo*, *DialV110*, *DialV120* and *Busy*.

- Multiple short codes with the same Code field can be entered so long as they have differing Telephone Number and or Line Group ID settings. In this case when a match occurs the system will use the first match that points to a route which is available.

* Alternate Route Priority: *Default = 3. Range = 1 (low) to 5 (high).*
  If the routes specified by this form are not available and an Alternate Route has been specified, that route will be used if the users priority is equal to or higher than the value set here. User priority is set through the User | User form and by default is *5*. If the users priority is lower than this value, the Alternate Route Wait Time is applied. This field is grayed out and not used if an ARS form has not been selected in the Alternate Route field.

  - If the caller's dialing matches a short code set to the *Barred* function, the call remains at that short code and is not escalated in any way.

* Alternate Route Wait Time: *Default = 30 seconds, Range = Off, 1 to 60 seconds.*
  If the routes specified by this form are not available and an Alternate Route has been specified, users with insufficient priority to use the alternate route immediately must wait for the period defined by this value. During the wait the user hears camp on tone. If during that period a route becomes available it is used. This field is grayed out and not used if an ARS form has not been selected in the Alternate Route field.

* Alternate Route: *Default = None.*
  This field is used when the route or routes specified by the short codes are not available. The routes it specifies are checked in addition to those in this ARS form and the first route to become available is used.

5.28.2 Cause Codes and ARS

ARS routing to digital trunks can be affected by signalling from the trunk. The response to cause codes received from the line is as follows:

- **Reroute with ARS**
  The following cause codes cause ARS to no longer target the line group (unless it is specified by an alternate ARS route).

Code	Cause Code
1	Unallocated Number.
2	No route to specific transit network / (5ESS) Calling party off hold.
3	No route to destination. / (5ESS) Calling party dropped while on hold.
4	Send special information tone/(NI-2) Vacant Code.
5	Misdialed trunk prefix.
8	Preemption / (NI-2) Prefix 0 dialed in error.
9	Preemption, cct reserved/ (NI-2) Prefix 1 dialed in error.
10	(NI-2) Prefix 1 not dialed.
11	(NI-2) Excessive digits received call proceeding.
22	Number Changed.
28	Invalid Format Number.
29	Facility Rejected.
50	Requested Facility Not Subscribed.
52	Outgoing calls barred.
57	Bearer Capability Not Authorized.
63	Service or Option Unavailable.
65	Bearer Capability Not Implemented.
66	Channel Type Not Implemented.
69	Requested Facility Not Implemented.
70	Only Restricted Digital Information Bearer Capability Is Available.
79	Service Or Option Not Implemented.
88	Incompatible.
91	Invalid Transit Network Selection.
95	Invalid Message.
96	Missing Mandatory IE.
97	Message Type Nonexistent Or Not Implemented.
98	Message Not Implemented.
99	Parameter Not Implemented.
100	Invalid IE Contents.
101	Msg Not Compatible.
111	Protocol Error.
127	Interworking Unspecified.

- **Stop ARS**
  The following cause codes stop ARS targeting completely.

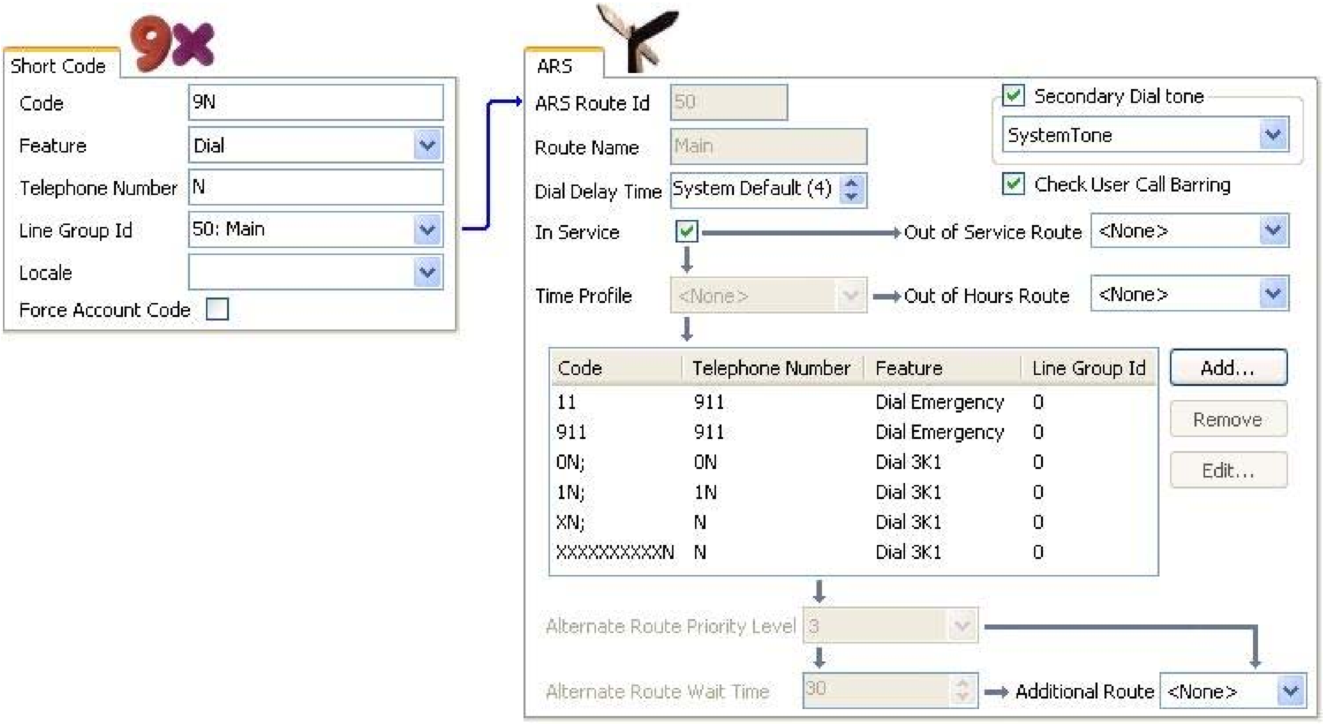
Code	Cause Code
17	Busy.
21	Call Rejected.
27	Destination Out of Order.

- **No Affect**
  All other cause codes do not affect ARS operation.

5.28.3 ARS Operation

The diagram below illustrates the default ARS routing applied to systems defaulted to the United States system locale. In summary:

- Any dialing prefixed with 9 will match the default system short code 9N.
- That short code routes calls to the default ARS form 50:Main.
- The short codes in that ARS form route all calls to an available line that has its **Outgoing Group ID** set to *0*.



The table describes in more detail the process that the system has applied to the user's dialing, in this example 91555707392200.

The user dials...	
9	• The **Dial Delay Count** is zero, so the system begins looking for short code matches in the system and user's short codes immediately. • Since there is only one match, the 9N system short code, it is used immediately. • The 9N short code is set to route the call to the ARS form **Main**. It only passes those digits that match the N part of the dialing, ie. the 9 is not passed to the ARS, only any further digits dialed by the user. • **Secondary Dial Tone** is selected in the ARS form. Since no digits for ARS short code matching have been received, secondary dial tone is played to the user.

1	• Having received some digits, the secondary dial tone stops.
	• The ARS form short codes are assessed for matches.
	  • The 11 and 1N; short codes are possible matches.
	  • The 911 and 0N; short codes are not possible matches.
	  • The XN; and XXXXXXXXXXN; short codes are also not matches because the 1N; short code is already a more exact match.
	• Since there is more than one possible match, the system waits for further digits to be dialed.

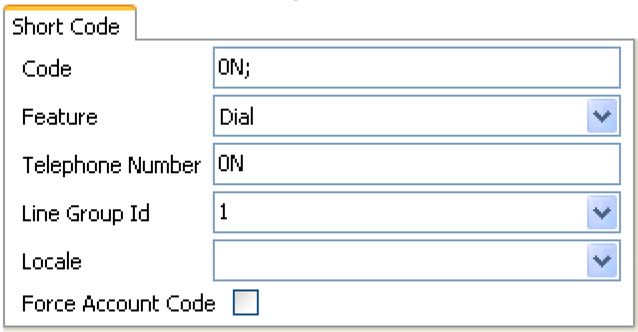
555	• The 11 short code is no longer a possible match. The only match is left is the 1N; short code. • The ; in the short code tells the system to wait for the **Dial Delay Time** to expire after the last digit it received before assuming that dialing has been completed. This is necessary for line providers that expect to receive all the routing digits for a call 'en bloc'. The user can also indicate they have completed dialing by pressing #.
707392200	• When the dialing is completed, a line that has its **Outgoing Group ID** set to *0* (the default for any line) is seized. • If no line is available, the alternate route settings would applied if they had been configured.

5.28.3.1 ARS Short Codes

The short codes in the default ARS form have the following roles:

Code	Feature	Telephone Number	Line Group ID	Description
11	Dial Emergency	911	0	These two short codes are used to route emergency calls. Note that the E911 [421] settings may override the routing to attempt to send the emergency call by a particular line if possible. However a Dial Emergency call is never blocked. If the required line is not available the system will use the first available line. Similarly calls using Dial Emergency ignore any outgoing call bar settings that would be normally applied to the user.
911	Dial Emergency	911	0	
0N;	Dial 3K1	0N	0	Matches international numbers.
1N;	Dial 3K1	1N	0	Matches national numbers.
XN;	Dial 3K1	N	0	Matches 7 digit local numbers.
XXXXXXXXXN;	Dial 3K1	N	0	Matches 10 digit local numbers.

ARS Short Code Settings



- Code
  The digits used for matching to the user dialing.
- Feature
  ARS short codes can use any of the *Dial* short code features or the *Barred* feature. When a *Barred* short code is matched, the call will not proced any further.
- Telephone Number
  The number that will be output to the line as the result of the short code being used as the match for the user dialing. Short code characters [427] can be used such as N to match any digits dialed for N or X in the Code.
- Line Group ID
  The line group from which a line should be seized once short code matching is completed. Another ARS form can also be specified as the destination.
- Locale
  Not used for outgoing external calls.
- Forced Account Code
  If enabled, the user will be prompted to enter a valid account code before the call can continue. The account code must match one set in the system configuration.

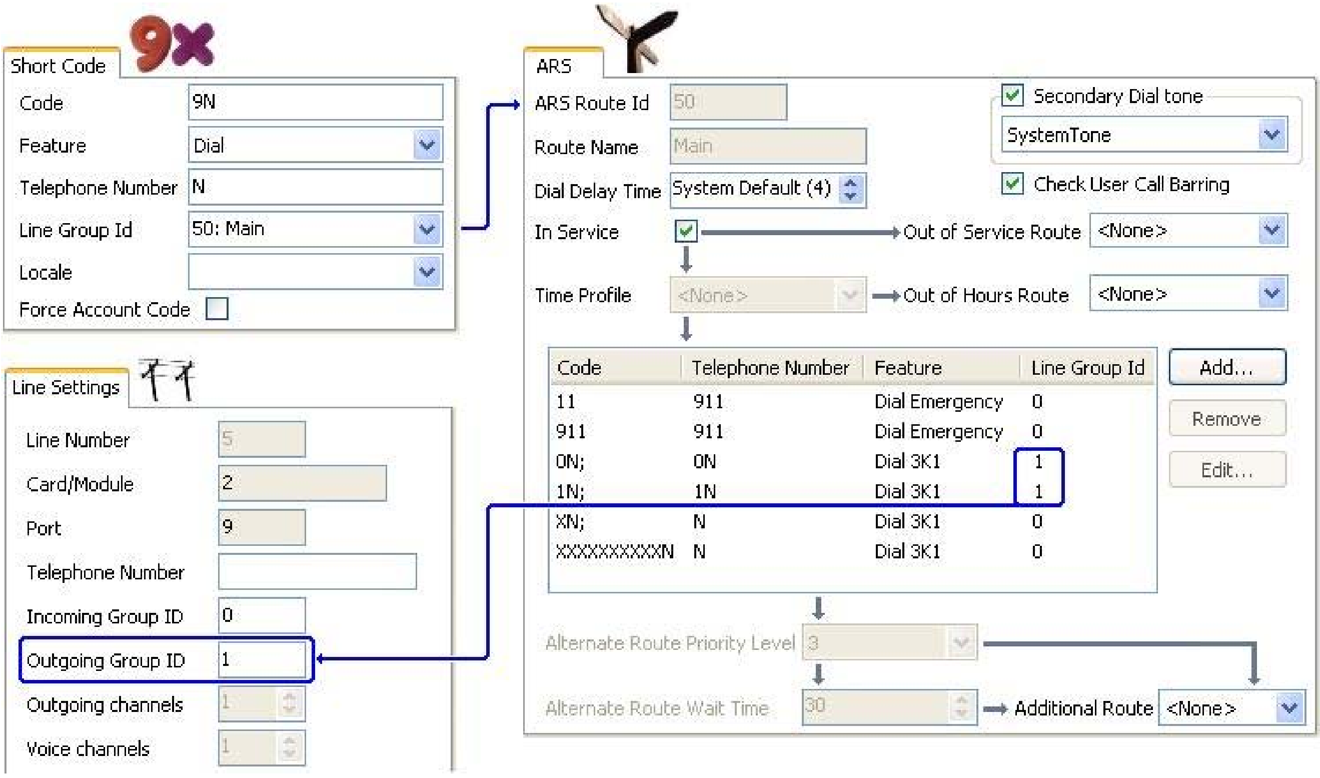
5.28.3.2 Simple Alternate Line Example

Using the default ARS settings 412, despite having several short codes in the ARS form, all outgoing calls are actually routed the same way using the same trunks. However, by having separate short codes for different call types present, it is easy to change the routing of each call type if required.

For this example, the customer has separate sets of lines for local calls and for national/international calls. These have been configured as follows:

- The lines for local and emergency calls have been left with the default **Outgoing Group ID** of *0*.
- The lines for national and international calls have been set with the **Outgoing Group ID** of *1*.

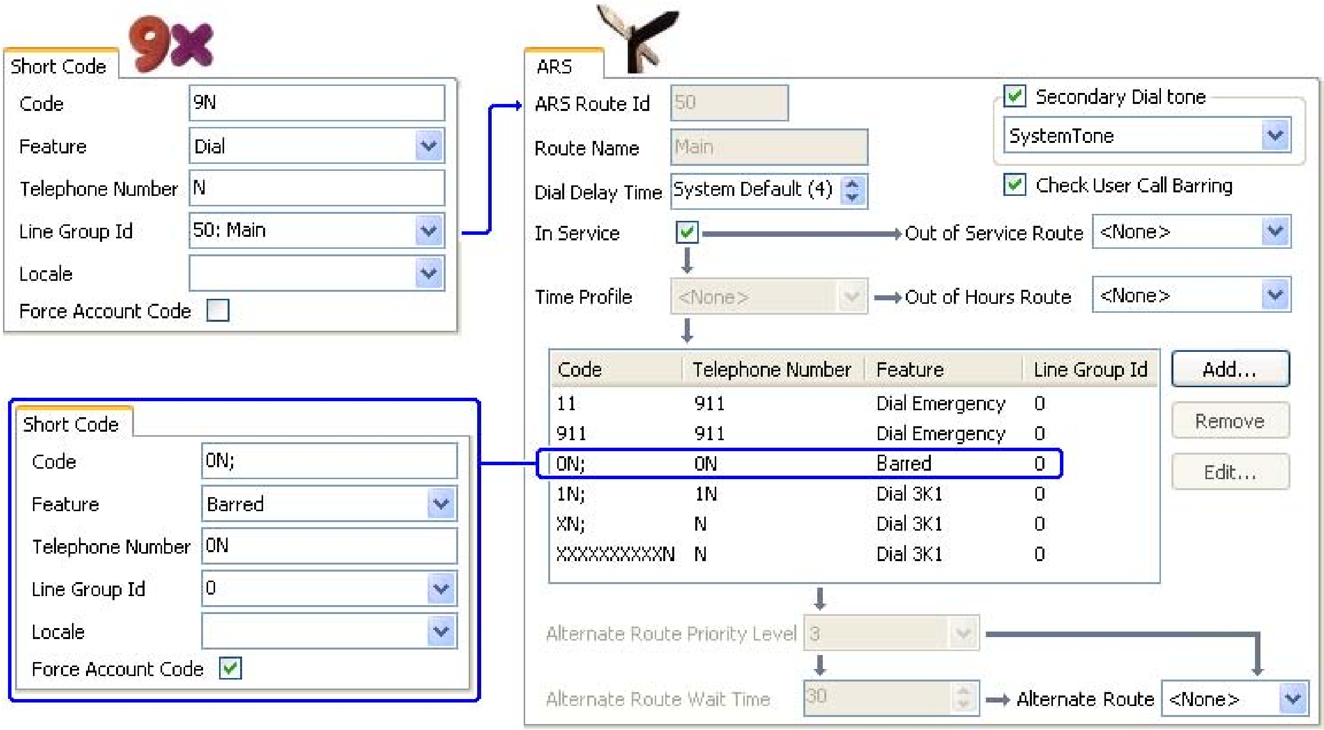
The default ARS can be configured to match this by just changing the **Line Group ID** settings of the default ARS short codes to match.



5.28.3.3 Simple Call Barring

All ARS short codes use one of the *Dial* short code features. The exception is the *Barred* short code feature. This can be selected for ARS short codes that match dialing that is not allowed.

In the example below, any user dialing an international number will be routed to the *Barred* short code. This prevents the dialing of external numbers prefixed with 0.

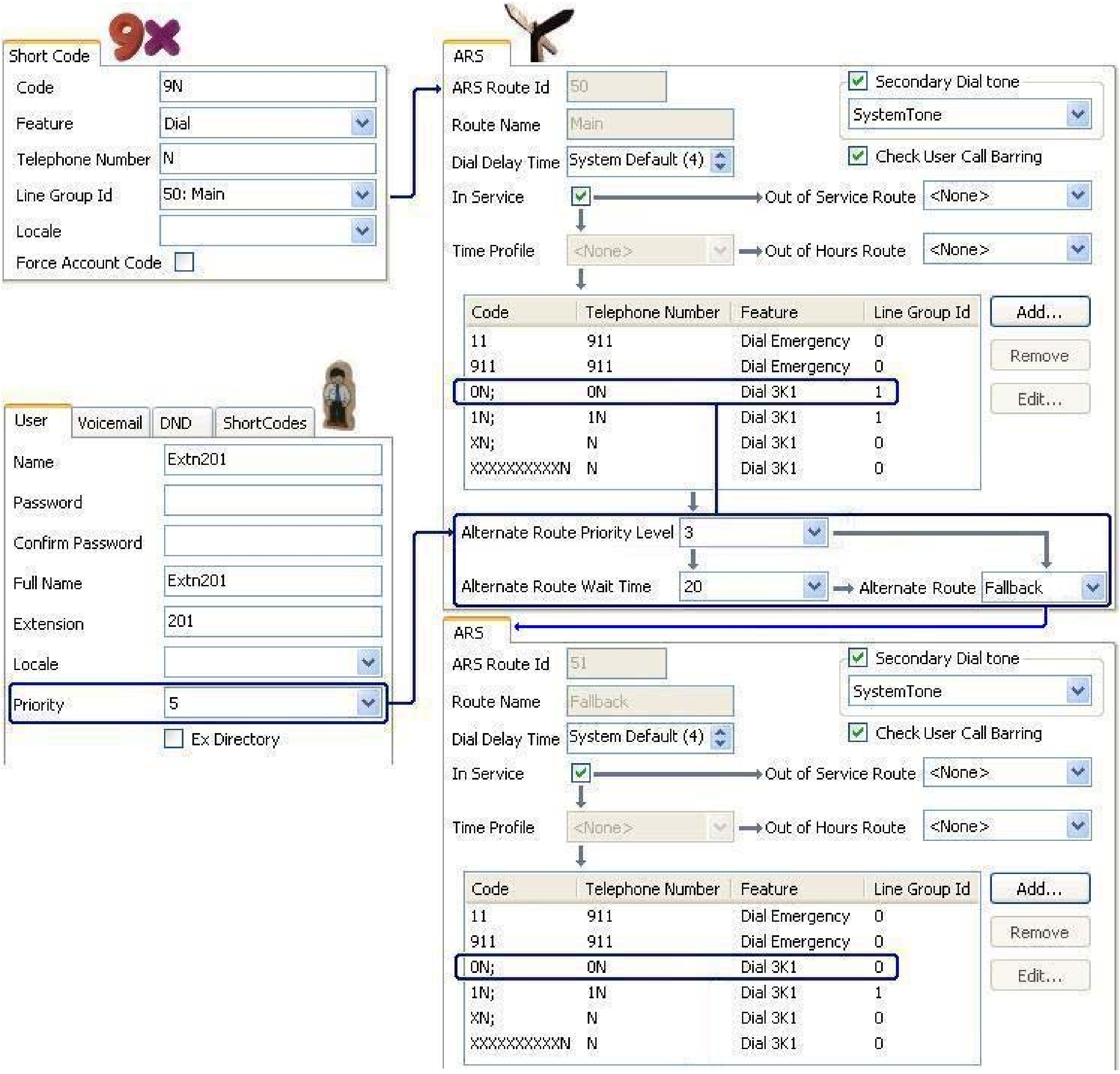


- To restrict a user from making any outgoing external calls, use the user's Outgoing Call Bar 276 option.

5.28.3.4 User Priority Escalation

User priority can be used to alter call routing when the required route is not available.

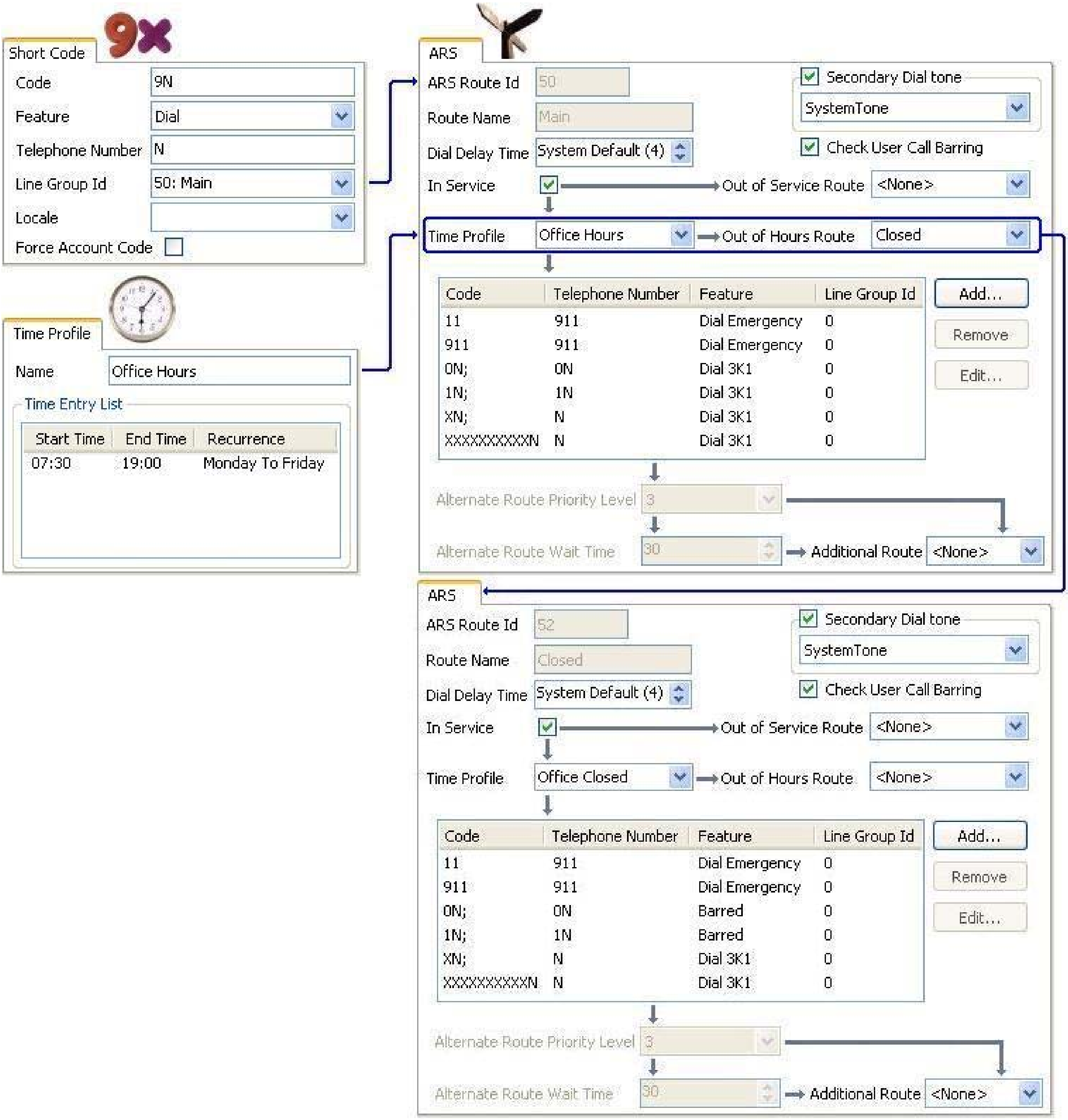
In this example, international calls are initially targeted to seize a line in outgoing line group 1. However an alternate route has been defined which will be used if no line in line group 1 is available. The fallback ARS form allows international calls to seize a line from line group 0. Whether this is done immediately or after a delay is set by whether the users priority is high enough.



5.28.3.5 Time Base Routing

Time profiles can be used to switch call routing from one ARS form to another.

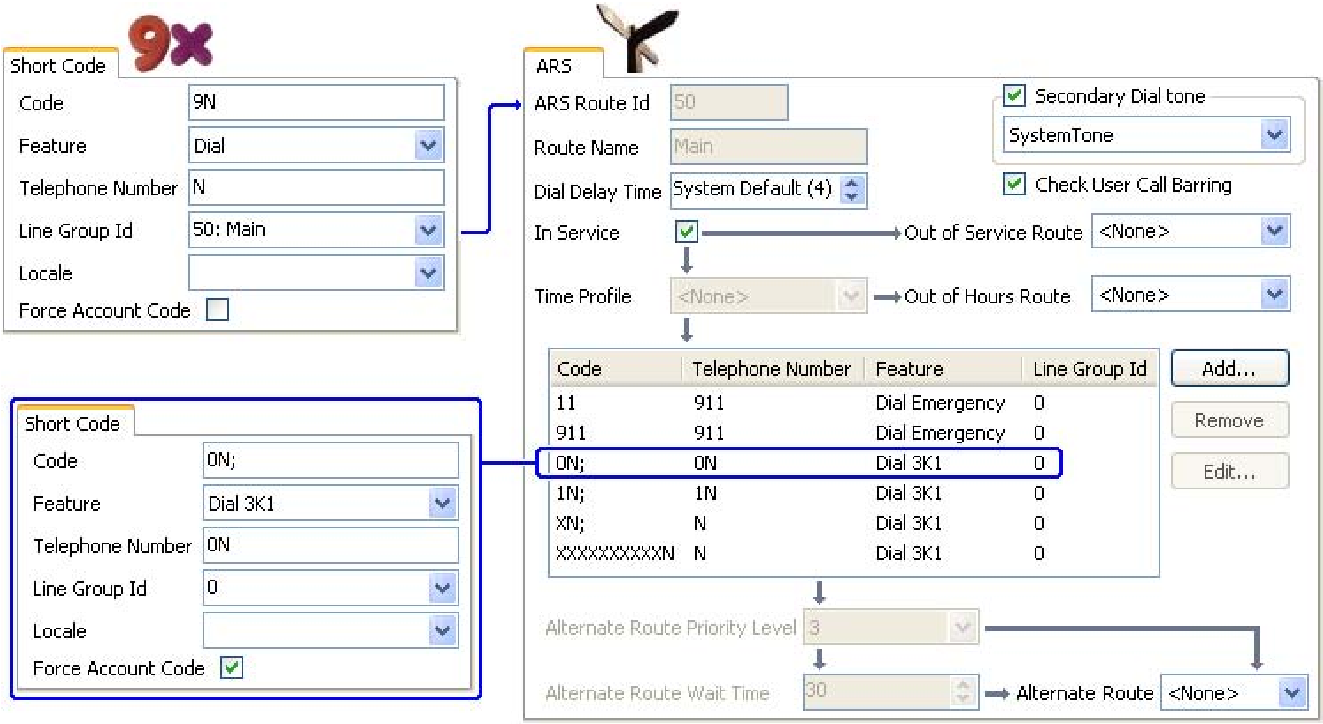
In the example below, a time profile has been define that sets the hours for normal operation. Outside the times set in the time profile, the other ARS form is used. This other ARS form only allows local and emergency calls.



5.28.3.6 Account Code Restriction

The short codes within an ARS form can be individually set to require an account code before allowing any call that matches it to proceed.

In the example below, the short code for international calls has been set to require the user to enter an account code. A valid account code must be dialed to continue with the call.



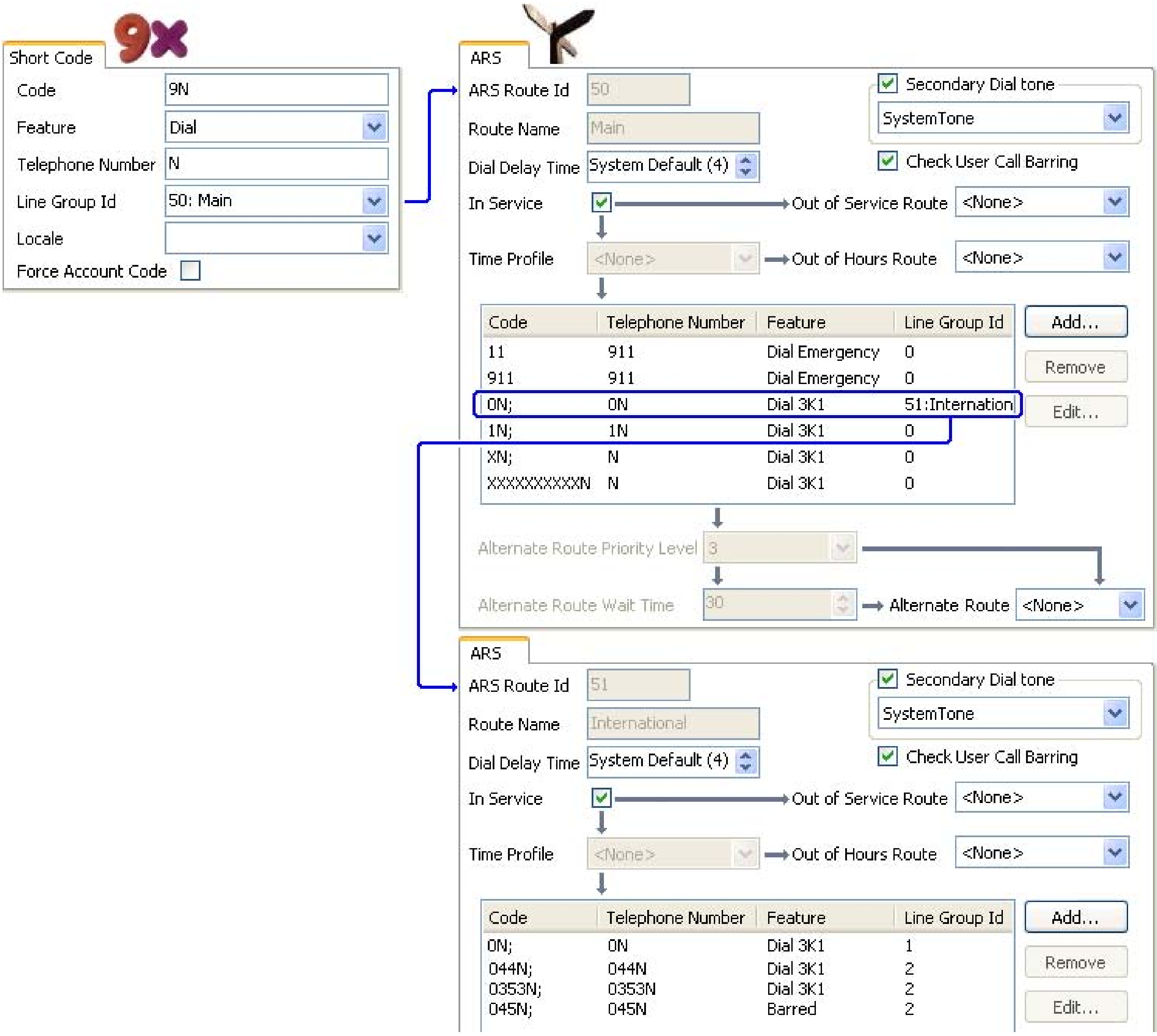
- If a user should always enter an account code to make any external call, the user option Force Account Code 276 should be used.
- When using account codes to restrict calls rather than just track calls it is recommended that Show Account Codes 160 is disabled. This stops the list of available account codes being shown by Phone Manager.

5.28.3.7 Tiered ARS Forms

It is possible for an ARS short code in one form to have another ARS form as its destination. Dialing that matches the short code is then subject to further matching against the short codes in the other ARS form.

In the example below, the user wants different routing applied to international calls based on the country code dialed. To do that in the default ARS form would introduce a large number of short codes in the one form, making maintenance difficult.

So the short code matching calls with the international dialing prefix 0 has been set to route matching calls to another ARS form. That form contains short codes for the different country dialing codes of interest plus a default for any others.



5.28.3.8 Planning ARS

Using the methods shown in the previous examples, it is possible to achieve ARS that meets most requirements. However the key to a good ARS implementation is planning.

A number of questions need to be assessed and answered to match the system's call routing to the customer's dialing.

- What
  What numbers will be dialed and what needs to be output by the system. What are the different call tariffs and the dialing codes.
- Where
  Where should calls be routed.
- Who
  Which users should be allowed to use the call routes determined by the previous questions.
- When
  When should outgoing external calls be allowed. Should barring be applied at any particular times? Does the routing of calls need to be adjusted for reasons such as time dependant call tariffs.

5.29 E911 System Settings

This section is only for Mu-law systems with their system Locale set to *United States*.

When the central office receives and emergency call, it routes the call to a dedicated emergency network. The call is then distributed to the correct emergency operator using either the automatic line identification information (ALI) received with the call or the registered billing address of the line on which the call was made.

E911 is system whereby emergency calls are routed through an E911 Adjunct which adds additional location information to the call based on the extension making the call. The use of E911 may be mandatory in some locations and may include the provision of an adjunct owned and managed by a third party or the central office.

Using Dial Emergency Short Codes

On all systems, regardless of locale; system and or ARS short codes using the Dial Emergency 458 feature should be created for any required emergency service numbers. Those short codes should be useable by all users from all extensions. Those short codes should route the calls to suitable lines. If the system uses prefixes for external dialing, the dialing of emergency numbers with and without the prefix should be allowed.

The blocking of emergency calls or the rerouting of emergency calls to a intermediate destination other than the central office may be against local and nation laws.

Emergency Dialing without an E911 Adjunct

The system E911 settings are only applicable for a system configured with an E911 Adjunct. If no adjunct is present, Dial Emergency short codes are used as described above.

Emergency Dialing with an E911 Adjunct

An E911 adjunct is an additional piece of equipment. It holds a database of location information based on the base extension number passed to it by the system. Connection from the system is by loop-start analog trunks. The adjunct is then connected to the central office's 911 emergency network by CAMA (Centralized Automatic Message Accounting) trunks.

When E911 is enabled and operating, all Dial Emergency calls are routed to the adjunct. The base extension ID is used by the adjuncts database to add the appropriate location information to the calls ALI before passing it to the emergency network.

- If Dial Emergency calls cannot be routed to the E911 adjunct due to congestion, the system will automatically fallback to using the E911 Zones.
- The E911 adjunct includes an alarm switch that can be connected to a system analog extension port. Dial Emergency calls will fallback to using E911 Zones if the E911 adjunct indicates an alarm.

If you are using E911 with an E911 Adjunct, the following needs to be administered:

1. Ensure that the appropriate Dial Emergency short codes have been configure either in system short codes or in an ARS form.
2. On the E911 Adjunct tab select Enable.
3. In the Adjunct Trunks list select those trunks that are connected to the E911 Adjunct.
4. In the Alarm Station field, indicate the extension number of the user to which the E911 adjunct's alarm connection has been connected.
    - On an operator or security extension, a programmable button can be set to monitor the status of the Alarm Station user. That button will then indicate in use (busy) when the E911 adjunct is in its alarm state.
5. E911 Zones should still be configured. These are used if the system cannot seize one of the Adjunct Trunks or the Alarm Station is busy (indicating a fault with the E911 adjunct connections).

5.29.1 E911 Adjunct

The settings on this tab relate to the use of an E911 adjunct if installed.

Dial Emergency calls routed to the E911 adjunct include the extension ID of the extension on the system. This is used to lookup information stored in a database on the E911 Adjunct and then send that information to the central office. The extension ID of each extension is shown on the Extension | Extn 252 tab.

- To call an extension using its Base Extension number, the short code feature Dial Physical Extn by ID 461 can be used. This is normally provided by the default short code ***71*N#* when *N* can be replaced by the required extension ID.

E911 System | E911 Adjunct Settings	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

- Enable: *Default = Off (Not Selected)*
  When selected, all Dial Emergency calls are routed via the selected Adjunct Trunks and includes the extension ID of the dialing extension.
- Alarm Station: *Default = Blank*
  This is an analog extension port on the system, connected to the alarm switch provided on the E911 adjunct. When the E911 adjunct detects an error, for example disconnection of the CAMA trunks, it will indicate this by taking the alarm connection off-hook, making the analog extension busy.
  - When an alarm occurs and is indicated to the system by the alarm station being busy, the system will reroute Dial Emergency calls using the E911 | Zones 423 settings.
  - To monitor the status of the alarm connection a user should be associated with the analog extension (one should exist by default). Another user such as the reception desk or security desk can then be provided with a programmable button set to the action User and the appropriate user. That button will then indicate busy when the alarm connection is in an alarm state.
- Adjunct Trunks:
  This section lists all the analog and digital trunks (except VoIP trunks) available on the system. Only the trunks which are connected to the E911 Adjunct should be selected.

5.29.2 Zones

Zones are used for Dial Emergency calls on US systems when a trunk to the E911 adjunct cannot be seized or the E911 adjunct is indicating an alarm through its alarm station being off-hook (busy).

Zones allow extensions in the same location to be grouped and to have the external trunks registered to the same location associated with them. This billing address of those trunks or other information held by the trunk provider can then be used identify the correct emergency center that should respond to Dial Emergency calls from extensions in that zone.

The extension number used for this function is the Base Extension number set through the Extension | Extn 252 tab. This allows the physical extension to be correctly identified even if currently being used by a hot desking user with a different associated extension number.

A zone named Default is maintained by the system. By default all extensions and lines are included in this zone. This zone can be edited but cannot be removed. Any extension included in another zone and then removed is automatically placed back into the Default zone. Trunks can be deleted as some trunks will not be public trunks suitable for emergency calls.

- To call an extension using its Base Extension number, the short code feature Dial Physical Extn by Number 461 can be used. This is normally provided by the default short code *70*N# when N can be replaced by the required Base Extension number.

E911 System | Zones	
Control Unit	SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓, IP500v2 ✓.
Software Level	1.0+.
Mergeable	✗.

This tab shows the existing E911 zones, the number of stations (extensions) and trunks in each zone and the base extension numbers of the extensions. The settings can be edited by clicking on an existing zone and selecting Edit. A new zone can be created by selecting Add.

- Name: *Default = Default, Range = Up to 15 characters.*
  Allows a unique name to be assigned to each zone.

- Stations: *Default = Contains all Extension ID's*
  Lists the extensions in the zone. Click Add to add and remove extensions. To just remove extensions, select the extension or extensions and click Remove. Extensions removed from any zone are automatically included in the Default zone and cannot be removed from that zone except by inclusion in another zone. Each extension can only be included in one zone.

- Trunks: *Default = Blank*
  List the trunks associated with the zone. Click Add to add and remove trunks. To just remove trunks, select the trunk or trunks and click Remove. Each trunk can only be included in one zone. Dial emergency calls will seize the first available trunk. If no available trunk can be seized, the call will fallback to using the routing indicated with the Line Group ID of the Dial Emergency short code and final to using any available analog or digital trunk.

Chapter 6.
Short Codes

6. Short Codes

The system uses short codes to match the number dialed to an action. The number dialed or part of the number dialed can be used as parameter for the feature.

- WARNING
  User dialing of emergency numbers must not be blocked by the addition of short codes. If short codes are added, the users ability to dial emergency numbers must be tested and maintained.

Examples
The method of detailing a short codes settings lists the short code fields separated by a /.

- *17/?U/VoicemailCollect
  A user dialing **17* is connected to voicemail.

- *14*N#/N/FollowMeTo
  If a user dials **14*210#* at their own extension, their calls are redirected to extension 210.

Dialing Short Codes
The following types of short code applied to on-switch dialing. The result may be an action to be performed by the system, a change to the user's settings or a number to be dialed. The order below is the order of priority in which they are used when applied to user dialing.

- User Short Codes
  These are useable by the specific user only.

- User Rights/User Restrictions Short Codes
  These are useable by any users associated with the user rights or restrictions in which they are set. They can be overridden by individual user short codes.

  - User and User Rights short codes are only applied to numbers dialed by that user. For example they are not applied to calls forwarded via the user.

- System Short Codes
  These are available to all users on the system. They can be overridden by user or user rights short codes.

Post-Dialing Short Codes
When any the short code above result in a number to be dialed, further short code can be applied to that number to be dialed. This is done using the following types of short codes.

- ARS (Alternate Route Selection) Short Codes
  The short code that matches dialing can specify that the resulting number should be passed to an ARS form. The ARS form can specify which routes should be used for the call by using further short code matches and also provide option to use other ARS forms based on other factors such as time and availability of routes. In IP Office 4.0+, ARS has replaced Least Cost Routes.

- Least Cost Route Short Codes
  Least Cost Routes are used by pre-4.0 systems only. On these systems any short code that results in a number to be dialed, has is telephone number further checked for a match against LCR short codes. Time profiles can be used to control when particular sets of LCR short codes are used.

- Transit Network Selection (TNS) Short Codes
  Used on T1 ISDN trunks set to use AT&T as the Provider. Applied to the digits presented following any other short code processing.

Incoming Number Short Codes
On certain types of trunks short codes can be applied to the incoming digits received with calls.

- Line Short Codes
  These short codes are used to translate incoming digits received with calls. The stage at which they are applied varies between different line types and may be overridden by an extension number match.

6.1 Short Code Fields and Characters

Each short code, regardless of its type, has the following fields:

- Short Code: *Default =Blank, Range = Up to 31 characters.*
  The digits which if matched trigger use of the short code. Characters can also be used to create short codes which cannot be dialed from a phone but can be dialed from application speed dials. However some characters have special meaning, see the table below.
- Telephone Number: *Default = Blank, Range = Up to 32 characters.*
  The number output by the short code. When necessary, this is used as parameter for the selected short code Feature. See the table below for the special characters that can be used here.
- Line Group ID: *Default = 0*
  This field is used for short codes that result in a number to be dialed. It acts as a drop-down from which either an outgoing line group. IP Office 4.0+: An ARS form can also be selected.
- Feature: *Default = Dial*
  This sets the action performed by the short code when used. See Short Code Features 439.
- Locale: *Default = Blank*
  Features that transfer the caller to voicemail can indicate the language locale required for prompts. This is subject to the language being supported and installed on the voicemail server.
- Force Account Code: *Default = Off*
  When selected, for short codes that result in the dialing of a number, the user is prompted to enter a valid account code before the call is allowed to continue.

Short Code Field Characters

- ? - Default Match
  This character can be used on its own to create a short code match in the absence of any other short code match. See ? Short Codes 433.
  - ?D - Default Number Dialing
    This character combination makes a call to the defined phone number as soon as the user goes off-hook. See ? Short Codes 433.
- N - Match Any Digits
  Matches any dialed digits (including none). The Dial Delay Time or a following matching character is used to resolve when dialing is complete.
- X - Match a Digit
  Matches a single digit. When a group of X's is used, the short code matches against the total number of X's.
- [ ] - Secondary Dial Tone Trigger
  For pre-4.0 IP Office systems used to trigger secondary dial tone. Not used for IP Office 4.0+. See Secondary Dial Tone 432.
- ; - Receive Sending Complete
  When used this must be the last character in the short code string. If the Dial Delay Count is 0, a ; instructs the system to wait for the number to be fully dialed, using the Dial Delay Time or the user dialing #, before acting on the short code. If the Dial Delay Count is non-zero, the dialing is only evaluated when # is pressed.
  - The majority of North-American telephony services use en-bloc dialing. Therefore the use of a ; is recommended at the end of all dialing short codes that use an N. This is also recommended for all dialing where secondary dial tone short codes are being used.

Telephone Number Field Characters

- A - Allow Outgoing CLI
Allow the calling party number sent with the call to be used. This character may be required by service providers in some locales.

- C - Use Called Number Field
Place any following digits in the outgoing call's Called number field rather than the Keypad field.

- D - Wait for Connect
Wait for a connect message before sending any following digits as DTMF.

- E - Extension Number
Replace with the extension number of the dialing user. Note that if a call is forwarded this will be the replaced with the extension number of the forwarding user.

- h - Hold Music Source *(IP Office 6.1+)*
When used as part of the short code telephone number field, this character allows the source for music on hold to be selected. Enter *h(X)* where *X* is 1 to 4 indicating the required hold music source 163 if available. This overrides any previous hold music selection that may have been applied to the call. When used with ParkCall shortcodes, the *h(X)* should be entered before the park slot number part of the telephone number.

- I - Use Information Packet
Send data in an Information Packet rather than Set-up Packet.

- K - Use Keypad Field
Place any following digits in the outgoing call's Keypad field rather than the Called Number field. Only supported on ISDN and QSIG.

- l - Last Number Dialed *(lower case L)*
Use the last number dialed.

- L - Last Number Received
Use the last number received.

- N - Dialed Digit Wildcard Match
Substitute with the digits used for the N or X character match in the Short Code number field.

- p - Priority *(IP Office 4.2+)*
The priority 344 of a call is normally assigned by the Incoming Call Route or else is *1-Low* for all other calls. Dial Extn 458 short codes can use p(*x*) as a suffix to the Telephone Number to change the priority of a call. Allowable values for *x* are *1*, *2* or *3* for low, medium or high priority respectively.
- In situations where calls are queued, high priority calls are placed before calls of a lower priority. This has a number of effects:

  - Mixing calls of different priority is not recommended for destinations where Voicemail Pro is being used to provided queue ETA and queue position messages to callers since those values will no longer be accurate when a higher priority call is placed into the queue. Note also that Voicemail Pro will not allow a value already announced to an existing caller to increase.

  - If the addition of a higher priority call causes the queue length to exceed the hunt group's Queue Length Limit 316, the limit is temporarily raised by 1. This means that calls already queued are not rerouted by the addition of a higher priority call into the queue.

- S - Calling Number
Place any following digits into the outgoing call's calling number field. Using S does not alter any allow or withhold CLI setting associated with the call, the short code characters A or W should be used respectively.
  - Outgoing CLI Warning
Changing the outgoing CLI for calls requires the line provider to support that function. You must consult with your line provider before attempting to change the outgoing CLI, failure to do so may result in loss of service. If changing the outgoing CLI is allowed, most line providers required that the outgoing CLI used matches a number valid for return calls on the same trunks. Use of any other number may cause calls to be dropped or the outgoing CLI to be replaced with a valid number.

  - On mobile twinned calls, if the original party information is used or a specific calling party information CLI is set, that number overrides setting the outgoing CLI using short codes.

- SS - Pass Through Calling Number
Pass through the Calling Party Number. For example, to provide the incoming ICLID at the far end of a VoIP connection, a short code ? with telephone number .SS should be added to the IP line.

- i - National
Both the *S* and *SS* characters can be followed by an *i*, that is *Si* and *SSi*. Doing this sets the calling party number plan to ISDN and number type to National. This may be required for some network providers.

- t - Allowed Call Duration
Set the maximum duration in minutes for a call plus or minus a minute. Follow the character with the number of minutes in brackets, for example *t(5)*.

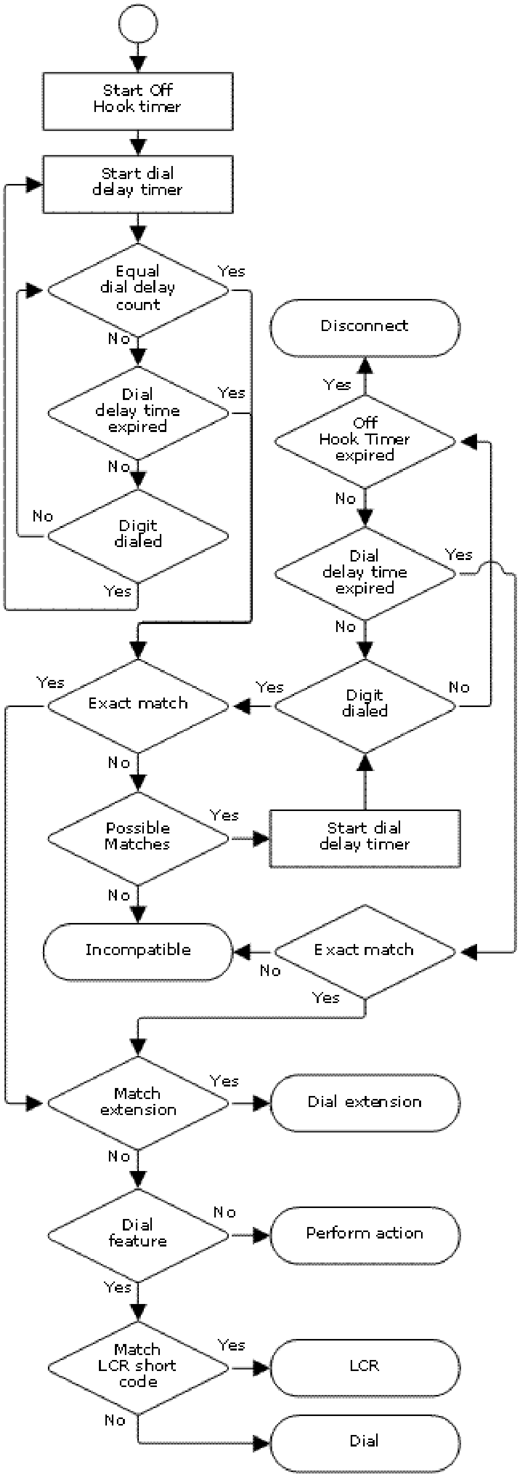
- U - User Name
Replace with the User Name of the dialing user. Used with voicemail.

- W - Withold Outgoing CLI
Withhold the sending of calling ID number. Operation is service provider dependent.

- Y - Wait for Call Progress Message
  Wait for a Call Progress or Call Proceeding message before sending any following digits as DTMF. For example, the Y character would be necessary at a site where they have signed up with their telephone service provider to withhold international dialing until a DTMF pin/account number is entered that initiates the call progress/proceeding message.
- Z - Calling Party Name *(IP Office Release 6.0+)*
  This option can be used with trunks that support the sending of name information. The Z character should be followed by the name enclosed in " " quotation marks. Note that their may be name length restrictions that vary between line providers. The changing of name information on calls being forwarded or twinned may also not be supported by the line provider.
- @ - Use Sub Address Field
  Enter any following digits into the sub-address field.
- . - Dialed Digits
  Replace with the full set of dialed digits that triggered the short code match.
- , - One Second Pause
  Add a one second pause in DTMF dialing.
- " " - Non Short Code Characters
  Use to enclose any characters that should not be interpreted as possible short code special characters by the system. For example characters being passed to the voicemail server.

6.2 User Dialing

Summary: Looks at how the system looks for possible short code matches to user dialing.



The following system settings influence user dialing.

- Dial Delay Count: *Default = 0 (US/Japan), 4 (ROW)*
  This value sets the number of digits dialed before the system looks for a short code match.
- Dial Delay Time: *Default = 4000ms (US/Japan), 1000ms (ROW)*
  This value sets the maximum allowed interval between the dialing of each digit. If exceeded, the system looks for a short code match even if the Dial Delay Count has not been reached.
- Off-Hook Timer: *Length Fixed by locale.*
  When a user goes off-hook, the system starts a 30 second off-hook timer (10 seconds in Italy). If the off-hook timer expires before a short code match occurs, the user is disconnected.

The following rules are used when short code matching is performed for user dialing:

- A short code is used immediately an exact match is found unless followed by a ;.
- If no match is found but partial matches exist, the user can continue dialing.
- If no match or partial matches are found, incompatible is returned.
- The following precedence is used to determine which short codes are used:
- Extension number matches override all short codes.
- User short codes override user rights and system short codes.
- User Rights short code matches override system short codes.
- When multiple exact matches occur,
  - The match with the most specified digits rather than wildcards is used.
  - If there are still more than one match, the match with the most exact length is used. This means X wildcards will override N when both match.

6.3 Application Dialing

Numbers speed dialed by system applications such as Phone Manager and SoftConsole are treated differently. Since the digits are received en bloc as a single group, they can override some short code matches. The same applies to short codes used within system configuration settings such as Incoming Call Route destinations.

Example:

- Telephone Number: 12345678
- Short Code 1: 1234XX/207/DialExtn
- Short Code 2: 12345678/210/DialExtn

If dialed manually by the user, as soon as they have dialed 123456 a match to short code 1 occurs. They can never dial short code 2.

If dial using a Phone Manager speed dial, 12345678 is sent as a string and a match to short code 2 occurs.

Partial Dialing
If the application dialing does not trigger an exact match, the user can dial additional digits through their extension. The processes for normal user dialing are applied.

Non-Digit Short Codes
Short codes can be created that use alphabetic characters instead of numbers. While these short codes cannot be dialed from a phone, they can be dialed using application speed dials and settings. However characters that are interpreted as special short code characters will still be interpreted as such.

6.4 Secondary Dial Tone

Some locales prefer to provide users with secondary dial tone once they have started dialing external calls. This dial tone is heard by the user until they have completed dialing and a trunk is seized at which point call progress tones are provided by the trunk, or camp on/busy tone is provided by the system if the required trunk cannot be seized.

IP Office 4.0 and Higher

The use of secondary dial tone is provided through the Secondary Dial Tone check box option on the ARS 407 form to which the call is routed. When on, this setting instructs the Manager to play secondary dial tone to the user.

- The tone used is set as either *System Tone* (normal dial tone) or *Network Tone* (secondary dial tone). Both tone types are generated by the system in accordance with the system specific locale setting 844. Note that in some locales normal dial tone and secondary dial tone are the same.
- When Secondary Dial Tone is selected, the ARS form will return tone until it receives digits with which it can begin short code matching. Those digits can be the result of user dialing or digits passed by the short code which invoked the ARS form. For example with the following system short codes:
    - In this example, the 9 is stripped from the dialed number and is not part of the telephone number passed to the ARS form. So in this case secondary dial tone is given until the user dials another digit or dialing times out.
        - Code: 9N
        - Telephone Number: N
        - Line Group ID: 50 Main
    - In this example, the dialed 9 is included in the telephone number passed to the ARS form. This will inhibit the use of secondary dial tone even if secondary dial tone is selected on the ARS form.
        - Code: 9N
        - Telephone Number: 9N
        - Line Group ID: 50 Main

Pre-4.0 IP Office Secondary Dial Tone

Pre-4.0 systems provided dial tone through the use of the short code feature Secondary Dial Tone and the [ ] special characters. For example, on a system where 9 is used as a prefix for external dialing, the system short code 9/./ Secondary Dial Tone/0 will trigger secondary dial tone when users dial a number prefixed with 9. This method is not supported by IP Office 4.0 which provides ARS forms for the control of outgoing calls.

In order to allow further digit matching, the digits dialed are put back through short code matching against any short codes that start with [n] where n is the digit used to trigger the system secondary dial tone short code.

- On all systems where secondary dial tone is used, a ; should also be used in dialing short codes that contain N.

For example:

- System Short Codes
    - 9/./SecondaryDialTone
    - [9]0N;/Dial/0
- User Short Code
    - [9]0N;/Busy/0

The user dials 90114445551234. The 9 is matches the system secondary dial tone short code and unlike other short codes this is applied immediately. The user's dialing is put through short code matching again using the normal order of precedence but matched to possible short codes beginning [9]. In this case the user's [9]0N; short code would take precedence over the system [9]0N; short code.

6.5 ? Short Codes

The ? character can be used in short codes in the following ways:

- **Default Short Code Matching**
  ? short codes are used in short code matching in the following way. If no user or system short code match is found, the system will then look for a ? short code match. It will look first for a user ? short code and then, if not found, a system ? short code.
  - Example: On systems outside North America, the system short code *?/./Dial/0* is added as a default short code. This short code provides a match for any dialing to which there is no other match. Therefore, on systems with this short code, the default is that any unrecognized number will be dialed to Outgoing Line Group 0.
- **Hot-Line Dialing**
  A user short code *?D* can be used to perform a short code action immediately the user extension goes off-hook. This is supported with Dial type short code features (except **Dial Direct Hotline**). Typically it is used with door, lift and lobby phones to immediately connect the phone to a number such as the operator or reception.
- **Voicemail Collect Short Codes**
  The ? character can appear in the **Telephone Number** field of a short code. This is done with short codes using the VoicemailCollect 501 feature. In this instance the ? character is not interpreted by the system, it is used by the voicemail server.

6.6 Short Code Matching Examples

The following examples are not meant as practical examples. However they are simple to implement and test on real system without conflicting with its normal operation. They illustrate the interaction between different short codes in resolving which short code is an exact match. They assume that extension numbers are in the 200 to 299 range.

The term 'dials' means dialing the indicated digit or digits without the inter-digit Dial Delay Time expiring.

The term 'pause' means a wait that exceeds the inter-digit Dial Delay Time.

Scenario 1		
Short Code 1 = 60/203/DialExtn. Dial Delay Count = 0. Dial Delay Time = 4 seconds.		
Test	**Dialing**	**Effect**
1	8	No possible match, incompatible returned immediately
2	6	No exact match but there is a potential match, so the system waits. When the Dial Delay Time expires, no exact match is found so incompatible is returned.
3	60	Exact match to Short Code 1. Extension 203 called immediately.
4	61	No possible match, the system returns incompatible.

Scenario 2		
Short Code 1 = 60/203/DialExtn. Short Code 2 = 601/210/Dial Extn. Dial Delay Count = 0. Dial Delay Time = 4 seconds.		
Test	**Dialing**	**Effect**
1	8	No possible match, incompatible returned immediately
2	60	Exact match to Short Code 1. Extension 203 called immediately.
3	601	Exact match to Short Code 1 as soon as the 0 is dialed. The user cannot manually dial 601. The only way they can dial 601 is using a Phone Manager speed dial set to dial that string.

Scenario 3		
Short Code 1 = 60/203/DialExtn. Short Code 2 = 601/210/Dial Extn. Dial Delay Count = 3. Dial Delay Time = 4 seconds.		
Test	**Dialing**	**Effect**
1	8	Insufficient digits to trigger matching. The system waits for additional digits or for Dial Delay Time to expire. When Dial Delay Time expires, no possible match is found so incompatible is returned.
2	60	Insufficient digits to trigger matching. The system waits for additional digits or for Dial Delay Time to expire. When Dial Delay Time expires, matching started and exact match to Short Code 1 occurs. .
3	601	Third digit triggers matching. Exact match to Short Code 2. Extension 210 dialed immediately.
4	60#	# is treated as a digit and as the third digit triggers matching. No exact match found. The system returns incompatible.

Scenario 4		
Short Code 1 = 60;/203/DialExtn. Short Code 2 = 601/210/Dial Extn. Dial Delay Count = 3. Dial Delay Time = 4 seconds.		
Test	**Dialing**	**Effect**
1	8	Insufficient digits to trigger matching. The system waits for additional digits or for Dial Delay Time to expire. When Dial Delay Time expires, no possible match is found so incompatible is returned.
2	6	Insufficient digits to trigger matching. The system waits for additional digits or for the interdigit Dial Delay Time to expire. If the Dial Delay Time expires, a potential match exists to a short code that uses ; so the system waits for an additional digit until the off-hook timer expires.
3	60	As above but an additional digit now may create a match. If 1 is dialed, it creates an exact match to Short Code 2 and is used immediately. If 0, * or 2 to 9 is dialed, no possible match exists. The system returns incompatible. If the next digit is a #, it is treated as signaling dialing complete rather than being a digit. Short code 1 becomes an exact match and is used immediately.
4	601	Third digit triggers matching. Exact match to Short Code 2. Extension 210 dialed immediately.

Scenario 5		
Short Code 1 = 601/203/DialExtn. Short Code 2 = 60N/210/Dial Extn. Dial Delay Count = 0. Dial Delay Time = 4 seconds.		
Test	**Dialing**	**Effect**
1	6	No exact match but there is a potential match, so the system waits for additional dialing. If the Dial Delay Time expires, no exact match is found so incompatible is returned.
2	60	Potential match to both short codes. The system waits for additional dialing. If the Dial Delay Time expires, Short Code 2 becomes an exact match with N blank.
3	601	Exact match to Short Code 1. Used immediately
4	602	Exact match to Short Code 2. Used immediately.

Scenario 6		
Short Code 1 = 601/203/DialExtn. Short Code 2 = 60N/210/Dial Extn. Short Code 3 = 60X/207/DialExtn. Dial Delay Count = 0. Dial Delay Time = 4 seconds.		
Test	**Dialing**	**Effect**
1	6	No exact match but there are potential matches so the system waits for additional dialing. If the Dial Delay Time expires, no exact match has occurred so incompatible is returned.
2	60	Potential match to all short codes. System waits for additional dialing. If the Dial Delay Time expires, Short Code 2 becomes an exact match with N blank. If a digit is dialed, Short Code 3 becomes a more exact match and is used.
3	601	Exact match all short code, however Short Code 1 is treated as being more exact (more matching digits) and is used immediately
4	602	Exact match to short codes 2 and 3, however the Short Code 3 is treated as being more exact (length match) and is used immediately.

Scenario 7		
Short Code 1 = 601/203/DialExtn. Short Code 2 = 60N/210/Dial Extn. Short Code 3 = 6XX/207/DialExtn. Dial Delay Count = 0. Dial Delay Time = 4 seconds.		
Test	**Dialing**	**Effect**
1	6	No exact match but there are potential matches so the system waits for additional dialing. If the Dial Delay Time expires, no exact match has occurred so incompatible is returned.
2	60	Potential match to all short codes. System waits for additional dialing. If the Dial Delay Time expires, Short Code 2 becomes an exact match with N blank. If a digit is dialed, Short Code 3 becomes an more exact match and is used.
3	601	Exact match all short code, however Short Code 1 is treated as being more exact (more matching digits) and is used immediately
4	602	Exact match to short codes 2 and 3, however the Short Code 2 is treated as being more exact (more matching digits) and is used immediately.
5	612	Exact match to Short Code 3.

6.7 Default System Short Code List

Most control units are available in A-Law and MU-Law models. Typically MU-Law models are supplied to North American locales, A-Law models are supplied to the rest of the world. In addition to the using different default companding for digital lines and phone, A-Law and MU-Law models support different default short codes.

The following table lists the default system short codes present in a system's configuration.

Short Code	Telephone Number	Feature	A-Law	MU-Law
*00	Blank	Cancel All Forwarding 450	✓	✓
*01	Blank	Forward Unconditional On 476	✓	✓
*02	Blank	Forward Unconditional Off 476	✓	✓
*03	Blank	Forward On Busy On 474	✓	✓
*04	Blank	Forward On Busy Off 474	✓	✓
*05	Blank	Forward On No Answer On 475	✓	✓
*06	Blank	Forward On No Answer Off 475	✓	✓
*07*N#	N	Forward Number 473	✓	✓
*08	Blank	Do Not Disturb On 466	✓	✓
*09	Blank	Do Not Disturb Off 466	✓	✓
*10*N#	N	Do Not Disturb Exception Add 465	✓	✓
*11*N#	N	Do Not Disturb Exception Del 465	✓	✓
*12*N#	N	Follow Me Here 470	✓	✓
*13*N#	N	Follow Me Here Cancel 471	✓	✓
*14*N#	N	Follow Me To 471	✓	✓
*15	Blank	Call Waiting On 449	✓	✓
*16	Blank	Call Waiting Off 449	✓	✓
*17	?U	Voicemail Collect 501	✓	✓
*18	Blank	Voicemail On 502	✓	✓
*19	Blank	Voicemail Off 502	✓	✓
*20*N#	N	Set Hunt Group Night Service 493	✓	✓
*21*N#	N	Clear Hunt Group Night Service 453	✓	✓
*22*N#	N	Suspend Call 499	✓	✗
*23*N#	N	Resume Call 488	✓	✗
*24*N#	N	Hold Call 478	✓	✗
*25*N#	N	Retrieve Call 489	✓	✗
*26		Clear CW 452	✓	✗
*27*N#	N	Hold CW 479	✓	✗
*28*N#	N	Suspend CW 499	✓	✗
*29	Blank	Toggle Calls 500	✓	✓
*30	Blank	Call Pickup Any 444	✓	✓
*31	Blank	Call Pickup Group 445	✓	✓
*32*N#	N	Call Pickup Extn 444	✓	✓
*33*N#	N	Call Queue 447	✓	✓
*34N;	N	Hold Music 479	✓	✓
*35*N#	N	Extn Login 468	✓	✓
*36	Blank	Extn Logout 468	✓	✓
*37*N#	N	Park Call 484	✓	✓
*38*N#	N	Unpark Call 500	✓	✓
*39	1	Relay On 487	✓	✓
*40	1	Relay Off 487	✓	✓
*41	1	Relay Pulse 488	✓	✓
*42	2	Relay On 487	✓	✓
*43	2	Relay Off 487	✓	✓
*44	2	Relay Pulse 488	✓	✓
*45*N#	N	Acquire Call 448	✓	✓
*46	Blank	Acquire Call 448	✓	✓
*47	Blank	Conference Add 454	✓	✓

Short Code	Telephone Number	Feature	A-Law	MU-Law
*48	Blank	Voicemail Ringback On 503	✓	✓
*49	Blank	Voicemail Ringback Off 503	✓	✓
*50	Blank	Forward Huntgroup On 472	✓	✓
*51	Blank	Forward Huntgroup Off 472	✓	✓
*52	Blank	Cancel or Deny 452	✓	✓
*53*N#	N	Call Pickup Members 446	✓	✓
*57*N#	N	Forward On Busy Number 473	✓	✓
*70	Blank	Call Waiting Suspend 449	✓	✗
*70*N#	N	Dial Physical Extn By Number 461	✗	✓
*71*N#	N	Dial Physical Extn By ID 461	✗	✓
9000	"MAINTENANCE"	Relay On 487	✓	✓
*91N;	N".1"	Record Message 486	✓	✓
*92N;	N".2"	Record Message 486	✓	✓
9N	N	Dial 455	✗	✓
?	.	Dial 455	✓	✗

Additional short codes of the form *DSSN, *SDN, *SKN, these are used by the system for internal functions and should not be removed or altered. Short codes *#N and **N may also visible, these are used for ISDN functions in Scandinavian locales.

For IP Office 4.2+, the default **34* short code for music on hold has changed to **34N;*.

Default auto attendant short codes of the form **81XX*, **82XX*, **83XX* and **84XX* are only added when an Embedded Voicemail auto attendant 400 is added to the system configuration.

6.8 Short Code Features

This following section details the available system short code features.

For each feature the following are listed:

- Telephone Number: The parameter required for the short code feature.
- Default Short Code: Whether the short code feature is used by any default system short code.
- Programmable Button Control: Whether the same action can be assigned to a programmable button.
- Toggling Short Codes
  Previously the Set Hunt Group Night Service 493, Set Hunt Group Out of Service 493 and Hunt Group Enable 480 short code features toggled. That behaviour is not supported in 4.0+.

6.8.1 Acquire Call

See Call Steal 448.

6.8.2 AOC Previous Call

Display of advice of charge information is only supported on T3 phones, T3 IP phones and Phone Manager.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 4.0+.

6.8.3 AOC Reset Total

Display of advice of charge information is only supported on T3 phones, T3 IP phones and Phone Manager.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 4.0+.

6.8.4 AOC Total

Display of advice of charge information is only supported on T3 phones, T3 IP phones and Phone Manager.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 4.0+.

6.8.5 Auto Attendant

This feature is used with embedded voicemail. It allows the recording of the greetings used by auto-attendant services and the transfer of calls to that auto attendant. This feature was previously called Record Greeting.

- Telephone Number: ✓
    - For pre-IP Office 4.1
      Four system short codes are automatically added for each auto attendant. These use a telephone number of the form AA:*Name*.*Y* where *Name* is replaced by the Auto Attendant name and *Y* is 1, 2, 3 or 4 for the morning, afternoon, evening or menu option greeting.
        - You can manually delete the short codes or add additional short codes as required.
        - To create a short code to access an auto attendant, for example to allow internal calls to an auto attendant, omit the *.Y* part of the short code telephone number.
    - For IP Office 4.1+
      Four system short codes (**81XX*, **82XX*, **83XX* and **84XX*) are automatically added for use with all auto attendants, for the morning, afternoon, evening and menu options greetings respectively. These use a telephone number of the form "AA:"*N*".*Y*" where *N* is the replaced with the auto attendant number dialed and *Y* is 1, 2, 3 or 4 for the morning, afternoon, evening or menu option greeting.
        - An additional short code of the form (for example) **80XX / Auto Attendant / "AA:"N* can be added manual if internal dialed access to auto attendants is required.
        - For IP Office 4.1+, to add a short code to access a specific auto attendant, the name method as used for pre-IP Office 4.1 should be used.

- Default Short Code: ✓ See Configuration Settings | Auto Attendant 400.
- Programmable Button Control: ✗
- Software Level: 2.0+.

6.8.6 Break Out

This feature is usable within a Small Community Network 810. It allows a user on one system in the network to specify that the following dialing be processed by another system on the network as if the user dialed it locally on that other system. This feature is not supported for SIP extensions.

For pre-IP Office 5 systems, this feature requires the IP Offices to have *Advanced Small Community Networking* licenses.

- Telephone Number: The IP Address or Name of the system, using * characters in place of . characters.
- Default Short Code: ✗
- Programmable Button Control: BkOut 581
- Software Level: 4.0+.

Example
On a system, to break out via a system called RemoteSwitch with the IP Address 192.168.42.3, either of the following short codes could be used.

Example 1	Example 2
• Code: *80*N#	• Code: *81
• Telephone Number: N	• Telephone Number: RemoteSwitch
• Feature: Break Out	• Feature: Break Out

Example 1 allows break out using any remote switch by dialing its IP address, for example *80*192*168*42*3#. Example 2 does this for a specific remote system by dialing just *81.

6.8.7 Barred

This short code feature can be used for call barring by using the short code as the call destination. This short code feature was previously called Busy. It has been renamed but its function has not changed.

When used in an ARS form 407 that has been configured with an Alternate Route, for callers whose dialing has matched the short code no further routing is applied.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 1.0+.

6.8.8 Busy On Held

When on, busy on held returns busy to new calls when the user has an existing call on hold. This short code feature is useful when a user does not want to be distracted by an additional incoming call when they have a call on hold.

- Telephone Number: ✓ Y or 1 for on, N or 0 for off.
- Default Short Code: ✗
- Programmable Button Control: ✓ BusyH 584
- Software Level: 1.0+.

Example: Turning Busy on Held on
If on, when the user has a call on hold, new calls receive busy tone (ringing if analog) or are diverted to Voicemail if enabled, rather than ringing the user. Note: this overrides call waiting when the user has a call on hold.

- Short Code: *12
- Telephone Number: Y
- Feature: BusyOnHeld

Example: Turning Busy on Held off
Another short code must be created to turn the Busy on Held feature off. If off, when the uses has a call on hold, new calls will still get directed to the user.

- Short Code: *13
- Telephone Number: N
- Feature: BusyOnHeld

6.8.9 Call Intrude

This feature intrudes on the existing connected call of the specified target extension. All call parties are put into a conference and can talk to and hear each other. Use of this feature is subject to the Can Intrude status of the intruder and the Cannot be Intruded status of the target.

A Call Intrude attempt to a user who is idle becomes a Priority Call 486.

- Note that this feature requires conference resources from the system for the duration of the intrusion.
- IP Office 4.0+: Users can use privacy features that to indicate that a call cannot be intruded on. See Private Calls 757.
- IP Office 4.2+: Intruding onto an a user doing silent monitoring (Call Listen 443) is turned into a silent monitoring call.

- Telephone Number: ✓ Target extension number.
- Default Short Code: ✗
- Programmable Button Control: ✓ Intru 586
- See also: Call Listen 443, Dial Inclusion 459.
- Software Level: 1.0+.

6.8.10 Call Listen

This feature allows a user to monitor another user's call without being heard. Monitoring is different from call intrusion. Note that this feature requires conference resources from the system for the duration of the intrusion.

- WARNING
  Monitoring is not enabled by default. The use of monitoring is may be subject to local laws and regulations. Before enabling monitoring you must ensure that you have complied with all applicable local laws and regulations. Failure to do so may result in severe penalties.

Monitoring can be accompanied by a tone heard by all parties. Use of the tone is controlled by the Beep on Listen 163 setting on the System | Telephony | Tones & Music 163 tab. The default for this setting is on. This is the only indication of monitoring, if enabled, given to the monitored user. There is no phone display indication of monitoring.

The use of call listen is dependant on:

- The target being a member of the group set as the user's Monitor Group 276 (User | Telephony | Supervisor Settings) 276.
- The Can Intrude setting of the user listening and the Cannot be Intruded setting of the target. Monitoring is independent of the settings of the third party to the call if they are internal.
- This feature uses system conference resources. If insufficient conference resource are available it will not be possible to use this feature.
- For IP Office 4.0+ a number of new features are supported for call listening:
  - Users can be given privacy features that allow them to indicate that a call cannot be monitored. See Private Calls 757.
  - IP extensions can be monitored including those using direct media. Previously the monitoring of IP extensions could not be guaranteed.
  - The monitoring call can be initiated even if the target user is not currently on a call and remains active until the monitoring user clears the monitoring call.
  - The user who initiated the call listen can also record the call.
- IP Office 4.2+: Intruding onto an a user doing silent monitoring (Call Listen 443) is turned into a silent monitoring call.
- 1400, 1600 and 9600 Series phones with a user button can initiate listening using that button if the target user meets the criteria for listening.

- Telephone Number: Target extension number (extension must be local).
- Default Short Code:
- Programmable Button Control: Listn 587
- See also: Call Intrude 442, Dial Inclusion 459.
- Software Level: 1.0+.

Example
User 'Extn205' wants to be able to monitor calls received by members of the Hunt Group 'Sales'.

1. For user 'Extn205', in the Monitor Group 276 (User | Telephony | Supervisor Settings 276) list box select the hunt group.
2. Ensure that Can Intrude is checked.
3. Create a user short code to allow Extn205 to start monitoring.
   - Short Code: *99*N#
   - Telephone Number: N
   - Line Group ID: 0.
   - Feature: CallListen
4. For each member of the hunt group, check that their Cannot be Intruded setting is unchecked.
5. Now when a member of the 'Sales' hunt group is on a call, Extn205 can replace N in the short code with the extension number of that member and monitor their call.

6.8.11 Call Pickup Any

Pick up the first available ringing call.

- Telephone Number: ✗
- Default Short Code: ✓ *30
- Programmable Button Control: ✓ PickA 590
- See also: Call Pickup Extn 444, Call Pickup Group 445, Call Pickup Members 446, Acquire Call 448, Call Pickup Line 445, Call Pickup User 446.
- Software Level: 1.0+.

Example
Below is an example of the short code setup:

- Short Code: *30
- Feature: CallPickupAny

6.8.12 Call Pickup Extn

Pick up a ringing call from a specific extension.

- Telephone Number: ✓ Target extension number.
- Default Short Code: ✓ *32*N#
- Programmable Button Control: ✓ CpkUp 590
- See also: Call Pickup Any 444, Call Pickup Group 445, Call Pickup Members 446, Acquire Call 448, Call Pickup Line 445, Call Pickup User 446.
- Software Level: 1.0+.

Example
This short code is a default within the Manager configuration. N represents the specific extension. For example, if a user dials *32*201#, they will pick up the call coming into extension 201.

- Short Code: *32*N#
- Telephone Number: N
- Feature: CallPickupAny

6.8.13 Call Pickup Group

Pick up a call ringing any hunt group of which the user is a member.

- Telephone Number: ✗
- Default Short Code: ✓ *31
- Programmable Button Control: ✓ PickG 591
- See also: Call Pickup Any 444, Call Pickup Extn 444, Call Pickup Members 446, Acquire Call 448, Call Pickup Line 445, Call Pickup User 446.
- Software Level: 1.0+.

Example
Below is an example of the short code setup.

- Short Code: *31
- Feature: CallPickupGroup

6.8.14 Call Pickup Line

Pick up an incoming call which is alerting, parked or held. The pickup uses the Line Appearance ID specified in Telephone Number field of the short code. It cannot be used to pickup conferenced calls. The normal user intrusion features are not applied to this pickup feature. This feature is not supported on T3 phones.

- Telephone Number: ✓ Target Line Appearance ID.
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Call Pickup Any 444, Call Pickup Extn 444, Call Pickup Group 445, Call Pickup Members 446, Acquire Call 448, Call Pickup User 446.
- Software Level: 4.0+ (Added in the IP Office 4.0 Q2 2007 maintenance release).

Example
This short code is a default within the Manager configuration. N represents the specific Line Appearance ID.

- Short Code: *99*N#
- Telephone Number: N
- Feature: CallPickupLine

6.8.15 Call Pickup Members

This feature can be used to pick up a ringing or queuing call at an extension that is a member of the Hunt Group specified. The call picked up does not have to be a hunt group call.

- Telephone Number: ✓ Group number or "Group name".
- Default Short Code: ✓ *53*N#
- Programmable Button Control: ✓ PickM 591
- See also: Call Pickup Any 444, Call Pickup Extn 444, Call Pickup Group 445, Acquire Call 448, Call Pickup Line 445, Call Pickup User 446.
- Software Level: 1.0+.

Example

Below is an example of the short code setup. N represents the extension number of the Hunt Group. For example, if a user dials *53*500#, they will pick up the call coming into extension 500 (the hunt group's extension).

- Short Code: *53*N#
- Telephone Number: N
- Feature: CallPickupMembers

6.8.16 Call Pickup User

Pick up an incoming call which is alerting, parked or held. The pickup uses the user extension number specified in Telephone Number field of the short code. If there are multiple calls, priority is given to picking up alerting, then parked and then held in that order of priority. It cannot be used to pickup conferenced calls. The normal user intrusion features are not applied to this pickup feature. This feature is not supported on T3 phones.

- Telephone Number: ✓ Target user extension number.
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Call Pickup Any 444, Call Pickup Extn 444, Call Pickup Group 445, Call Pickup Members 446, Acquire Call 448, Call Pickup Line 445.
- Software Level: 4.0+ (Added in the IP Office 4.0 Q2 2007 maintenance release).

Example

N represents the specific user.

- Short Code: *99*N#
- Telephone Number: N
- Feature: CallPickupUser

6.8.17 Call Queue

Queue the current call to the destination phone, even when the destination phone is busy. This is the same as a transfer except it allows you to transfer to a busy phone.

- Telephone Number: ✓ Target extension number.
- Default Short Code: ✓ *33*N#
- Programmable Button Control: ✓ Queue 592
- Software Level: 1.0+.

Example

Below is an example of the short code setup. N represents the extension the caller wishes to queue for. For example, if a user dials *33*201# while connected to a caller, this caller will be queued for extension 201.

- Short Code: *33*N#
- Telephone Number: N
- Feature: CallQueue

6.8.18 Call Record

This feature allows you to record a conversation. To use this requires Voicemail Pro. Refer to your local regulations in relation to the recording of calls.

  - This feature uses system conference resources. If insufficient conference resource are available it will not be possible to use this feature.
  - IP Office 4.0+: The system provides privacy features that allow users to indicate that a call should not be recorded. See Private Calls 757.

- Telephone Number: ✓ Target extension number.
- Default Short Code: ✗
- Programmable Button Control: ✓ Recor 592
- Software Level: 1.0+.

Example: Record your own extension's call

To use this short code, the user should place the call on hold and dial *55. They will automatically be reconnected to the call when recording begins.

- Short Code: *55
- Telephone Number: None
- Feature: CallRecord

6.8.19 Call Steal

This function can be used with or without a specified user target.

- If the specified target has alerting calls, the function will connect to the longest waiting call.
- If the specified target has no alerting calls but does have a connected call, the function will take over the connected call, disconnecting the original user. This usage is subject to the Can Intrude setting of the user and the Cannot Be Intruded setting of the target. The feature is independent the intrude settings of the third party to the call.
- If no target is specified, the function attempts to reclaim the users last transferred call if it has not been answered or has been answered by voicemail.

- Telephone Number: ✓ Target extension number or blank for last call transferred.
- Default Short Code: ✓ *45*N# and *46
- Programmable Button Control: ✓ Aquire 577
- Software Level: 2.1+

Example: Taking Over a Call
In this example, N represents the extension to be taken over. For example, if a user dials *45*201#, they will take over the current call on extension 201.

- Short Code: *45*N#
- Telephone Number: N
- Feature: Call Steal

Example: Reclaiming a Call
This short code reclaims the last call from your extension. This function is useful when you want to catch a call you have just missed that has gone off to Voicemail.

- Short Code: *46
- Feature: Call Steal

6.8.20 Call Waiting On

Enables call waiting on the user's extension. When on, if the user receives a second calls when already on a call, they hear a call waiting tone in the speech path.

Call waiting settings are ignored for users with multiple call appearance buttons. In this case the appearance buttons are used to indicate additional calls. Call waiting is automatically applied for users with 'internal twinned' phones.

- Telephone Number: ✗
- Default Short Code: ✓ *15
- Programmable Button Control: ✓ CWOn 594
- See also: Call Waiting Off 449, Call Waiting Suspend 449.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *15
- Feature: CallWaitingOn

6.8.21 Call Waiting Off

Disables call waiting on the user's extension. Call waiting may be applied for users with internal twinned phones regardless of their call waiting settings.

- Telephone Number: ✗
- Default Short Code: ✓ *16
- Programmable Button Control: ✓ CWOff 594
- See also: Call Waiting On 449, Call Waiting Suspend 449.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *16
- Feature: Call Waiting Off

6.8.22 Call Waiting Suspend

For phones using call waiting, this feature temporarily disables call waiting for the duration of the user's next call.

- Telephone Number: ✗
- Default Short Code: ✓ *70 (A-Law only)
- Programmable Button Control: ✓ CWSus 595
- See also: Call Waiting On 449, Call Waiting Off 449.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup. This short code is a default within the Manager configuration.

- Short Code: *70
- Feature: CallWaitingSuspend

6.8.23 Cancel All Forwarding

This feature cancels all forms of forwarding on the user's extension including "Follow Me" and "Do Not Disturb".

- Telephone Number: ✗
- Default Short Code: ✓ *00
- Programmable Button Control: ✓ FwdOf 595
- See also: Forward On Busy On 474, Forward On Busy Off 474, Forward On No Answer On 475, Forward On No Answer Off 475, Forward Unconditional On 476, Forward Unconditional Off 476, Do Not Disturb On 466, Do Not Disturb Off 466.
- Software Level: 1.0+.

Example

Below is a sample of the short code setup.

- Short Code: *00
- Feature: CancelCallForwarding

6.8.24 Cancel Ring Back When Free

Cancels any existing ring back (also known as callback) set by the user.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✓ RBak- 596
- See also: Ring Back When Free 489.
- Software Level: 1.0+.

Example: Cancel Ring Back When Free

This example Short Code will cancel Ring Back When Free on the specified extension. N represents the target extension from which you have set a ring back. For example, if Paul has set a ring back on extension 201, he must dial *84*201# to cancel that ring back request.

- Short Code: *84*N#
- Telephone Number: N
- Feature: CancelRingBackWhenFree

6.8.25 Channel Monitor

For Avaya use only.

- Software Level: 1.0+.

6.8.26 Change Login Code

Allows a user to change their log in code.

- Telephone Number: The user's current and new log in codes separated by a *, see the examples below.
- Default Short Code:
- Programmable Button Control:
- Software Level: 4.2+ (Also added to IP Office 4.1 2008Q2 Maintenance release).

Example

The user has a Login Code of *1234* and wants to change it to *5678*. To use the short code below below, the user must dial **60*1234*5678#*.

- Short Code: *60*N#
- Telephone Number: N
- Feature: Change Login Code.

Example

For a user with no login code currently set, they can still use the short code to set a login code. For example using the short code created above to set their login code to 1234 they should dial *60**1234#.

Example

System phone users can also use this short code to change the login code of an other user. For example 403 is configured as a System Phone 765 with a login code of *1234*. User 410 has forgotten their login code and needs it changed. User 403 can do this by dialing the following:

- *60*410*1234*<*new code*>#

6.8.27 Clear After Call Work

This feature can be users who have been configured as CCR agents. It allows them to dial a short code to exit the After Call Work (ACW) state as reported by the Customer Call Reporter (CCR) application.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✓ ACW 579
- See also: Start After Call Work 500.
- Software Level: 4.2 4Q 2008 Maintenance release+.

6.8.28 Clear Call

This feature can be used to end the current call.

- Telephone Number: ✗
- Default Short Code: ✓ *52
- Programmable Button Control: ✓ Clear 598
- Software Level: 1.0+.

Example
Below is a sample of the short code setup. This example could be used in a situation where you are doing a supervised transfer and the party to be transferred to does not want to take the call. In this scenario, you can put the call on hold and dial *52. This will clear the last connected call (for example the party who has just refused the transfer), and retrieve the original call or dial tone.

- Short Code: *52
- Feature: Deny/ClearCall

6.8.29 Clear CW

This feature is most commonly used to end the user's current call and answer the waiting call. Note: Call waiting settings are ignored for users with multiple call appearance buttons.

- Telephone Number: ✗
- Default Short Code: ✓ *26 (A-Law only)
- Programmable Button Control: ✓ ClrCW 598
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *26
- Feature: ClearCW

6.8.30 Clear Hunt Group Night Service

This feature changes the specified hunt group from 'Night Service' mode to 'In Service' mode. This will not override a hunt group in night service due to a time profile.

- Telephone Number: ✓ Hunt group extension number. IP Office 4.0+: If left blank, the short code will affect all hunt groups of which the user is a member.
- Default Short Code: ✓ *21*N#
- Programmable Button Control: ✓ HGNS- 599
- See also: Clear Hunt Group Out Of Service 453, Set Hunt Group Night Service 493, Set Hunt Group Out Of Service 493.
- Software Level: 1.0+.

Example

Below is a sample of the short code setup. N represents the telephone number of the hunt group to be taken out of "Night Service" mode and placed into "In Service" mode. For example, when *21*201# is dialed, the hunt group associated with extension 201 will be taken out of "Night Service" mode.

- Short Code: *21*N#
- Telephone Number: N
- Feature: ClearHuntGroupNightService

6.8.31 Clear Hunt Group Out Of Service

This feature changes the specified hunt group from 'Out of Service' mode to 'In Service' mode. This will not override a hunt group in night service due to a time profile.

- Telephone Number: ✓ Hunt group extension number. IP Office 4.0+: If left blank, the short code will affect all hunt groups of which the user is a member.
- Default Short Code: ✗
- Programmable Button Control: ✓ HGOS- 599
- See also: Clear Hunt Group Night Service 453, Set Hunt Group Night Service 493, Set Hunt Group Out Of Service 493.
- Software Level: 1.0+.

Example

Below is a sample short code using the Clear Hunt Group Out Of Service feature. N represents the telephone number of the hunt group to be taken out of "Out of Service" mode. For example, when *55*201# is dialed, the hunt group associated with extension 201 will be placed into "In Service" mode.

- Short Code: *55*N#
- Telephone Number: N
- Feature: ClearHuntGroupOutOfService

6.8.32 Clear Quota

This feature refreshes the time quota for all services or a specific service.

- Telephone Number: ✓ "Service name" or "" (all services).
- Default Short Code: ✗
- Programmable Button Control: ✓ Quota 600
- Software Level: 1.0+.

6.8.33 Conference Add

Places any calls the user has on hold into a conference with the user. This feature is useful for impromptu conferences.

- Telephone Number: ✗
- Default Short Code: ✓ *47
- Programmable Button Control: ✓ Conf+ 600
- See also: Conference Meet Me 454.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *47
- Feature: ConferenceAdd

6.8.34 Conference Meet Me

This feature allows a user to join a specific conference.

For pre-IP Office 5 systems, this function is not supported on IP500 systems without an *IP500 Upgrade Standard to Professional* license. For IP Office Release 6.0 and higher, IP500 and IP500v2 systems require a *Preferred Edition* 832 license.

- Telephone Number: ✓ Conference number. This can be an alphanumeric value up to 15 characters.
- Default Short Code: ✗
- Programmable Button Control: ✓ CnfRv 601
- See also: Conference Add 454.
- Software Level: 1.0+.

6.8.35 CW

Pick up the waiting call. This feature provides same functionality as pressing the Recall or Hold key on the phone. Unlike the Clear CW feature, this feature does not disconnect you from the existing call when the second call is picked up.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 1.0+.

6.8.36 Dial

This short code feature allows users to dial the number specified to an outside line.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✓ Various depending on locale.
- Programmable Button Control: ✓ Dial 604
- See also: Dial Direct 457, Dial Emergency 458, Dial Extn 458, Dial Inclusion 459, Dial Paging 459.
- Software Level: 1.0+.

Example: Creating a Speed Dial
In this example, users entering 401 on their telephone key pad will dial the New Jersey Office on 212 555 0000.

- Short Code: 401
- Telephone Number: 2125550000

Example: Replace Outgoing Caller ID
This short code is useful in a "call center" environment where you do not want customers to have access to the number of your direct line; you want the general office number displayed. The sample short code below will force the outgoing caller ID to display 123. Note: The usability of this feature is dependent upon your local service provider.

- Short Code: ?
- Telephone Number: .s123

Example: External Dialing Prefix
The short code is for dialing a prefix for an outside line N represents the external number you want to call.

- Short Code: 9N
- Telephone Number: N

Example: Blocking Caller ID
This is for blocking Caller ID for external calls. This feature can be applied to specific external numbers or to all out going calls. In most situations, the company will choose to block the caller ID for all external calls or leave it available for all external calls.

- Short Code: 9N
- Telephone Number: NW

Example: Maximum Call Length
The character t can be used in dialing short codes to set the maximum allowed duration of a call. For example, the following short code will dial a number but then disconnect the call after 20 minutes (plus or minus a minute).

- Short Code: 9N
- Telephone Number: Nt(20)

6.8.37 Dial 3K1

Sets the ISDN bearer capabilities to 3.1Khz audio call.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ D3K1 604
- Software Level: 1.0+.

6.8.38 Dial 56K

Sets the ISDN bearer capabilities to 56Kbps data call.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ D56K 605
- Software Level: 1.0+.

6.8.39 Dial 64K

Sets the ISDN bearer capabilities to 64Kbps data call.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ D64K 605
- Software Level: 1.0+.

6.8.40 Dial CW

Call the specified extension number and force call waiting indication on if the extension is already on a call.

If the user has call appearance buttons programmed, call waiting will not get activated. The next incoming call will appear on an available call appearance button. When there are no available call appearance buttons, the next incoming call will receive busy tone.

- Telephone Number: ✓ Extension number.
- Default Short Code: ✗
- Programmable Button Control: ✓ DCW 606
- Software Level: 1.0+.

Example

N represents the extension number to be dialed. For example, a user dialing *97*201# will force call waiting indication on at extension 201 if extension 201 is already on a call.

- Short Code: *97*N#
- Telephone Number: N
- Feature: DialCW

6.8.41 Dial Direct

Call the extension specified and force automatic answer if supported by the telephone type.

- Telephone Number: ✓ Extension number
- Default Short Code: ✗
- Programmable Button Control: ✓ Dirct 606
- See also: Dial Paging 459.
- Software Level: 1.0+.

Example

This allows the extension specified to be automatically answered. N represents the extension that will be forced to automatically answer. For example, when a user dials *83*201#, extension 201 will be forced to automatically answer the call.

- Short Code: *83*N#
- Telephone Number: N
- Feature: DialDirect

6.8.42 Dial Direct Hot Line

When the line appearance is mapped to a short code using the DialDirectHotLine short code feature, no secondary dial tone is generated and the number is dialed directly. This feature should not be confused with the hot line feature enabled using ?D 433 short codes.

- Telephone Number: ✓
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 3.0 to 4.0.

Example

Below is a sample short code using the DialDirectHotLine feature. The short code *83* should then be set as the prefix for the particular line required.

- Short Code: *83*
- Telephone Number: .
- Feature: DialDirectHotLine

6.8.43 Dial Emergency

Dials the number specified regardless of any call barring applicable to the user.

On all systems, regardless of locale; system and or ARS short codes using the Dial Emergency 458 feature should be created for any required emergency service numbers. Those short codes should be useable by all users from all extensions. Those short codes should route the calls to suitable lines. If the system uses prefixes for external dialing, the dialing of emergency numbers with and without the prefix should be allowed.

The blocking of emergency calls or the rerouting of emergency calls to a intermediate destination other than the central office may be against local and nation laws.

For systems with a United States locale, the Dial Emergency short code acts differently if the system is configured to use an E911 421 adjunct.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ Emrgy 607
- Software Level: 1.0+.

6.8.44 Dial Extn

This feature can be used to dial an internal extension number (user or hunt group).

- Telephone Number: ✓ Extension number.
  - IP Office 4.2+: p(*x*) can be added as a suffix to the Telephone Number to change the priority 344 of a call. Allowable values for *x* are *1*, *2* or *3* for low, medium or high priority respectively. For example *Np(1)*.
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Dial Direct 457, Dial Paging 459, DialPhysicalExtensionByNumber 461, DialPhysicalNumberByID 461.
- Software Level: 1.0+.

Example: Dial on Pick up

The following user short code dials the extension specified the moment the user's handset is picked up.

- Short Code: ?D
- Telephone Number: 201
- Line Group ID: 0
- Feature: DialExtn

6.8.45 Dial Fax

This feature is used to route fax calls via Fax Relay 749.

- Telephone Number: ✓ Fax destination number.
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 5.0+.

Example:
In this example, the line group ID matches the URI configured on a SIP line that has been configured for Fax Relay.

- Short Code: 6N
- Telephone Number: N"@192.16.42.5"
- Line Group ID: 17
- Feature: Dial Fax

6.8.46 Dial Inclusion

This feature intrudes on the existing call of the specified target extension. The intruder and the target extension can then talk but cannot be heard by the other party. This can include intruding into a conference call, where the conference will continue without the intrusion target.

During the intrusion all parties hear a repeated intrusion tone. When the intruder hangs-up the original call parties are reconnected.

Use of this feature is subject to the Can Intrude status (configured in Manager via the User form's Telephony 274 tab) of the intruder and the target extension (the extension to be intruded upon).

Attempting to hold a dial inclusion call simply ends the intrusion part of the call. The inclusion cannot be parked.

- Telephone Number: ✓ Target extension number.
- Default Short Code: ✗
- Programmable Button Control: ✓ Inclu. 607
- See also: Call Intrude 442.
- Software Level: 1.4+.

Example
N represents the extension to be intruded upon. For example, if a user dials *97*201# while extension 201 is on a call, then the user is intruding into extn. 201's current call.

- Short Code: *97*N#
- Telephone Number: N
- Feature: DialInclusion

6.8.47 Dial Paging

This feature makes a page call to an extension or group. The target extension or group members must support page calls (that is be able to auto-answer calls).

- Telephone Number: ✓ Extension or group number.
- Default Short Code: ✗
- Programmable Button Control: ✓ Page 609
- See also: Dial Direct 457.
- Software Level: 1.0+.

Paging Limits
The table below lists the maximum recommended size for paging groups.

Control Unit	Software Level

	4.0+	3.2
IP500/IP500v2	64	–
IP412	48	24
IP406 V2	32	16
Small Office Edition	16	16
IP406 V1	–	16
IP403	–	16

6.8.48 Dial Physical Extension By Number

Dial a specified extension number regardless of the current user logged in at that extension and any forwarding, follow me or do not disturb settings applied by the current extension user. Note that the extension number used is the Base Extension number set against the extension configuration settings.

- Telephone Number: ✓ Base Extension number.
- Default Short Code: ✓ *70*N# (MU-Law only)
- Programmable Button Control: ✓ PhyEx 610
- See also: Dial Physical Number By ID 461, Priority Call 486.
- Software Level: 1.4+.

Example
The example below allows the extension with the base extension number 201 to be called regardless of the extension number of the user currently logged in at that extension.

- Short Code: *97
- Telephone Number: 201
- Feature: DialPhysicalExtnByNumber

6.8.49 Dial Physical Number By ID

Dial a specific extension using its system ID. This may be necessary in hot desking environments where some extensions have been created with no default extension number. Without an extension number, a call can not be made to that extension unless a short code is created.

- Telephone Number: ✓ Extension ID
- Default Short Code: ✓ *71*N# (MU-Law only)
- Programmable Button Control: ✓ DialP 610
- See also: DialPhysicalExtensionByNumber 461, Priority Call 486.
- Software Level: 1.4+.

Example
In the above example, if the telephone at extension ID 16 is not associated with an extension number, a user can dial *97 to connect to that phone. This may be useful in hot desking environments where some extensions may not have a dedicated base extension number.

- Short Code: *97
- Telephone Number: 16
- Feature: DialPhysicalNumberByID

6.8.50 Dial Speech

This feature allows a short code to be created to force the outgoing call to use the Speech bearer capability.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ DSpch 611
- Software Level: 1.0+.

6.8.51 Dial V110

Sets the ISDN bearer capabilities to V110. The call is presented to local exchange as a "Data Call".

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ DV110 611
- Software Level: 1.0+.

6.8.52 Dial V120

Sets the ISDN bear capabilities using V.120. .

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ DV120 612
- Software Level: 1.0+.

6.8.53 Dial Video

The call is presented to the local exchange as a "Video Call".

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✗
- Programmable Button Control: ✓ Dvide 613
- Software Level: 1.0+.

6.8.54 Disable ARS Form

This feature can be used to put an ARS form out of service. It can be used with ARS forms for which an Out of Service Route has been configured in Manager. The short code feature Enable ARS Form can be used to return an ARS form to in service.

- Telephone Number: ARS form number.
- Default Short Code: ❌
- Programmable Button Control: ❌
- See also: Enable ARS Form 466
- Software Level: 4.0+.

6.8.55 Disable Internal Forwards

This feature turns off the forwarding of internal calls for the user. It applies to Forward Unconditional, Forward on Busy and Forward on No Answer.

- Telephone Number: ❌
- Default Short Code: ❌
- Programmable Button Control: ❌
- See also: Disable Internal Forward Unconditional 463, Disable Internal Forward Busy or No Answer 464, Cancel All Forwarding 450, Enable Internal Forwards 467, Enable Internal Forward Unconditional 467, Enable Internal Forward Busy or No Answer 467.
- Software Level: 3.2+.

6.8.56 Disable Internal Forward Unconditional

This feature turns off the forwarding of internal calls for the user. It applies to Forward Unconditional only.

- Telephone Number: ❌
- Default Short Code: ❌
- Programmable Button Control: ❌
- See also: Disable Internal Forwards 463, Disable Internal Forward Busy or No Answer 464, Cancel All Forwarding 450, Enable Internal Forwards 467, Enable Internal Forward Unconditional 467, Enable Internal Forward Busy or No Answer 467.
- Software Level: 3.2+.

6.8.57 Disable Internal Forward Busy or No Answer

This feature turns off the forwarding of internal calls for the user. It applies to Forward on Busy and Forward on No Answer.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Disable Internal Forwards 463, Disable Internal Forward Unconditional 463, Cancel All Forwarding 450, Enable Internal Forwards 467, Enable Internal Forward Unconditional 467, Enable Internal Forward Busy or No Answer 467.
- Software Level: 3.2+.

6.8.58 Display Msg

Allows the sending of text messages to digital phones on the local system.

- Telephone Number: ✓ The telephone number takes the format *N";T"* where:
  - *N* is the target extension.
  - *T* is the text message. Note that the *";* before the text and the *"* after the text are required.
- Default Short Code: ✗
- Programmable Button Control: ✓ Displ 612
- Software Level: 1.0+.

Example

Below is a sample of the short code setup. When used, the target extension will hear a single ring and then see the message. If the target extension is on a call then may need to scroll the display to a free call appearance in order to see the text message.

- Short Code: *78*N#
- Feature: Display Msg
- Telephone Number: N";Visitor in Reception"

6.8.59 Do Not Disturb Exception Add

This feature adds a number to the user's "Do Not Disturb Exception Numbers List". This can be an internal extension number or external ICLID. Calls from that number, except hunt group calls, will ignore the user's Do Not Disturb setting. For further details see Do Not Disturb (DND) 769 in the Telephone Features section.

- Telephone Number: ✓ Telephone number or ICLID. Up to 31 characters.
  For ICLID numbers any prefix added by the system must also be included.
- Default Short Code: ✓ *10*N#
- Programmable Button Control: ✓ DNDX+ 616
- See also: Do Not Disturb Exception Delete 465, Do Not Disturb On 466, Do Not Disturb Off 466.
- Software Level: 1.0+.

Example
N represents the number to be added to the user's "Do Not Disturb Exception List". For example, when a user has DND turned on and dials *10*4085551234#, incoming calls from telephone number (408) 555-1234. All other calls, except those numbers on the exception list hear busy tones or are redirected to voicemail if available.

- Short Code: *10*N#
- Telephone Number: N
- Feature: DoNotDisturbExceptionAdd

Example
In this example, the last number received by the user is added to their exception list.

- Short Code: *99
- Telephone Number: L
- Feature: DoNotDisturbExceptionAdd

6.8.60 Do Not Disturb Exception Delete

This feature removes a number from the user's "Do Not Disturb Exception List". For further details see Do Not Disturb (DND) 769 in the Telephone Features section.

- Telephone Number: ✓ Telephone number or ICLID.
- Default Short Code: ✓ *11*N#
- Programmable Button Control: ✓ DNDX- 616
- See also: Do Not Disturb Exception Add 465, Do Not Disturb On 466, Do Not Disturb Off 466.
- Software Level: 1.0+.

Example
N represents the number to be deleted from the user's "Do Not Disturb Exception List". For example, when a user has DND turned on and the telephone number (408) 555-1234 in their "Do Not Disturb Exception List", dialing *10*4085551234# will remove this phone number from the list. Incoming calls from (408) 555-1234 will no longer be allowed through; instead they will hear busy tone or be redirected to voicemail if available.

- Short Code: *11*N#
- Telephone Number: N
- Feature: DoNotDisturbExceptionDel

6.8.61 Do Not Disturb On

This feature puts the user into 'Do Not Disturb' mode. When on, all calls, except those from numbers in the user's exception list hear busy tones or are redirected to voicemail if available. For further details see Do Not Disturb (DND) 769.

- Telephone Number: ✗
- Default Short Code: ✓ *08
- Programmable Button Control: ✓ DNDOn 617
- See also: Do Not Disturb Off 466, Do Not Disturb Exception Add 465, Do Not Disturb Exception Delete 465.
- Software Level: 1.0+.

Example

Below is a sample of the short code setup.

- Short Code: *08
- Feature: DoNotDisturbOn

6.8.62 Do Not Disturb Off

Cancels the user's 'do not disturb' mode if set. For further details see Do Not Disturb (DND) 769 in the Telephone Features section.

- Telephone Number: ✗
- Default Short Code: ✓ *09
- Programmable Button Control: ✓ DNDOf 617
- See also: Do Not Disturb On 466, Do Not Disturb Exception Add 465, Do Not Disturb Exception Delete 465.
- Software Level: 1.0+.

Example

This short code is a default within the Manager configuration. Below is a sample of the short code setup.

- Short Code: *09
- Feature: DoNotDisturbOff

6.8.63 Enable ARS Form

This feature can be used to put an ARS form in service. It can be used with ARS forms that have been put out of service through Manager or the use of a Disable ARS Form short code.

- Telephone Number: ARS form number.
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 4.0+

6.8.64 Enable Internal Forwards

This feature turns on the forwarding of internal calls for the user. It applies to Forward Unconditional, Forward on Busy and Forward on No Answer.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Disable Internal Forwards 463, Disable Internal Forward Unconditional 463, Disable Internal Forward Busy or No Answer 464, Cancel All Forwarding 450, Enable Internal Forward Unconditional 467, Enable Internal Forward Busy or No Answer 467.
- Software Level: 3.2+.

6.8.65 Enable Internal Forward Unconditional

This feature turns on the forwarding of internal calls for the user. It applies to Forward Unconditional only.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Disable Internal Forwards 463, Disable Internal Forward Unconditional 463, Disable Internal Forward Busy or No Answer 464, Cancel All Forwarding 450, Enable Internal Forwards 467, Enable Internal Forward Busy or No Answer 467.
- Software Level: 3.2+.

6.8.66 Enable Internal Forward Busy or No Answer

This feature turns on the forwarding of internal calls for the user. It applies to Forward on Busy and Forward on No Answer.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Disable Internal Forwards 463, Disable Internal Forward Unconditional 463, Disable Internal Forward Busy or No Answer 464, Cancel All Forwarding 450, Enable Internal Forwards 467, Enable Internal Forward Unconditional 467.
- Software Level: 3.2+.

6.8.67 Extn Login

This feature allows a user to take over ownership of an extension. This requires the user to have a Login Code 276 (User | Telephony | Supervisor Settings 276).

- Note that on some phones, for example IP DECT phones, the dialed digits are displayed and recorded and may include the log in code used.
- Login codes of up to 15 digits are supported with Extn Login 619 buttons. Login codes of up to 31 digits are supported with Extn Login 468 short code 468.

- Telephone Number: ✓ Extension Number*Login Code.
  If just a single number is dialed containing no * separator, the system assumes that the extension number to use is the physical extension's Base Extension 252 number and that the number dialed is the log in code.
- Default Short Code: ✓ *35*N#
- Programmable Button Control: ✓ Login 619
- See also: Extn Logout 468.
- Software Level: 1.0+.

Example: Individual Hot Desking
Based on the above sample short code, Paul (extension 204) can go to another phone (even if it is already logged in by another user) and log in as extension 204 by simply dialing 299. Once Paul has logged into this phone, extension 204 is logged out at Paul's original phone. For Paul to make use of this short code, his log in code must match that configured in the above short code. When Paul logs out of the phone he has "borrowed", his original extension will automatically be logged back in.

- Short Code: 299
- Telephone Number: 204*1234
- Feature: Extnlogin

Example: Log In
The default short code for logging into a phone is configured as shown below. N represents the users extension number followed by a * and then their log in code, for example *35*401*123#.

- Short Code: *35*N#
- Telephone: N
- Feature: ExtnLogin

6.8.68 Extn Logout

This feature logs the user off the phone at which they are logged in.

IP Office 4.0+: This feature cannot be used by a user who does not have a log in code or by the default associated user of an extension unless they are set to forced log in.

- Telephone Number: ✗
- Default Short Code: ✓ *36
- Programmable Button Control: ✓ Logof 619
- See also: Extn Login 468.
- Software Level: 1.0+.

Example
Below is a sample short code using the Extn Logout feature. This short code is a default within the Manager configuration.

- Short Code: *36
- Feature: ExtnLogout

6.8.69 Flash Hook

This feature sends a hook flash signal to the currently connected line if it is an analog line.

- Telephone Number: ❌ Pre-IP Office 4.0/ ✔ IP Office 4.0+: Optional
  IP Office 4.0+: The telephone number field can be used to set the transfer destination number for a Centrex Transfer 783. In this case the use of the short code Forced Account Code and Forced Authorization Code are not supported and the Line Group Id must match the outgoing line to the Centrex service provider.
- Default Short Code: ❌
- Programmable Button Control: ✔ Flash 620
- Software Level: 1.4+.

Example
Below is a sample short code using the Flash Hook feature.

- Short Code: *96
- Feature: FlashHook

6.8.70 FNE

This short code feature is used for Mobile Call Control 730 and one-X Mobile Client 734 support.

- Telephone Number: ✔
  This number sets the required FNE function. For a list of supported functions see one-X Mobile Control 734.
- Default Short Code: ❌
- Programmable Button Control: ❌
- Software Level: 4.2+.

6.8.71 Follow Me Here

Causes calls to the extension number specified to be redirected to the extension initiating the 'Follow Me Here'. If the redirected call receives a busy tone or is not answered, then the call behaves as though the User's extension had failed to answer. For further details see Follow Me 771 in the Telephone Features section.

- Telephone Number: ✓ Extension to redirect to the dialing extension.
- Default Short Code: ✓ *12*N#
- Programmable Button Control: ✓ Here+ 621
- See also: Follow Me Here Cancel 471, Follow Me To 471.
- Software Level: 1.0+.

Example

This feature is used at the Follow Me destination. N represents the extension number of the user wanting their calls redirected to that destination. For example, User A's extension is 224. However they are working at extension 201 and want their calls redirected there. If the following short code is available, they can do this by dialing *12*224# at extension 201.

- Short Code: *12*N#
- Telephone Number: N
- Feature: FollowMeHere

6.8.72 Follow Me Here Cancel

Cancels any Follow Me set on the specified extension. This action can only be performed at the extension to which the Follow Me Here is targetted. For further details see Follow Me 771 in the Telephone Features section.

- Telephone Number: ✓ Extension being redirected to the dialing extension.
- Default Short Code: ✓ *13*N#
- Programmable Button Control: ✓ Here- 621
- See also: Follow Me Here 470, Follow Me To 471.
- Software Level: 1.0+.

Example
This feature is used at the Follow Me destination. N represents the extension number of the user whose calls are being redirected to that destination. For example, User A's extension is 224. However they are working at extension 201 and so have set a Follow Me on their own extension to redirect their calls to 201. If the following short code is available, they can cancel the Follow Me by dialing *13*224# at extension 201.

- Short Code: *13*N#
- Telephone Number: N
- Feature: FollowMeHereCancel

6.8.73 Follow Me To

Causes calls to the extension to be redirected to the Follow Me destination extension specified. For further details see Follow Me 771 in the Telephone Features section.

- Telephone Number: ✓ Target extension number or blank (cancel Follow Me To)
- Default Short Code: ✓ *14*N#
- Programmable Button Control: ✓ FolTo 622
- See also: Follow Me Here 470, Follow Me Here Cancel 471.
- Software Level: 1.0+.

Example
This feature is used at the extension that wants to be redirected. N represents the extension number to which the user wants their calls redirected. For example, User A's extension is 224. However they are working at extension 201 and want their calls redirected there. If the following short code is available, they can do this by dialing *14*201# at extension 224.

- Short Code: *14*N#
- Telephone Number: N
- Feature: FollowMeTo

6.8.74 Forward Hunt Group Calls On

Forward the user's hunt group calls (internal and external) to their forward number when the user has Forward Unconditional active. For further details see Forward Unconditional 773 in the Telephone Features section.

This option is only applied for calls to *Sequential* and *Rotary* type hunt groups. Calls from other types of hunt group types are not presented to the user when they have Forward Unconditional active. Note also that hunt group calls cannot be forwarded to another hunt group.

- Telephone Number: ✗
- Default Short Code: ✓ *50
- Programmable Button Control: ✓ FwdH+ 623
- See also: Forward Hunt Group Calls Off 472, Forward Unconditional On 476, Forward Unconditional Off 476.
- Software Level: 1.0+.

Example
This short code is useful if the hunt group member temporarily uses another workstation and so does not require a permanent extension change.

- Short Code: *50
- Feature: ForwardHuntgroupCallsOn

6.8.75 Forward Hunt Group Calls Off

This feature cancels the forwarding of the user's hunt group calls. For further details see Forward Unconditional 773 in the Telephone Features section.

- Telephone Number: ✗
- Default Short Code: ✓ *51
- Programmable Button Control: ✓ FwdH- 623
- See also: Forward Hunt Group Calls On 472, Forward Unconditional On 476, Forward Unconditional Off 476.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *51
- Feature: ForwardHuntgroupCallsOff

6.8.76 Forward Number

Sets the number to which the user's calls are redirected. This can be an internal or external number. The number is still subject to the user's call barring settings. For further details see Forward Unconditional 773 in the Telephone Features section.

This feature does not activate forwarding; it only sets the number for the forwarding destination.

This number is used for all forward types; Forward Unconditional, Forward on Busy and Forward on No Answer, unless the user has a separate Forward on Busy Number set for forward on busy and forward on no answer functions.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✓ *07*N#
- Programmable Button Control: ✓ FwdNo 624
- See also: Forward On Busy Number 473.
- Software Level: 1.0+.

Example
N represents the forward destination. For example, if extension 224 wants to set the forwarding number to extension 201, the user can dial *07*201#.

- Short Code: *07N*#
- Telephone Number: N
- Feature: ForwardNumber

6.8.77 Forward On Busy Number

Sets the number to which the user's calls are forwarded when Forward on Busy or Forward on No Answer are on. If no Forward on Busy Number is set, those functions use the Forward Number.

This feature does not activate the forwarding, it only sets the number for the forwarding destination.

- Telephone Number: ✓ Telephone number.
- Default Short Code: ✓ *57*N#
- Programmable Button Control: ✓ FwBNo 624
- See also: Forward Number 473.
- Software Level: 1.0+.

Example
N represents the extension number to be forwarded to. For example, if Paul (whose extension is 224) wants to set the forwarding number for his 'Forward on Busy' and/or 'Forward on No Answer' feature to extension 201, Paul can dial *57*201# followed by the short code for the forwarding function.

- Short Code: *57N*#
- Telephone Number: N
- Feature: ForwardOnBusyNumber

6.8.78 Forward On Busy On

This feature enables forwarding when the user's extension is busy. It uses the Forward Number destination or, if set, the Forward on Busy Number destination. If the user has call appearance buttons programmed, the system will not treat them as busy until all the call appearance buttons are in use.

IP Office 3.2+: Forward Internal (User | Forwarding) can also be used to control whether internal calls are forwarded.

- Telephone Number: ✗
- Default Short Code: ✓ *03
- Programmable Button Control: ✓ FwBOn 626
- See also: Forward On Busy Off 474, Cancel All Forwarding 450, Enable Internal Forward Busy or No Answer 467.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *03
- Feature: ForwardOnBusyOn

6.8.79 Forward On Busy Off

This feature cancels forwarding when the user's extension is busy.

- Telephone Number: ✗
- Default Short Code: ✓ *04
- Programmable Button Control: ✓ FwBOf 626
- See also: Forward On Busy On 474, Cancel All Forwarding 450.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *04
- Feature: ForwardOnBusyOff

6.8.80 Forward On No Answer On

This feature enables forwarding when the user's extension is not answered within the period defined by their No Answer Time. It uses the Forward Number destination or, if set, the Forward on Busy Number destination.

IP Office 3.2+: Forward Internal (User | Forwarding) can also be used to control whether internal calls are forwarded.

- Telephone Number: ✗
- Default Short Code: ✓ *05
- Programmable Button Control: ✓ FwNOn 627
- See also: Forward On No Answer Off 475, Cancel All Forwarding 450.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup. Remember that the forwarding number for this feature uses the 'Forward on Busy Number'.

- Short Code: *05
- Feature: ForwardOnNoAnswerOn

6.8.81 Forward On No Answer Off

This feature cancels forwarding when the user's extension is not answered.

- Telephone Number: ✗
- Default Short Code: ✓ *06
- Programmable Button Control: ✓ FwNOf 627
- See also: Forward On No Answer On 475.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *06
- Feature: ForwardOnNoAnswerOff

6.8.82 Forward Unconditional On

This feature enables forwarding of all calls, except group calls, to the Forward Number set for the user's extension. To also forward hunt group calls, Forward Hunt Group Calls On must also be used. For further details see Forward Unconditional 773 in the Telephone Features section.

IP Office 3.2+: Forward Internal (User | Forwarding) can also be used to control whether internal calls are forwarded.

- Telephone Number: ✗
- Default Short Code: ✓
- Programmable Button Control: ✓ FwUOn 628
- See also: Forward Unconditional Off 476.
- Software Level: 1.0+.

Example
Remember that this feature requires having a forward number configured.

- Short Code: *01
- Feature: ForwardUnconditionalOn

6.8.83 Forward Unconditional Off

This feature cancels forwarding of all calls from the user's extension. Note: This does not disable Forward on No Answer and or Forward on Busy if those functions are also on. For further details see Forward Unconditional 773 in the Telephone Features section.

- Telephone Number: ✗
- Default Short Code: ✓ *02
- Programmable Button Control: ✓ FwUOf 628
- See also: Forward Unconditional On 476.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *02
- Feature: ForwardUnconditionalOff

6.8.84 Group Listen Off

Disables the group listen function for the user’s extension. See Group Listen On 477.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✓ GroupListenOn 630
- Software Level: 4.1+.

Example

Below is a sample short code using the Group Listen Off feature.

- Short Code: *27
- Feature: GroupListenOff

6.8.85 Group Listen On

Using group listen allows callers to be heard through the phone's handsfree speaker but to only hear the phone's handset microphone. When group listen is enabled, it modifies the handsfree functionality of the user’s phone in the following manner

- When the user’s phone is placed in handsfree / speaker mode, the speech path from the connected party is broadcast on the phone speaker but the phone's base microphone is disabled.
- The connected party can only hear speech delivered via the phone's handset microphone.
- Group listen is not supported for IP phones or when using a phone's HEADSET button.
- Currently connected calls are not affected by changes to this setting. If group listen is required it must be selected before the call is connected.

This enables listeners at the user’s phone to hear the connected party whilst limiting the connected party to hear only what is communicated via the phone handset.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✓ GroupListenOn 630
- Software Level: 4.1+.

Example

Below is a sample short code using the Group Listen Off feature.

- Short Code: *28
- Feature: GroupListenOn

6.8.86 Headset Toggle

Toggles between the use of a headset and the telephone handset.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✓ HdSet 632
- Software Level: 1.4+.

Example

Below is a sample short code using the Headset Toggle feature. This short code can be used to toggle the feature on/off. If an Avaya supported headset is connected to your telephone, this short code can be used to toggle between using the headset and the telephone handset.

- Short Code: *55
- Feature: HeadsetToggle

6.8.87 Hold Call

This uses the Q.931 Hold facility, and "holds" the incoming call at the ISDN exchange, freeing up the ISDN B channel. The Hold Call feature "holds" the current call to a slot. The current call is always automatically placed into slot 0 if it has not been placed in a specified slot. Only available if supported by the ISDN exchange.

- Telephone Number: ✓ Exchange hold slot number or blank (slot 0).
- Default Short Code: ✗
- Programmable Button Control: ✓ Hold 632
- See also: Hold CW 479, Hold Music 479, Suspend Call 499.
- Software Level: 1.0+.

Example

Below is a sample short code using the Hold Call feature. This short code is a default within the Manager configuration. N represents the exchange hold slot number you want to hold the call on. For example, while connected to a call, dialing *24*3# will hold the call onto slot 3 on the ISDN.

- Short Code: *24*N#
- Telephone Number: N
- Feature: HoldCall

6.8.88 Hold CW

This uses the Q.931 Hold facility, and "holds" the incoming call at the ISDN exchange, freeing up the ISDN B channel. The Hold CW feature "holds" the current call to an exchange slot and answers the call waiting. The current call is always automatically placed into slot 0 if it has not been placed in a specified slot. Only available if supported by the ISDN exchange.

- Telephone Number: ✓ Exchange slot number or blank (slot 0).
- Default Short Code: ✓ *27*N# (A-Law only)
- Programmable Button Control: ✓ HoldCW 633
- See also: Hold Call 478, Suspend Call 499.
- Software Level: 1.0+.

Example
Below is a sample short code using the Hold CW feature.

- Short Code: *27*N#
- Feature: HoldCW

6.8.89 Hold Music

This feature allows the user to check the system's music on hold. See Music On Hold 763 for more information.

- Telephone Number: *Optional.*
  If no number is specified, the default system source is assumed. IP Office 4.2+: (not Small Office Edition) The system supports up to 4 hold music sources, numbered 1 to 4. 1 represents the System Source 163. 2 to 4 represent the Alternate Sources 163.
- Default Short Code: ✓
  - Pre-IP Office 4.2: **34*. If a system is upgraded from pre-4.2 to 4.2+, existing hold music short codes may need to be modified.
  - IP Office 4.2+: **34N;* where *N* is the number of the hold music source required.
- Programmable Button Control: ✓ Music 633
- Software Level: 1.0+.

Example
Below is a sample short code using the Hold Music feature. This short code is a default within the configuration.

- Short Code: *34N;
- Feature: HoldMusic

6.8.90 Hunt Group Disable

This feature disables the user's membership of the specified hunt group. They will no longer receive call to that hunt group until their membership is enabled again. To use this feature, you must already belong to the hunt group. See also Hunt Group Enable 480.

- Telephone Number: ✓ Group number.
- Default Short Code: ✗
- Programmable Button Control: ✓ HGDis 635
- See also: Hunt Group Enable 480.
- Software Level: 1.0+.

Example
N represents the hunt group number from which the user wants to be disabled from. For example, if Paul wants to be disabled from the Sales hunt group (extn. 500), he needs to dial *90*500#.

- Short Code: *90*N#
- Telephone Number: N
- Feature: HuntGroupDisable

6.8.91 Hunt Group Enable

This feature enables the user's membership of a hunt group so they can begin to receive calls to the specified hunt group. To use this feature, the user must already belong to the hunt group. This short code can not be used to add someone to a hunt group, that must be done within Manager's Hunt Group 306 form.

- Telephone Number: ✓ Group number.
- Default Short Code: ✗
- Programmable Button Control: ✓ HGEna 634
- See also: Hunt Group Disable 480.
- Software Level: 1.0+.
  Previously in IP Office 3.2 the Set Hunt Group Night Service, Set Hunt Group Out of Service and Hunt Group Enable short code features toggled. That behaviour is not supported in 4.0 and higher.

Example
This short code can be used to turn the feature on. N represents the hunt group number for which the user wants to start receiving calls. For example, if Paul is already a member of the sales hunt group (extn. 500) but has changed his availability status for that hunt group using hunt group disable, he can make himself available for receiving calls to the Sales hunt group again by dialing *91*500#.

- Short Code: *91*N#
- Telephone Number: N
- Feature: HuntGroupEnable

6.8.92 Last Number Redial

This feature allows an extension to redial the last number they dialed.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 3.0+.

6.8.93 MCID Activate

This feature should only be used in agreement with the ISDN service provider and the appropriate local legal authorities. It allows users with Can Trace Calls 276 (User | Telephony | Supervisor Settings 276) set to trigger a malicious call trace of their previous call at the ISDN exchange. Refer to Telephone Features Malicious Call Tracing 753 for further details.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: Advanced | Miscellaneous | MCID Activate.
- Software Level: 4.0+.

6.8.94 Mobile Twinned Call Pickup

This short code feature allows the user to pickup a call ringing or connected at the destination of their mobile twinning number. This short code can only be used from the primary extension which is being used for the twinning operation.

Note that the use of mobile twinning requires entry of a Mobile Twinning license and may be subject to a time profile.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Set Mobile Twinning Number 495, Set Mobile Twinning On 495, Set Mobile Twinning Off 495.
- Software Level: 3.2+.

6.8.95 Off Hook Station

Enables or disables whether the user's extension acts as a fully handsfree unit. Typically this is used when the answering and clearing of calls is done through an application such as Phone Manager. This feature is also configurable via Phone Manager. For more details see Off Hook Station 274 (User | Telephony | Call Settings 274).

- Telephone Number: ✓ "Y" for on or "N" for off.
- Default Short Code: ✗
- Programmable Button Control: ✓ OHStn 641
- Software Level: 1.0+.

Example: Turning the off hook station off

- Short Code: *99
- Telephone Number: N
- Feature: OffHookStation

Example: Turning the off hook station on

- Short Code: *98
- Telephone Number: Y
- Feature: OffHookStation

6.8.96 Outgoing Call Bar Off

Allows a user to switch off their outgoing call bar status. The short code user must enter their log in code, if set, in order to be successful.

- If you add a short code using this feature to a system it is recommended that you also assign a login code to the No User 767 user to prevent the short code being used to change the status of that user.

- Telephone Number: ✓ The user's log in code.
  - System phone 765 users can use <target user>*<system phone user's login code>.
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 4.1+ (Added to IP Office 4.1 2008Q2 Maintenance release).

Example

The user has a Login Code of *1234*. To use the short code below below, the user must dial *59*1234#.

- Short Code: *59*N#
- Telephone Number: N
- Feature: Outgoing Call Bar Off.

Example

A user set as a System Phone 765 can also switch off the Outgoing Call Bar status of another user. This is done using their own login code. For example the system phone 401 with login code 1234 can switch off the outgoing call bar status of extension 403 as follows:

- *59*403*1234

6.8.97 Outgoing Call Bar On

Allows a user to switch on their outgoing call bar status.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 4.1+ (Added to IP Office 4.1 2008Q2 Maintenance release).

Example

To use the short code below below, the user must dial *58.

- Short Code: *58
- Telephone Number: <blank>
- Feature: Outgoing Call Bar On.

6.8.98 Park Call

Parks the user's current call into the specified park slot number. The call can then be retrieved by other extensions (refer to the appropriate telephone user guide). While parked the caller hears music on hold if available.

Park Timeout 160 (System | Telephony | Telephony 160) controls how long a call will remain parked. When this expires the call will recall to the parking user if they are idle or when they next become idle. The recall call will continue ring and does follow any forwards or go to voicemail.

The 'Unpark Call 500' feature can be used to retrieve calls from specific park slots.

- Telephone Number: ✓ Park slot number.
  If no park slot number is specified when this short code is used, the system automatically assigns a park slot number based on the extension number of the user parking the call plus one digit 0 to 9.
  - Park slot IDs can be up to 9 digits in length. Names can also be used for application park slots.
- Default Short Code: ✓ *37*N#
- Programmable Button Control: ✓ Call Park 588
- See also: Unpark Call 500.
- Software Level: 1.0+.

Example
This short code is a default within the Manager configuration. This short code can be used to toggle the feature on/off. N represents the park slot number in which the call will be parked. For example, if a user wants to park a call to slot number 9, the user would dial *37*9#. The call will be parked there until retrieved by another extension or the original extension.

- Short Code: *37*N#
- Telephone Number: N
- Feature: ParkCall

6.8.99 Private Call

Short codes using this feature toggle on or off private call status.

When on, any subsequent calls cannot be intruded on, bridged into or silently monitored until the user's private call status is switched off.

Note that use of private calls is separate from the user's intrusion settings. If a user is set to Cannot be Intruded, switching private calls off does not affect that status. To allow private calls to be used to fully control the user status, Cannot be Intruded should be disabled for that user.

Private call status can also be switched on or off using a short code features Private Call On 485 and Private Call Off 485 feature or a programmed button set to the Private Call 645 action.

- Telephone Number: Optional. Number to dial for private call.
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 4.0+.

6.8.100 Private Call Off

Short codes using this feature turn off private call status for the user if set. The short code features Private Call 484 and Private Call On 485 can be used to turn private call on.

When on, any subsequent calls cannot be intruded on, bridged into or silently monitored until the user's private call status is switched off.

Note that use of private calls is separate from the user's intrusion settings. If a user is set to Cannot be Intruded, switching private calls off does not affect that status. To allow private calls to be used to fully control the user status, Cannot be Intruded should be disabled for that user.

- Telephone Number: ✘
- Default Short Code: ✘
- Programmable Button Control: Advanced | Call | Private Call. 645
- Software Level: 4.0+.

6.8.101 Private Call On

Short codes using this feature turn on the private call settings for the user regardless.

When on, any subsequent calls cannot be intruded on, bridged into or silently monitored until the user's private call status is switched off.

Note that use of private calls is separate from the user's intrusion settings. If a user is set to Cannot be Intruded, switching private calls off does not affect that status. To allow private calls to be used to fully control the user status, Cannot be Intruded should be disabled for that user.

Private call status can be switched off using a short code with the Private Call Off 485 feature or a programmed button set to the Private Call action. To enable private call status for a single following call only the Private Call 484 short code feature should be used.

- Telephone Number: ✘
- Default Short Code: ✘
- Programmable Button Control: Advanced | Call | Private Call. 645
- Software Level: 4.0+.

6.8.102 Priority Call

This feature allows the user to call another user even if they are set to 'do not disturb'. Priority calls to a user without DND will follow forwarding and follow me settings but will not go to voicemail.

Telephone Number: ✓ Extension number.

- Default Short Code: ✗
- Programmable Button Control: ✓ PCall 644
- See also: DialPhysicalExtensionByNumber 461, DialPhysicalNumberByID 461.
- Software Level: 1.0+.

Example
N represents the extension number to be called, despite the extension being set to 'do not disturb'. For example, if extension 201 has 'do not disturb' enabled, a user can dial *71*201# and still get through. This short code is useful for companies that frequently use the 'do not disturb' feature and can be given to Managing Directors or people who may need to get through to people regardless of their 'do not disturb' status.

- Short Code: *71*N#
- Telephone Number: N
- Feature: PriorityCall

6.8.103 Record Message

This short code feature is used to record hunt group announcements on Embedded Voicemail, see Hunt Group | Announcements 320. IP Office 5+: It is also used to record mailbox user name prompts for the auto attendant 403 Dial by Name function.

- Telephone Number: ✓
  - For a hunt group queue announcement, use the hunt group extension number followed by *".1"*.
  - For a hunt group still queue announcement, use the hunt group extension number followed by *".2"*.
  - For a mailbox user name prompt, use the user extension number followed by *".3"*.
- Default Short Code: ✓ *91N; and *92N;
- Programmable Button Control: ✗
- Software Level: 4.0+.

Example
For a hunt group with extension number 300, the default short codes *91N; / Record Message / N".1" and *92N; / Record Message / N".2" can be used to allow recording of the announcements by dialing *91300# and *92300#.

To allow users to record their own name prompt, the short code *99# / Record Message / E."3" can be used. The E is replace by the extension number of the dialing user.

6.8.104 Relay On

This feature closes the specified switch in the system's external output (EXT O/P) port.

- Telephone Number: ✓ Switch number (1 or 2).
- Default Short Code: ✓ *39 (Switch 1), *42 (Switch 2), *9000*
- Programmable Button Control: ✓ Rely+ 646
- See also: Relay Off 487, Relay Pulse 488.
- Software Level: 1.0+.

Example
This short code is a default within the Manager configuration. This short code is useful for companies that have external devices, such as door controls, connected to the system. Based on this sample short code, a user dialing *42 is closing switch number 2 to activate an external device.

- Short Code: *42
- Telephone Number: 2
- Feature: RelayOn

Analog Modem Control
On systems with an analog trunk card in the control unit, the first analog trunk can be set to answer V.32 modem calls. This is done by selecting the Modem Enabled option on the analog line settings 189 or using the default short code *9000* to toggle this service on or off. This short code uses the RelayOn feature with the Telephone Number set to "MAINTENANCE". Note that the short code method is always returned to off following a reboot or if used for accessing the system date and time menu.

6.8.105 Relay Off

This feature opens the specified switch in the system's external output (EXT O/P) port.

- Telephone Number: ✓ Switch number (1 or 2).
- Default Short Code: ✓ *40 (Switch 1), *43 (Switch 2)
- Programmable Button Control: ✓ Rely- 646
- See also: Relay On 487, Relay Pulse 488.
- Software Level: 1.0+.

Example
This short code is a default within the Manager configuration. This short code is useful for companies that have external devices, such as door controls, connected to the system. Based on this sample short code, a user dialing *43 is opening switch number 2 to activate an external device.

- Short Code: *43
- Telephone Number: 2
- Feature: RelayOff

6.8.106 Relay Pulse

This feature closes the specified switch in the system's external output (EXT O/P) port for 5 seconds and then opens the switch.

- Telephone Number: ✓ Switch number (1 or 2).
- Default Short Code: ✓ *41 (Switch 1), *44 (Switch 2)
- Programmable Button Control: ✓ Relay 647
- See also: Relay On 487, Relay Off 487.
- Software Level: 1.0+.

Example

This short code is a default within the Manager configuration. This short code is useful for companies that have external devices, such as door controls, connected to the system. Based on this sample short code, a user dialing *44 is opening switch number 2 to activate an external device.

- Short Code: *44
- Telephone Number: 2
- Feature: RelayPulse

6.8.107 Resume Call

Resume a call previously suspended to the specified ISDN exchange slot. The suspended call may be resumed from another phone/ISDN Control Unit on the same line.

- Telephone Number: ✓ Exchange suspend slot number.
- Default Short Code: ✓ *23*N# (A-Law only)
- Programmable Button Control: ✓ Resum 648
- See also: Suspend Call 499.
- Software Level: 1.0+.

Example

Below is sample short code using the Resume Call feature. N represents the exchange slot number from which the call has been suspended. For example, if a user has suspended a call on slot number 4, this user can resume that call by dialing *23*4#.

- Short Code: *23*N#
- Telephone Number: N
- Feature: ResumeCall

6.8.108 Retrieve Call

Retrieves a call previously held to a specific ISDN exchange slot.

- Telephone Number: ✓ Exchange hold slot number.
- Default Short Code: ✓ *25*N# (A-law only)
- Programmable Button Control: ✓ Retriv 648
- See also: Hold Call 478.
- Software Level: 1.0+.

Example

Below is sample short code using the Retrieve Call feature. N represents the exchange slot number from which the call has been placed on hold. For example, if a user has placed a call hold on slot number 4, the user can resume that call by dialing *25*4#.

- Short Code: *25*N#
- Telephone Number: N
- Feature: RetrieveCall

6.8.109 Ring Back When Free

This feature sets a ringback on the specified extension. This sets a 'ringback when free' on an extension currently on a call or a 'ringback when next used' for an extension that is free but does not answer.

When the target extension is next used or ends its current call, the users is rung and when they answer a call is made to the target extension.

- Telephone Number: ✓ Target extension number.
- Default Short Code: ✗
- Programmable Button Control: ✓ RBak+ 649
- See also: Cancel Ring Back When Free 450.
- Software Level: 1.0+.

Example

N represents the target extension from which you want to receive the callback. For example, if you call extension 201 but the line is busy, hang up and then dial *71*201#. When extension 201 disconnects from its current call, your phone will ring. Once you pick up the phone, extension 201's line will start ringing to indicate an incoming call.

- Short Code: *71*N#
- Telephone Number: N
- Feature: RingBackWhenFree

6.8.110 Secondary Dial Tone

Secondary dial tone is a system feature to generate a secondary dial tone after the user has begun dialing an external number. This dial tone is then played until the number dialing and an external trunk seized.

- Pre-IP Office 4.0: Secondary dial tone is triggered through the use of the secondary dial tone short code feature.
- IP Office 4.0+: The use of this short code feature has been replaced by the Secondary Dial Tone check box option on ARS forms.

- Telephone Number: ✓ Digit which triggers secondary dial tone.
- Default Short Code: ✓ 9 (MU-Law only)
- Programmable Button Control: ✗
- Software Level: 1.0+.

Example

For pre-4.0 systems secondary dial tone works in two parts. The following system short code will trigger secondary dial tone. To use it to trigger secondary dial tone and then continue dialing, other user, user rights and system short codes should begin with [9].

- **Short Code:** 9
- **Telephone Number:** .
- **Feature:** Secondary Dial Tone

6.8.111 Set Absent Text

This feature can be used to select the user's current absence text. This text is then displayed to internal callers who have suitable display phones or applications. It doesn't changes the users status.

The text is displayed to callers even if the user has forwarded their calls or is using follow me. Absence text is supported across a Small Community Network (SCN).

The absence text message is limited to 128 characters. Note however that amount displayed will depend on the caller's device or application.

- Telephone Number: ✓ The telephone number should take the format *"y,n,text"* where:
  - *y* = 0 or 1 to turn this feature on or off.
  - *n* = the number of the absent statement to use, see the list below:
    - 0 = None.
    - 1 = On vacation until.
    - 2 = Will be back.
    - 3 = At lunch until.
    - 4 = Meeting until.
    - 5 = Please call.
    - 6 = Don't disturb until.
    - 7 = With visitors until.
    - 8 = With cust. til.
    - 9 = Back soon.
    - 10 = Back tomorrow.
    - 11 = Custom.
  - *text* = any text to follow the absent statement.
- Default Short Code: ✗
- Programmable Button Control: ✓ Absnt 653
- Software Level: 1.0+.

Example
The following short code can be used to turn an absent text message on:

- Short Code: *88
- Telephone Number: "1,5,me on 208"
- Line Group ID: 0
- Feature: SetAbsentText

Example
The following short code could be used to turn this facility off. In the Telephone Number the first 0 is used to turn this facility off and the second 0 is used to select the absent statement "None".

- Short Code: *89
- Telephone Number: "0,0"
- Line Group ID: 0
- Feature: SetAbsentText

6.8.112 Set Account Code

This short code feature is used to allow system users to enter a valid account code prior to making a phone call. This short code feature is essential for allowing analog phone users to enter account codes as they cannot enter account code through the phone during a call or after dial a number. Once this short code is set up, any existing account code 375 in the system configuration can be used in conjunction with it.

- Telephone Number: ✓ A valid account code.
- Default Short Code: ✗
- Programmable Button Control: ✓ Acct. 654
- Software Level: 2.1+.

Example

In this example, N represents any valid account code. For the purpose of this example, we will imagine the account code to be 1234. Once this short code is created, a user can dial 11*1234# to get a dial tone for dialing the restricted telephone number or the phone number needing to be tracked for billing purposes.

- Short code: 11*N#
- Telephone Number: N
- Feature: SetAccountCode

6.8.113 Set Authorization Code

This short code feature is only available on systems configured to use authorization codes. See Authorization Codes 404. The feature is used to allow a user to enter a valid authorization code prior to making a phone call.

This short code feature is essential for allowing analog phone users to enter authorization codes. Note that the authorization code must be associated with the user or the user rights to which the user belongs.

- Telephone Number: ✓ A valid authorization code.
- Default Short Code: ✗
- Programmable Button Control: ✗
- Software Level: 3.2+.

6.8.114 Set Hunt Group Night Service

This feature puts the specified hunt group into 'Night Service' mode.

- Telephone Number: ✓ Hunt group extension number.
  IP Office 4.0+: If left blank, the short code will affect all hunt groups of which the user is a member.
- Default Short Code: ✓ *20*N#
- Programmable Button Control: ✓ HGNS+ 654
- See also: Set Hunt Group Out Of Service 493, Clear Hunt Group Night Service 453, Clear Hunt Group Out Of Service 453.
- Software Level: 1.0+.

Previously in IP Office 3.2, the Set Hunt Group Night Service, Set Hunt Group Out of Service and Hunt Group Enable short code features toggled. That behaviour is not supported in 4.0 and higher.

Example
This short code is a default within the Manager configuration. N represents the telephone number of the hunt group to be placed into "Night Service" mode. For example, when *20*201# is dialed, the hunt group associated with extension 201 will be placed into "Night Service" mode.

- Short Code: *20*N#
- Telephone Number: N
- Feature: SetHuntGroupNightService

6.8.115 Set Hunt Group Out Of Service

This feature manually puts the specified hunt group into 'Out of Service' mode. If a time profile has also been defined to control hunt group night service, the action may vary:

- Pre-IP Office 4.0: Set Hunt Group Out of Service cannot be used to override a hunt group put into night service by a time profile.
- IP Office 4.0+: Set Hunt Group Out of Service can be used to override a time profile and change a hunt group from night service to out of service.

- Telephone Number: ✓ Hunt group extension number.
  For IP Office 4.0+, if left blank, the short code will affect all hunt groups of which the user is a member.
- Default Short Code: ✗
- Programmable Button Control: ✓ HGOS+ 656
- Software Level: 1.0+.
  Previously the Set Hunt Group Night Service 493 and Set Hunt Group Out of Service 493 short code features toggled. That behaviour is not supported in 4.0 and higher.

Example
Below is a sample short code using the Set Hunt Group Out Of Service feature. N represents the telephone number of the hunt group to be placed into "Out of Service" mode. For example, when *56*201# is dialed, the hunt group associated with extension 201 will be placed into "Out of Service" mode.

- Short Code: *56*N#
- Telephone Number: N
- Feature: SetHuntGroupOutOfService

6.8.116 Set Inside Call Seq

This feature allows the user to select the ringing used on their extension for internal calls. The number entered corresponds to the ring pattern required. This is 0 for Default Ring, 1 for RingNormal, 2 for RingType1, etc. For more information on selectable ringing patterns, see Ring Tones.

Use of this short code function is applicable to analog phone users only. The distinctive ring used by DS port phones is fixed by the phone type.

- Telephone Number: ✓ Number corresponding to the desired ring pattern. See Ring Tones 762.
- Default Short Code: ✗
- Programmable Button Control: ✓ ICSeq 657
- See also: Set Ringback Seq 497, Set Inside Call Seq 494.
- Software Level: 1.0+.

Example
This Short Code allows a user to change their inside call pattern. N represents the number corresponding to the Call Sequence the user wishes to choose, the numbering starts at 0 selecting Default Ring, 1 selects Ring Normal, 2 selects RingType1, etc. For example, if a user wants to set her/his internal ring pattern to RingType1, the user would dial *80*2# because 2 corresponds to RingType1. This short code is useful for distinguishing an external call from an internal one simply by the ring tone.

- Short Code: *80*N#
- Telephone Number: N
- Feature: SetInsideCallSeq

6.8.117 Set Mobile Twinning Number

This short code feature can be used to set a mobile twinning number. The destination can be any external number the user is able to dial normally. It should include any prefix if necessary.

Note that the use of mobile twinning requires entry of a Mobile Twinning license and may be subject to a time profile.

- Telephone Number: ✓ Twinning destination.
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Set Mobile Twinning On 495, Set Mobile Twinning Off 495, Mobile Twinned Call Pickup 481.
- Software Level: 3.2+.

6.8.118 Set Mobile Twinning On

This short code feature turns on the user's mobile twinning. It requires a mobile twinning number to have been set for the user. That can be done through using the Set Mobile Twinning Number 495 short code feature or through the User | Twinning 293 tab within Manager.

Note that the use of mobile twinning requires entry of a Mobile Twinning license and may be subject to a time profile.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Set Mobile Twinning Off 495, Set Mobile Twinning Number 495, Mobile Twinned Call Pickup 481.
- Software Level: 3.2+.

6.8.119 Set Mobile Twinning Off

This short code feature turns off the user's mobile twinning.

Note that the use of mobile twinning requires entry of a Mobile Twinning license and may be subject to a time profile.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Set Mobile Twinning On 495, Set Mobile Twinning Number 495, Mobile Twinned Call Pickup 481.
- Software Level: 3.2+.

6.8.120 Set No Answer Time

This short code feature allows the user to change their No Answer Time 274 (User | Telephony | Call Settings 274).

- Telephone Number: ✓ Time in seconds.
- Default Short Code: ✗
- Programmable Button Control: ✓ NATim 659
- See also: Set Wrap Up Time 497.
- Software Level: 1.0+.

Example

This short code allows a user to change the length of time they have to answer the phone before it goes to divert or voicemail. N represents the number of seconds. For example, if a user wants to set the allocated answer interval to 15 seconds, the following information needs to be entered: *81*15#.

- Short Code: *81*N#
- Telephone Number: N
- Feature: SetNoAnswerTime

6.8.121 Set Outside Call Seq

This feature allows the user to select the ringing used on their extension for external calls. The number entered corresponds to the ring pattern required. This is 0 for Default Ring, 1 for RingNormal, 2 for RingType1, etc. For more information on selectable ringing patterns, see Ring Tones. Use of this short code function is applicable to analog phone users only. The distinctive ring used by DS port phones is fixed by the phone type.

- Telephone Number: ✓ Number corresponding to the desired ring pattern. See Ring Tones 762.
- Default Short Code: ✗
- Programmable Button Control: ✓ OCSeq 659
- See also: Set Ringback Seq 497, Set Outside Call Seq 496.
- Software Level: 1.0+.

Example

This short code allows a user to change the ringing tone for an external call. N represents the number corresponding to the Call Sequence the user wishes to choose, the numbering starts at 0 selecting Default Ring, 1 selects RingNormal, 2 selects RingType1, etc. For example, if a user wants to set her/his ring pattern for external calls to RingType1, the user would dial *81*2# because 2 corresponds to RingType1. This short code is useful for distinguishing an external call from an internal one simply by the ring tone.

- Short Code: *81*N#
- Telephone Number: N
- Feature: SetOutsideCallSeq

6.8.122 Set Ringback Seq

This feature allows the user to select the ringing used on their extension for ringback calls. The number entered corresponds to the ring pattern required. This is 0 for Default Ring, 1 for RingNormal, 2 for RingType1, etc. For more information on selectable ringing patterns, see Ring Tones.

Use of this short code function is applicable to analog phone users only. The distinctive ring used by DS port phones is fixed by the phone type.

- Telephone Number: ✓ Number corresponding to the desired ring pattern. See Ring Tones 762.
- Default Short Code: ✗
- Programmable Button Control: ✓ RBSeq 659
- See also: Set Outside Call Seq 496, Set Inside Call Seq 494.
- Software Level: 1.0+.

Example
This short code allows a user to change the ringing tone for a ringback call. N represents the number corresponding to the ring tone the user wishes to choose, the numbering starts at 0 selecting Default Ring, 1 selects RingNormal, 2 selects RingType1, etc. For example, if a user wants to set her/his ring pattern for ringback calls to RingType1, the user would dial *81*2# because 2 corresponds to RingType1. This short code is useful for distinguishing a ringback call from any other call simply by the ring tone.

- Short Code: *81*N#
- Telephone Number: N
- Feature: SetRingbackSeq

6.8.123 Set Wrap Up Time

Allows users to change their Wrap-up Time 274 (User | Telephony | Call Settings 274) setting, which specifies the amount of time, after disconnecting from a call, before the user can take another call.

- Telephone Number: ✓ Time in seconds.
- Default Short Code: ✗
- Programmable Button Control: ✓ WUTim 660
- See also: Set No Answer Time 496.
- Software Level: 1.0+.

Example
N represents the number of seconds. For example, if a user wants to set her/his wrap up time to 8 seconds, this user would dial *82*5#. This short code is useful in a "call center" environment where users may need time to log call details before taking the next call. If set to 0 the user does not receive any calls. It is recommended that this option is not set to less than the default of 2 seconds.

- Short Code: *82*N#
- Telephone Number: N
- Feature: SetWrapUpTime

6.8.124 Shutdown Embedded Voicemail

Allows the embedded voicemail service provided by an Avaya memory card in a control unit to be shut down. To restart the service, a **Startup Embedded Voicemail** short code should be used.

The short code has the following effects:

1. Immediately disconnect all current users within embedded voicemail. This is not a polite shutdown.
2. Mark the embedded voicemail as inactive so that it will not receive any new calls.

- **Telephone Number:** ✗
- **Default Short Code:** ✗
- **Programmable Button Control:** ✗
- **Software Level:** 4.0+ (Added in the IP Office 4.0 Q2 2007 maintenance release).

6.8.125 Startup Embedded Voicemail

Restarts the embedded voicemail service provided by an Avaya Memory in a control unit.

- **Telephone Number:** ✗
- **Default Short Code:** ✗
- **Programmable Button Control:** ✗
- **Software Level:** 6.0+

6.8.126 Suspend Call

This feature uses the Q.931 Suspend facility. It suspends the incoming call at the ISDN exchange, freeing up the ISDN B channel. The call is placed in exchange slot 0 if a slot number is not specified.

- Telephone Number: ✓ Exchange slot number or blank (slot 0).
- Default Short Code: ✗
- Programmable Button Control: ✓ Suspe 661
- See also: Resume Call 488.
- Software Level: 1.0+.

6.8.127 Suspend CW

This feature uses the Q.931 Suspend facility. Suspends the incoming call at the ISDN exchange and answer the call waiting. The call is placed in exchange slot 0 if a slot number is not specified. Only available when supported by the ISDN exchange.

- Telephone Number: ✓ Exchange slot number or blank (slot 0).
- Default Short Code: ✓ *28*N# (A-Law only)
- Programmable Button Control: ✓ SusCW 661
- See also: Resume Call 488.
- Software Level: 1.0+.

Example
Sample short code using the Suspend CW feature.

- Short Code: *28*N#
- Feature: Suspend CW

6.8.128 Start After Call Work

This feature can be users who have been configured as CCR agents. It allows them to dial a short code to enter the After Call Work (ACW) state as reported by the Customer Call Reporter (CCR) application.

- Telephone Number: ✗
- Default Short Code: ✗
- Programmable Button Control: ✓ ACW 579
- See also: Clear After Call Work 452.
- Software Level: 4.2 4Q 2008 Maintenance release+.

6.8.129 Toggle Calls

This feature cycles through each call that the user has on hold on the system. This feature is useful when a user with a single-line telephone has several calls on hold and needs to respond to each one in turn.

- Telephone Number: ✗
- Default Short Code: ✓ *29
- Programmable Button Control: ✓ Toggl 663
- Software Level: 1.0+.

Example
Below is sample short code using the Toggle Calls feature (via Manager).

- Short Code: *29
- Feature: ToggleCalls

6.8.130 Unpark Call

Retrieve a parked call from a specified system park slot. In pre-3.2 IP Office, this feature was call Ride Call.

- Telephone Number: ✓ System park slot number.
- Default Short Code: ✓ *38*N#
- Programmable Button Control: ✓ Ride 665
- See also: Park Call 484.
- Software Level: 1.0+.

Example
Below is a sample short code using the Unpark Call feature. N represents the park slot number in which the call you want to retrieve was parked. For example, if a user parked a call to slot number 9, you can retrieve that call by dialing *38*9#.

- Short Code: *38*N#
- Telephone Number: N
- Feature: Unpark Call

6.8.131 Voicemail Collect

This feature connects to the voicemail system. Normally the telephone number field is used to indicate the name of the mailbox to be accessed, for example "?Extn201" or "#Extn201".

- ? indicates 'collect messages'.
- # indicates 'leave a message'. It also instructs the voicemail server to give a brief period of ringing before connecting the caller. The # can be omitted for immediate connection.
- " " quotation marks must be used to enclose the number being sent to the voicemail server. Any text not enclosed by quote marks will be treated as short code wildcards to be interpreted and replaced by the system.

When using Voicemail Pro, names of specific call flow start points can also be used to directly access those start points via a short code. In these cases, ? is not used and # is only used if ringing is required before the start point's call flow begins.

Note: Short codes using the Voicemail Collect feature, with either "Short Codes.name" and "#Short Codes.name" entries in the Telephone Number field are automatically converted to the Voicemail Node feature and name.

- Telephone Number: ✓ See above.
- Default Short Code: ✓ *17
- Programmable Button Control: ✓ VMCol 669
- See also: Voicemail On 502, Voicemail Off 502, Voicemail Node 502.
- Software Level: 1.0+.

Example: Retrieve Messages from Specific Mailbox
This short code allows a user to retrieve messages from the mailbox of the hunt group 'Sales'. This usage is not supported on Voicemail Pro running in Intuity emulation mode unless a custom call flow has been created for the hunt group, refer to the Voicemail Pro help.

- Short Code: *99
- Telephone Number: "?Sales"
- Feature: VoicemailCollect

Example: Record Message to Specific Mailbox
To allow users to deposit a message directly to Extn201's Voicemail box. This short code is useful when you know the person is not at her/his desk and you want to immediately leave a message rather than call the person and wait to be redirected to voicemail.

- Short Code: *201
- Telephone Number: "#Extn201"
- Feature: VoicemailCollect

Example: Accessing a Specific Voicemail Pro Module
This short code can be used in instances where you have a conference bridge set up on the system and a module has been created via Voicemail Pro to access this conference bridge. A short code can be created for internal access to the module. In the sample short code below, the telephone number field contains the name of the module. In this example, if a short burst of ringing is required before connecting the module, "#conferenc" would be used as the telephone number.

- Short Code: *100
- Telephone Number: "conferenc"
- Feature: VoicemailCollect

6.8.132 Voicemail Node

Similar to Voicemail Collect but used for calls being directed to a Voicemail Pro Short Codes start point. Useful if you have set up a short code start point with Voicemail Pro and want to give direct internal access to it.

- Telephone Number: ✓ Voicemail Pro Short Code start point name without quotation marks.
- Default Short Code: ✗
- Programmable Button Control: ✗
- See also: Voicemail Collect 501.
- Software Level: 2.0+.

Example
Having created a short codes start point call flow called Sales, the following system short code can be used to route calls to that call flow:

- Short Code: *96
- Telephone Number: Sales
- Feature: VoicemailNode

6.8.133 Voicemail On

This feature enables the user's voicemail mailbox to answer calls which ring unanswered or arrive when the user is busy.

- Telephone Number: ✗ None.
- Default Short Code: ✓ *18
- Programmable Button Control: ✓ VMOn 670
- See also: Voicemail Off 502.
- Software Level: 1.0+.
  For IP Office 3.2, the Voicemail On and Voicemail Ringback On short code features toggled. For IP Office 4.0 and higher, they no longer toggle.

Example
This short code can be used to toggle the feature on.

- Short Code: *18
- Feature: VoicemailOn

6.8.134 Voicemail Off

This feature disables the user's voicemail mailbox box from being used to answer calls. It does not disable the voicemail mailbox being used as the target for other functions such as call recording or messages forwarded from other mailboxes.

- Telephone Number: ✗ None.
- Default Short Code: ✓ *19
- Programmable Button Control: ✓ VMOff 670
- See also: Voicemail On 502.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *19
- Feature: VoicemailOff

6.8.135 Voicemail Ringback On

This feature enables voicemail ringback to the user's extension. Voicemail ringback is used to call the user when they have new voicemail messages. The ringback takes place each time the extension is used. This feature is useful for users who do not have voicemail light/button indicators on their telephone.

If the user has been configured to receive message waiting indication for any hunt groups, a separate voicemail ringback will occur for each such group and for the users own mailbox.

- Telephone Number: ✗
- Default Short Code: ✓ *48
- Programmable Button Control: ✓ VMRB+ 671
- See also: Voicemail Ringback Off 503.
- Software Level: 1.0+.
  For IP Office 3.2, the Voicemail On and Voicemail Ringback On short code features toggled. For IP Office 4.0 and higher, they no longer toggle.

Example
This short code can be used to turn the feature on.

- Short Code: *48
- Feature: VoicemailRingbackOn

6.8.136 Voicemail Ringback Off

This feature disables voicemail ringback to the user's extension.

- Telephone Number: ✗
- Default Short Code: ✓ *49
- Programmable Button Control: ✓ VMRB- 671
- See also: Voicemail Ringback On 503.
- Software Level: 1.0+.

Example
Below is a sample of the short code setup.

- Short Code: *49
- Feature: VoicemailRingbackOff

Chapter 7.
Button Programming

7. Button Programming

This section covers the system actions that can be assigned to programmable buttons on Avaya phones. This assignment can be done through the system configuration using Manager and for some functions using the phone itself.

- **Appearance Functions**
  The functions *Call Appearance*, *Bridged Appearance*, *Coverage* and *Line Appearance* are collectively known as "appearance functions". For full details of their operation and usage refer to the Key and Lamp Operation 674 section. The following restrictions must be observed for the correct operation of phones.

Software	
2.1 or 3.0DT	On systems running IP Office 2.1 or 3.0DT software, the first programmable button must be set as a call appearance button. This requirement is necessary for the correct operation of phone functions such as call log.
3.0+	For systems running 3.0 or higher (excluding 3.0DT), the first 3 programmable buttons must be set as call appearance buttons. This requirement is necessary for the correct operation of functions such as transfer, conference and call log. • On phones with only two physical programmable buttons both buttons must be used as call appearance buttons.
4.2+	T3 phones support Line Appearance buttons only and are exempt from the above rules.
5.0+	• Users can have line appearances programmed before call appearances. This option is only supported by users using 1400, 1600 and 9600 Series phones. • On 1400, 1600 and 9600 Series phones, all the appearance buttons must be programmed as a continuous block with no gaps between the appearance buttons.

- **Phone Support**
  Note that not all functions are supported on all phones with programmable buttons. Where possible exceptions have been indicated.
  - Actions that use status feedback are only supported on buttons that provide that feedback through lamps or icons.

7.1 Programming Buttons

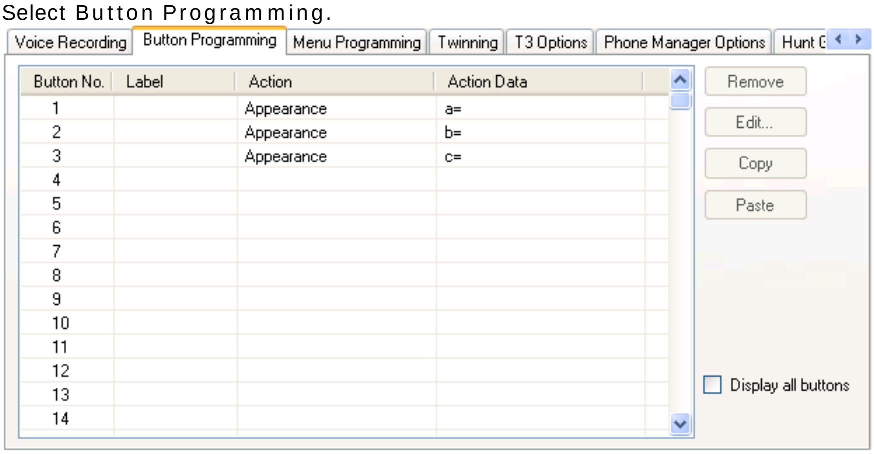
7.1.1 Programming Buttons with Manager

Using Manager, if only button programming changes are required, the configuration changes can be merged back to the system without requiring a reboot.

- **Appearance Functions**
  The functions *Call Appearance*, *Bridged Appearance*, *Coverage* and *Line Appearance* are collectively known as "appearance functions". For full details of their operation and usage refer to the Key and Lamp Operation 674 section. The following restrictions must be observed for the correct operation of phones.

Software	
2.1 or 3.0DT	On systems running IP Office 2.1 or 3.0DT software, the first programmable button must be set as a call appearance button. This requirement is necessary for the correct operation of phone functions such as call log.
3.0+	For systems running 3.0 or higher (excluding 3.0DT), the first 3 programmable buttons must be set as call appearance buttons. This requirement is necessary for the correct operation of functions such as transfer, conference and call log. • On phones with only two physical programmable buttons both buttons must be used as call appearance buttons.
4.2+	T3 phones support Line Appearance buttons only and are exempt from the above rules.
5.0+	• Users can have line appearances programmed before call appearances. This option is only supported by users using 1400, 1600 and 9600 Series phones. • On 1400, 1600 and 9600 Series phones, all the appearance buttons must be programmed as a continuous block with no gaps between the appearance buttons.

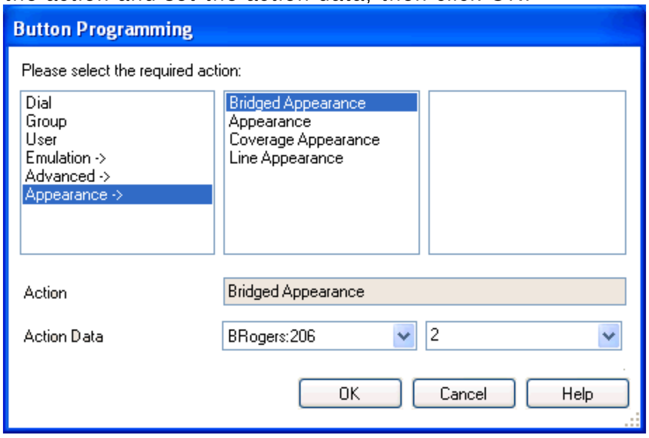
1. Using Manager load the current configuration from the system.
2. Select the  User required to display their configuration details.
3. Select **Button Programming**.



- The number of button displayed is based on the phone associated with the user when the configuration was loaded. This can be overridden by selecting **Display All Buttons**. This may be necessary for users who switch between different phones using hot desking or have an expansion unit attached to their phone.

4. For the required button, either select the button and then click **Edit** or double-click the button.

5. Edit the settings as required. Use the ... button to display the menu for selecting the required button action. Select the action and set the action data, then click OK.



6. Click OK. Repeat for any other buttons.

7. Click OK.

An alternate method for the above programming is to right-click on the various fields. To do this start with the Action field, then Action Data and then Label if required.

7.1.2 Programming Button via the Menu Key

On 4412D+, 4424D+, 4612IP, 4624IP, 6408D, 6416D, 6424D phones the Menu button can be used to program some functions against other buttons. This programming also includes programmable buttons on any associated add-on units associated with the phone. Buttons already programmed as appearance buttons cannot be altered using these methods.

- A Self-Administer 651 button can also be added to allow the phone user to program the functions on their other buttons, see Self-Administer 651.
- For IP Office 4.2+, T3 phone users can also program buttons using a Menu function, see T3 Self-Administration 512.

Setting a Button to Dial a Number

This process sets the selected programmable button to the Dial 604 function in the system configuration.

1. With the phone idle and on-hook, press MENU.
2. Press ▶ and select PROG.
3. Enter the number required. The left-most display button can be used to backspace and the right-most display button can be used to Clear the whole number.
4. Press the programmable button against which the number should be set.
5. If the button is already programmed, options to replace (Repla), keep (Keep) or delete (Delet) the buttons existing programming appear. Select the option required.
6. The message BUTTON PROGRAMMED! indicates that the button is now programmed. Select Cont and then press Exit.

Setting a Button to a Switch Function

This process allows users to program there own Group 629, User 666 and Park 642 slot monitor buttons. It also allows the programming of Dial 604 and Flash Hook 620 buttons.

1. With the phone idle and on-hook, press Menu twice.
2. Press ▶ and select ProgA.
3. Press ▶ and select DSS.
4. Use the ◀ and ▶ buttons to display the function required. Press the display button below the function to select it.
5. If the function requires a telephone number value set, enter the number. The left-most display button can be used to backspace and the right-most display button can be used to Clear the whole number.
6. Press the programmable button against which the number should be set.
7. If the button is already programmed, options to replace (Repla), keep (Keep) or delete (Delet) the buttons existing programming appear. Select the option required.
8. The message BUTTON PROGRAMMED! indicates that the button is now programmed. Select Cont and then press Exit.

Setting Buttons to Admin Function

Phones with a Menu key can program a range of self-administer functions onto their programmable buttons. These are:

- Dir - Directory. 615
- Drop - Drop. 618
- HFAns - Internal Auto-Answer. 637
- Timer - Timer. 662
- AutCB - Automatic Callback. 580
- Prog - Abbreviated Dial Program. 574
- CFrwd - Call Forwarding All. 585
- CPark - Call Park. 588
- SAC - Send All Calls. 652
- TmDay - Time of Day. 662
- Admin - Self-Administer. 651
- Acct - Account Code Entry. 575
- AD - Abbreviated Dial. 573
- Park - Call Park to Other Extn. 642
- GrpPg - Group Paging. 631
- CPkUp - Call Pickup. 590
- DPkUp - Directed Call Pickup. 614
- RngOf - Ringer Off. 650
- Spres - AD Suppress. 578
- HdSet - Headset Toggle. 632
- HGNS+ - Set Hunt Group Night Service. 654

This is the same set of functions that can be programmed by users with a button set to Self-Administer (see Self-Administer 651).

1. With the phone idle and on-hook, press Menu .
2. Press ▶ twice and select Admin.
3. Use the ◀ and ▶ keys to display the function required and then select it by pressing the display button below the feature.
   - Selecting Expl? changes the display from short name mode to long name mode. In this mode the full names of the features are displayed. Select SHORTMODE to return to that mode.
4. If the function requires a telephone number value set, enter the number. The left-most display button can be used to backspace and the right-most display button can be used to Clear the whole number.
5. Press the programmable button against which the number should be set.
6. If the button is already programmed, options to replace (Repla), keep (Keep) or delete (Delet) the buttons existing programming appear. Select the option required.
7. The message BUTTON PROGRAMMED! indicates that the button is now programmed. Select Cont and then press Exit .

7.1.3 Programming Button via an Admin Button

The Admin [651] (also called Self-Administer [651]) function can be assigned to a programmable button on a user's phone. That button then allows the user to program functions against other programmable buttons on their phone, except those already set as appearance buttons.

- Admin buttons are only supported on 2410, 2420, 4406D+, 4412D+, 4424D+, 4606IP, 4612IP, 4624IP, 5410, 5420, 6408D, 6416D and 6424D.
- On 4412D+, 4424D+, 4612IP, 4624IP, 6408D, 6416D, 6424D phones:
  - Admin can be permanently accessed via Menu , , , Admin.
  - Admin1 can be permanently accessed via Menu , Menu , , ProgA, , , DSS.

Using an Admin Button

1. With the phone idle and on-hook, press the button programmed to Admin or Admin1.
2. The list of available functions is shown. Use the and buttons to move through the list.
   - Selecting Expl? changes the display from short name mode to long name mode. In this mode the full names of the features are displayed. Select SHORTMODE to return to that mode.
3. Select the function required.
4. If the function requires a telephone number value set, enter the number. The left-most display button can be used to backspace and the right-most display button can be used to Clear the whole number.
5. Press the programmable button against which the number should be set. On phones with multiple pages of buttons use the and button to select the required page before pressing the button to program.
6. If the button is already programmed, options to replace, keep or delete the button's existing programming appear. Select the option required.
7. The message BUTTON PROGRAMMED! indicates that the button is now programmed.
8. Select Cont. and then press Exit or lift the handset to go off-hook.

7.1.4 T3 Self-Administration

IP Office 4.2+ supports functions for T3 phone users to be able to program their own buttons. This is similar to the existing Self-Administer 651 button supported on other phones but is configured and accessed via different methods.

The user accesses button programming through Menu | Settings | Button programming. This function is not available by default, instead it must be configured as available for the user using the method detailed below.

Once enabled, the user is able to configure the following functions on buttons:

Function	Description
empty	Returns the button to it normal default function.
Account Code	Allows the user to enter an account code before or during a call. The account code can be preset or entered after the button press. See the Account Code Entry 575 function.
Callback	Set a callback from the currently dialed extension number. See the Automatic Callback 580 function.
Call list	Displays a list of calls received. See the Call List 586 function.
Call Tracing	Activate malicious call tracing. See the MCID Activate 640 function and Malicious Call Tracing (MCID) 753.
Dial	Dial a preset number or partial number that can be completed after the button press. See the Dial 604 function.
Dial Intercom	Make a page call to the selected target if it supports handsfree answer. See Dial Intercom 608.
Directory	Display the system directory. See the Directory 615 function.
Do not disturb	Toggle the phone between do not distrub on and off. See the Send All Calls 652 function.
Follow me here	Activate/cancel follow me here. See the Follow Me Here 621 function.
Forward unconditional	Activate/cancel forward all calls. See the Forward Unconditional On 628 function.
Group Paging	Page a group of phones. See the Group Paging 631 function.
Group Membership	Enable/disable the user membership of a group or all groups. See the Hunt Group Enable 634 function.
Group State	Change a hunt group's out of service status. See the Set Hunt Group Out of Service 656 function.
Headset	Switch between handset and headset modes. See the Headset Toggle 632 function.
Internal Auto-Answer	Auto connect internal calls after a single tone. See the Internal Auto-answer 637 function.
Login	Access the menu for phone log in. See the Extn Login 619 function.
Logout	Log out from the phone. See the Extn Logout 619 function.
Night Service	Change a hunt group's night service status. See the Set Hunt Group Night Service 654 function.
Paging	Page an extension or group. See the Dial Paging 609 function.
Pickup	Answer a call alerting on the system. See the Call Pickup 590 function.
Pickup Member	Answer a call alerting the hunt group of which the user is a member. See the Call Pickup Members 591 function.
Twinning	Switch mobile twinning on/off and set the twinning destination. Also used to pull a call answered at the twinned number back to the users primary extension. See the Twinning 663 function.
User	Monitor the status of a user. Also used to call them or to pickup calls alerting them. See the User 666 function.
Visual Voice	Create a visual voice access button. See Visual Voice 668.
Voicemail	Equivalent to the Voicemail Collect 669 function.
Voicemail on/off	Switch the use of the user's mailbox to answer unanswered calls calls on/off. See the Voicemail On 670 function.

The user will need to be made aware of which physical buttons can be programmed as this varies between the different T3 phones. See T3 Compact 566, T3 Classic 567 and T3 Comfort 568.

Configuring a T3 User for Button Programming

1. Using Manager, receive the configuration from the system.
2. Select the T3 user and then select Menu Programming.
3. Set the action for one of the menus to Self-Administer.

4. Send the configuration back to the system.
5. The user will now be able to access button programming from their phone via *Menu -> Settings -> Button programming*.

7.1.5 Customizing Text Labels

On Avaya phones where the programmable button has an adjacent display area, a text label is displayed to indicate the button's function.

Each function has a default label. However this default label can be replaced by a custom text label of up to 13 characters (A-Z, a-z, 0-9 plus ., *, - and #). This custom label can be entered either by the user from their phone or through the system configuration.

Entering a Custom Label from a Phone

This option is supported on 2410, 2420, 4610, 4620, 4621, 4625, 5410, 5420, 5610, 5620 and 5621 phones.

1. First exit any other phone menu mode and ensure that the phone is idle.
2. The next step depends on the type of phone:
   - 2410, 5410: Press any key below the display. The press Label.
   - 2420, 5420: Press Label.
   - IP Phones: Press ✓ Options. Press and then Feature Button Labeling.
3. The options may vary slightly between phone types but are similar.
   - Edit or Relabel feature buttons allows you replace the current labels, see the steps below.
   - Inspect or View default labels displays the default labels.
   - Restore or Restore default labels replaces any custom labels with the default labels.
   - Done exits the button labelling options.
4. Press Edit.
5. Select the function key whose label you wish to change.
6. Begin entering the New Label text using the telephone keypad.
   - Each number key is marked with the letters it provides. You may have to press the key more than once depending on the character required. For example, the key 6 is also marked as M, N and O. To enter an O, re-press the 6 key until an O is displayed.
   - If the next character you want is on another key, simply key the next character.
   - If the next character you want to enter is on the same key just used, press to move the cursor right and then enter the character.
   - Pressing the * key once enters a . (period). Pressing it twice enters a *.
   - Pressing the # key once enters a - (dash). Pressing it twice enters a #.
   - By default the first letter entered and the first letter after any space are entered in upper-case whilst all other character are entered in lower-case. To change the case of the current character press Case.
   - To enter a space press Space.
   - Use the key to move the cursor one space right.
   - If you make a mistake, use Backspace to delete the character to the left of the cursor.
   - If you have made a mistake in the middle of a character string and do not wish to backspace and re-enter all the characters use the key to step back to one character before the point where you wish to edit. Either insert the new character or press Backspace to delete the character to the left of the cursor.
   - Press Clear to delete all the current text.
7. When the new name is set as required, press Save. To return to the label options screen without saving the changes, press Cancel.
8. Select another button to re-label or press Done.

7.1.6 Interactive Button Menus

IP Office 4.0+: For display phones where the a button has been configured without a specific number, the menu for number entry. The menu includes a Dir option for selecting a number from the directories held by the system.

Functions that use the interactive menu are:

Feature	Directory lists...	Feature	Directory lists...
Automatic Intercom 581	Users	Follow Me Here Cancel 621	Users
Acquire Call/Call Steal 577	Users	Follow Me Here 621	Users
Call Forwarding All 585	Users	Follow Me To 622	Users
Call Intrude 586	Users	Forward Number 624	Users/Groups
Call Park to Other Extension 589	Users	Forward Busy Number 624	Users/Groups
Dial Inclusion 607	Users	Group Paging 631	Users/Groups
Dial Intercom 608	Users	Leave Word Calling 638	Users/Groups
Directed Call Pickup 614	Users/Groups	Priority Calling 644	Users/Groups

- User 666 and Group 629 buttons can be used to indicate the required user or hunt group only if those buttons are on an associated button module. User and Group buttons on the users extension are not accessible while the interactive button menu is being displayed.
- For functions supported across Small Community Networks, the directory will include SCN users and advertised hunt groups.

7.1.7 Label Templates

The attached zip file below contains Word document templates for the paper programmable key labels on various phones supported by the system. Two templates are provided, one for A4 size paper, the other for US Letter sized paper.

- DSS Key Label Template File (Microsoft Word .dot Files)

For ETR, 1400 and 1600 phones, a number of tools and perforated printable labels are available. For further details visit http://support.avaya.com and search for information on DESI. Alternatively visit http://www.desi.com.

Manager is able to pass user button information to a DESI application on the same PC. This allows printing of labels using the label text set within the system configuration. Currently only ETR, 1400 and 1600 phonesphones are supported by DESI.

7.2 Phone Details

The table below lists the feature phones supported by IP Office Standard Version mode. For the sub-set of phones supported by IP Office Essential Edition - PARTNER® Version and IP Office Essential Edition - Norstar Version modes refer to the appropriate documentation for each mode.

Note that the range of phones supported varies with the control unit software level.

For some phones the annotation 24 (4x6) or similar is used. This can be read as 24 programmable buttons, typically arranged as 4 screen pages with 6 physical buttons.

	Phone	Programmable Buttons	Call Appearances	Button Type	Add-On	Software Level
1.	1010 523	0	–	–	–	6.1+
2.	1040 523	0	–	–	–	6.1+
3.	1120E 524	4	✓	–	Yes	6.1+
4.	1140E 524	6	✓	–	Yes	6.1+
5.	1220 525	8 (2x4)	✓	–	Yes	6.1+
6.	1230 525	12 (2x6)	✓	–	Yes	6.1+
7.	1403 526	3	✓		–	6.0+
8.	1408 526	8	✓		–	6.0+
9.	1416 527	16	✓		DBM32 518	6.0+
10.	1603 528	3	✓		–	4.2 Q4 '08+
11.	1603I 528	3	✓		–	6.0+
12.	1603SW 528	3	✓		–	5.0+
13.	1603I SW 528	3	✓		–	6.0+
14.	1608 528	8	✓		–	4.2 Q4 '08+
15.	1608I 528	8	✓		–	6.0+
16.	1616 529	16	✓		BM32 518	4.2 Q4 '08+
17.	1616I 529	16	✓			6.0+
18.	2010	0	–	–	–	1.0-3.0DT
19.	2030 530	8	✗		20DS 531	1.0-3.0DT
20.	2050 530	8	✗		20DS 531	1.0-3.0DT
21.	20CC 530	8	✗		20DS 531	1.0-3.0DT
22.	20DT	0	–	–	–	1.0-4.2
23.	2402D 532	2 (+12)	✓		–	3.0+
24.	2410D 533	12 (2x6)	✓	CA1	–	3.0+
25.	2420 534	24 (3x8)	✓	CA1	EU24 518	1.4+
26.	3616 536	6	✓	1	–	2.0+
27.	3620 536	6	✓	1	–	3.2+
28.	3626 536	6	✓	1	–	2.0+
29.	3641 537	12	✓	1	–	4.0+
30.	3645 537	12	✓	1	–	4.0+
31.	3701 538	0	–	–	–	3.1+
32.	3711 538	0	–	–	–	3.1+
33.	3720 538	0	–	–	–	5.0+
34.	3725 538	0	–	–	–	5.0+
37.	3810 538	4	✓	–	–	2.1+

	Phone	Programmable Buttons	Call Appearances	Button Type	Add-On	Software Level
38.	3910 [539]	0	–	–	–	6.0+
39.	3920 [539]	0	–	–	–	6.0+
46.	4406D [540]	6	✓		–	1.0+
47.	4412D [541]	24	✓	/	4450 [543]	1.0+
48.	4424D [542]	24	✓		4450 [543]	1.0+
49.	4601 [555]	2	✓		–	3.0+
50.	4602IP [556]	2	✓		–	1.3+
51.	4602SW [556]	2	✓		–	1.3+
52.	4606 [544]	6	✓		–	1.0-3.2
53.	4610SW [549]	24 (4x6)	✓		–	3.0+
54.	4612 [545]	12	✓		–	1.0-3.2
55.	4620 [550]	24 (2x12)	✓		EU24BL [518]	1.4+
56.	4620SW [550]	24 (2x12)	✓		EU24BL [518]	1.4+
57.	4621 [550]	24 (2x12)	✓		EU24BL [518]	3.1+
58.	4624 [546]	24	✓		–	1.0-3.2
59.	4625 [550]	24 (2x12)	✓		EU24BL [518]	3.2+
60.	5402 [551]	2 (+12)	✓		–	3.0+
61.	5410 [552]	12 (2x6)	✓		–	3.0+
62.	5420 [553]	24 (3x8)	✓		EU24 [518]	3.0+
63.	5601 [555]	2	✓		–	3.0+
64.	5602IP [556]	2	✓		–	3.0+
65.	5602SW [556]	2	✓		–	3.0+
66.	5610SW [557]	24 (4x6)	✓		–	3.0+
67.	5620SW [558]	24 (2x12)	✓		EU24BL [518]	3.0+
68.	5621 [558]	24 (2x12)	✓		EU24BL [518]	3.2+
69.	6408D [559]	8	✓		–	1.0+
70.	6416D [559]	16	✓		XM24 [518]	1.0+
71.	6424D [560]	24	✓		XM24 [518]	1.0+
85.	9620L [562]	12	✓		–	6.0+
86.	9620G [562]	12	✓		–	6.0+
88.	9630G [562]	24	✓		SMB24 [518]	6.0+
89.	9640G [563]	24	✓		SMB24 [518]	6.0+
90.	9640 [563]	24	✓		SMB24 [518]	
92.	9650 [565]	24	✓		SMB24 [518]	6.0+
93.	9650C [565]	24	✓		SMB24 [518]	6.0+
116.	T3 Compact [566]	4	✓ [1]		T3 DSS [518]	3.1+
117.	T3 Classic [567]	10	✓ [1]		T3 DSS [518]	3.1+
118.	T3 Comfort [568]	22	✓ [1]		T3 DSS [518]	3.1+
119.	T3 IP Compact [566]	4	✓ [1]		T3 DSS [518]	3.2+
120.	T3 IP Classic [567]	10	✓ [1]		T3 DSS [518]	3.2+
121.	T3 IP Comfort [568]	22	✓ [1]		T3 DSS [518]	3.2+

1. Line appearances only.
2. Supported on systems running IP Office Essential Edition - PARTNER® Version and IP Office Essential Edition - Norstar Version modes only.

- A number of phones only have two physical programmable buttons. On systems running IP Office 3.0 or higher (excluding 3.0DT) these button should only be used as call appearance buttons.
- The 2402 and 5402 phones have an additional 12 programmable slots that can be accessed by pressing FEATURE.
- *T3 and T3 IP phones support line appearance buttons only and only for IP Office 4.2+.

7.2.1 Phone Add-Ons

The following add-on's can be used to provide some Avaya phones will additional programmable buttons.

- For IP Office 5.0 and higher, the maximum combined number of buttons on buttons modules per system is 1024.
- For IP Office 4.2 2Q 2009+, in addition to the individual limits above, the maximum combined number of buttons per system is 512, regardless of whether the buttons are programmed or not.
- T3 DSS modules are not included in the combined limits stated above but are limited to 30 T3 DSS modules (1080 buttons).

Additional buttons can be used for button programming actions 569 and in some cases as appearance buttons for key & lamp operation 674.

- Those marked ✓✗ symbol support programmable buttons but those buttons cannot be used as appearance buttons.
- Those marked ✓✓ symbol support programmable buttons which can be used as appearance buttons.

Button Module	Description	Per Phone	Per DS Module	Per System
1100 KEM: ✓✓ *+18*	Add-on for 1120E 524 and 1140E 524 phones. Each module provides 18 additional programmable buttons. Up to 3 modules can be added per phone.	3	–	56
1200 KEM: ✓✓ *+12*	Add-on for 1220 525 and 1230 525 phones. Each module provides 12 additional programmable buttons. Up to 7 modules can be added per phone.	7	–	85
20DS: ✓✗ *+42*	Add-on for 2030 530, 2050 530 and 20CC 530 phones. One per phone. Provides an additional 42 programmable buttons with dual-color lamps.	1	–	–
4450: ✓✗ *+60 (50x, 10x)*	Add-on for 4412D+ 541 and 4424D+ 542 phones. Provides an additional 60 programmable buttons with a single lamp - red except for the bottom two rows which are green. Due to the single lamp not recommended for appearance functions as not all button states can be indicated.	2	2	8
BM32: ✓✗ *+32*	Add-on for the 1616 529 phones that provides two columns of 16 buttons. Up to 3 BM32 modules are supported with any 1616. For IP Office 5.0, up to a maximum of 32 BM32 modules total are supported the system. For pre-IP Office 5 systems, only a total of 16 BM32 modules is supported.	3	–	32
DBM32: ✓✗ *+32*	Similar to the BM32 but used for the 1416 527 phone.	3	–	32
EU24: ✓✓ *+24 (2x12)*	Add-on for the 2420, 4620, 4620SW, 4625, 5420, 5620, 5620SW and 5621. Provides an additional 24 programmable buttons. Button display icons are on two switchable pages with 12 icons on each page. Connects direct to the phone. One per phone. Maximum of eight per system.	1	–	8
EU24BL: ✓✓ *+24 (2x12)*	As per the EU24 above but with a backlight function to match the 4621. Not supported on the 2420 and 5420.	1	–	8
SMB12: ✓✗ *+24*	Supported with 9600 Series phones except 9620 variants. SMB12 and SMB24 modules cannot be mixed on the same phone.	3	–	42
SMB24: ✓✗ *+24*		3	–	42
T3 DSS: ✓✓ *+36 LED*	Up to 3 of these units can be connected to any of the system T3 phones. Each provides an additional 36 programmable buttons. Each button includes a single red status LED. Connection of the T3 DSS varies: With non-IP models, the first T3 DSS connects directly to a link port on the phone. No additional power supplies are required. With T3 IP Models, the first T3 DSS connects to a DSS Link Unit fitted to the phone. A power supply is required for the DSS.	3	–	30

Button Module	Description	Per Phone	Per DS Module	Per System
XM24: ✓✓ *+24*	Add-on for 6416D 559 and 6424D 560 phones. Provides an additional 24 programmable buttons. Connects direct to phone. One per phone. Maximum of two per system DS module or control unit.	1	2	21

7.2.2 Button Types

Key and lamp operation requires Avaya phones with programmable buttons. These are found on the majority of Avaya phones. The list below shows different programmable button types and the phones on which they are found.

- 1603, 1608, 1616
  Programmable button with twin lamps.
- 2030
  Programmable button with no lamp.
- 2050, 20CC, 20DS
  Programmable button with dual-color lamp (red or green).
- 4406D, 4412D, 4424D, 4606, 4612, 4624 Phones
  Programmable button with twin lamps. For appearance functions the red lamp is used to indicate the current selected button.
- 4412 Phone
  Programmable button with no lamps. Found on the last 12 buttons on the Avaya 4412 phone.
- / 4450 Phone Add-On
  Programmable buttons with a single lamp (either red or green). Not recommended for appearance functions as not all button states can be indicated.
- 4601, 5601
  Programmable button with single red lamp.
- 6408D, 6416D, 6424D Phones and XM24 Add-On
  Programmable button with twin lamps. For appearance functions the red lamp is used to indicate the current selected button.
- 2410D, 2420, 5410, 5420, EU24
  Display key with adjacent icon and text label display area. For appearance functions an _ underscore on the text label is used to indicate the current selected button.
- 4610SW, 4620, 4621, 4625, 5610SW, 5620, 5621
  Display key with adjacent icon and text label display area. For appearance functions a * symbol is used to indicate the current selected button.
  2402D, 4602, 4602SW, 5402, 5602, 5602SW Phones
  Display key with adjacent icon display area but no text label display. For appearance functions a * symbol is used to indicate the current selected button.
  - The 2402D and 5402 have an additional 12 programmable buttons. These are accessed by pressing FEATURE and then any key from 0 to 9, * and #. These additional buttons are not suitable for appearance functions.
- Transtalk 9040
  Display keys with icon display area above. For appearance functions a icon is used to indicate the current selected button.
- 3616, 3620, 3626
  Accessed by pressing LINE followed by the required button number.
- 3641, 3645
  Accessed by pressing LINE or FCN followed by the required button number.
- / T3 Compact, T3 Classic and T3 Comfort
  4, 10 and 18 buttons respectively which a combination of buttons having indicator lamps. On each phone 0, 4 and 10 of those button are located below the display and use display icons to indicate their status if applicable.

7.2.3 Status Lamps

The following table summarizes the different status indications used by programmable buttons with lamps or icon indicators. This does include appearance functions which are covered in the Key and Lamp Operation 674 section.

Button Type	2400, 5400	4600, 5600	4400, 6400	1400, 1600, 9600	Status
AD Suppress 578	Spres◀	Spres	Green on	Green on	On
	Spres	Spres	Off	Off	Off
Automatic Callback 580	AutCB◀	AutCB	Green on	Green on	On
	AutCB	AutCB	Off	Off	Off
Call Forwarding All 585	CFrwd◀	CFrwd	Green on	Green on	On
	CFrwd	CFrwd	Off	Off	Off
Call Park 588	Cpark◆	Cpark	Green flash	Green flash	Parked by extension.
	Cpark	Cpark	Red flash	Red flash	Parked by other extension.
	Cpark	Cpark	Off	Off	No parked calls.
Call Park To Other Extension 589	Rpark◆	Rpark	Green flash	Green flash	Parked call.
	Rpark	Rpark	Off	Off	No parked call.
Call Waiting On 594	CWOn◀	CWOn	Green on	Green on	On
	CWOn	CWOn	Off	Off	Off
Do Not Disturb On 617	DNDOn◀	DNDOn	Green on	Green on	On
	DNDOn	DNDOn	Off	Off	Off
Follow Me To 622	FolTo◀	FolTo	Green on	Green on	On
	FolTo	FolTo	Off	Off	Off
Forward Hunt Group Calls On 623	FwdH+◀	FwDH+	Green on	Green on	On
	FwdH+	FwDH+	Off	Off	Off
Forward On Busy On 626	FwBOn◀	FwBOn	Green on	Green on	On
	FwBOn	FwBOn	Off	Off	Off
Forward On No Answer On 627	FwNOn◀	FwNOn	Green on	Green on	On
	FwNOn	FwNOn	Off	Off	Off
Forward Unconditional On 628	FwUOn◀	FwUOn	Green on	Green on	On
	FwUOn	FwUOn	Off	Off	Off
Group 629	Main	Main	Off	Off	No calls.
	Main◆	Main◆	Green flash	Green flash	Call alerting.
	Main	Main	Red flash	Red flash	Calls queued.
Hunt Group Enable 634	HGEna◀	HGEna	Green on	Green on	On
	HGEna	HGEna	Off	Off	Off
Internal Auto-Answer 637	HFAns◀	HFAns	Green on	Green on	On
	HFAns	HFAns	Off	Off	Off
Off Hook Station 641	OHStn◀	OHStn	Green on	Green on	On
	OHStn	OHStn	Off	Off	Off
Park Call 643	PARK1	PARK1	Off	Off	No parked call.
	PARK1◆	PARK1◆	Green flash	Green flash	Parked here.
	PARK1	PARK1	Red flash	Red flash	Parked elsewhere.
Private Call 645	PrivC◀	PrivC	Green on	Green on	On
	PrivC	PrivC	Off	Off	Off
Ring Back When Free 649	AutCB◀	AutCB+	Green on	Green on	On
	AutCB	AutCB+	Off	Off	Off
Ringer Off 650	RngOf◀	RngOf	Green on	Green on	On (no ring)
	RngOf	RngOf	Off	Off	Off (ring)
Send All Calls 652	SAC◀	SAC	Green on	Green on	On

Button Type	2400, 5400	4600, 5600	4400, 6400	1400, 1600, 9600	Status
	SAC	SAC	Off	Off	Off
Set Hunt Group Night Service 654	HGNS+200◄	HGNS+200	Green on	Green on	On
	HGNS+200	HGNS+200	Off	Off	Off
Set Hunt Group Out Of Service 656	HGOS+200◄	HGOS+200	Green on	Green on	On
	HGOS+200	HGOS+200	Off	Off	Off
Time of Day 662	TmDay◄	TmDay	Green on	Green on	On
	TmDay	TmDay	Off	Off	Off
Timer 662	Timer◄	Timer	Green on	Green on	On
	Timer	Timer	Off	Off	Off
Twinning 663	Twinning◄	Twinning	Green on	Green on	On
	Twinning	Twinning	Off	Off	Off
	Twinning◄	Twinning◆	Red on	Red on	Twinned call at secondary.
User 666	Extn221	Extn221	Off	Off	Idle
	Extn221◄	Extn221◆	Green flash	Red flash	Alerting
	Extn221	Extn221	Green on	Red wink	In Use/Busy
	Extn221	Extn221	Green on	Red on	DND
Voicemail On 670	VMOn◄	VMOn	Green on	Green on	On
	VMOn	VMOn	Off	Off	Off
Voicemail Ringback On 671	VMRB+◄	VMRB+	Green on	Green on	On
	VMRB+	VMRB+	Off	Off	Off

7.2.4 1000 Series

The 1000 Series phones are high-quality SIP video phone devices. The 1010 and 1040 phones are supported. Each consists of a main module to which a range of video camera and microphone/speaker devices can be attached. The main module provides outputs for display of video on HD video compatible devices.

7.2.5 1100 Series

These Avaya SIP phones are supported on IP500 and IP500v2 systems running IP Office 6.1 and higher. They are currently only supported when using the system as the file server for phone firmware and other files.

Expansion modules providing an additional 18 programmable buttons per module are supported. Up to 3 such modules are supported per phone.

1120E SIP Telephone

1140E SIP Telephone

7.2.6 1200 Series

These Avaya SIP phones are supported on IP500 and IP500v2 systems running IP Office 6.1 and higher. They are currently only supported when using the system as the file server for phone firmware and other files.

Expansion modules providing an additional 12 programmable buttons per module are supported. Up to 7 such modules are supported per phone.

1220 SIP Telephone

1230 SIP Telephone

7.2.7 1400 Series

7.2.7.1 1403

Manager is able to pass user button information to a DESI application on the same PC. This allows printing of labels using the label text set within the system configuration. Currently only ETR, 1400 and 1600 phonesphones are supported by DESI.

Note: For IP Office Essential Edition - PARTNER® Version and IP Office Essential Edition - Norstar Version mode systems, the programmable button order varies according to the system configuration. Refer to the PARTNER Version documentation for details.

01 02 03

As these phones only have 3 programmable buttons, those buttons should only be used for call appearance buttons.

- 1403: IP Office 6.0+.

7.2.7.2 1408

Manager is able to pass user button information to a DESI application on the same PC. This allows printing of labels using the label text set within the system configuration. Currently only ETR, 1400 and 1600 phonesphones are supported by DESI.

Note: For IP Office Essential Edition - PARTNER® Version and IP Office Essential Edition - Norstar Version mode systems, the programmable button order varies according to the system configuration. Refer to the PARTNER Version documentation for details.

01 02 03 04

05 06 07 08

The appearance buttons on these phones must be programmed as a continuous block starting from button 1.

- 1408: IP Office 6.0+.

7.2.7.3 1416

For 1416 phones, additional buttons can be added using DBM32 modules.

Manager is able to pass user button information to a DESI application on the same PC. This allows printing of labels using the label text set within the system configuration. Currently only ETR, 1400 and 1600 phonesphones are supported by DESI.

Note: For IP Office Essential Edition - PARTNER® Version and IP Office Essential Edition - Norstar Version mode systems, the programmable button order varies according to the system configuration. Refer to the PARTNER Version documentation for details.

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15
16

The appearance buttons on these phones must be programmed as a continuous block starting from button 1.

- 1416: IP Office 6.0+.

7.2.8 1600 Series

7.2.8.1 1603

Manager is able to pass user button information to a DESI application on the same PC. This allows printing of labels using the label text set within the system configuration. Currently only ETR, 1400 and 1600 phonesphones are supported by DESI.

01 02 03

As these phones only have 3 programmable buttons, those buttons should only be used for call appearance buttons.

- 1603: system 4.2 Q4 2008+.
- 1603SW: IP Office 5.
- 1603I: IP Office 6.0+.
- 1603I SW: IP Office 6.0+.

7.2.8.2 1608

Manager is able to pass user button information to a DESI application on the same PC. This allows printing of labels using the label text set within the system configuration. Currently only ETR, 1400 and 1600 phonesphones are supported by DESI.

01 02 03 04

05 06 07 08

The appearance buttons on these phones must be programmed as a continuous block starting from button 1.

- 1608: system 4.2 Q4 2008+.
- 1608I: IP Office 6.0+.

7.2.8.3 1616

For 1616 phones, additional buttons can be added using a BM32 module. The use of a BM32 requires the phone to be powered by a 1600 Series power supply unit rather than PoE.

Manager is able to pass user button information to a DESI application on the same PC. This allows printing of labels using the label text set within the system configuration. Currently only ETR, 1400 and 1600 phonesphones are supported by DESI.

01
02
03
04
05
06
07
08

09
10
11
12
13
14
15
16

The appearance buttons on these phones must be programmed as a continuous block starting from button 1.

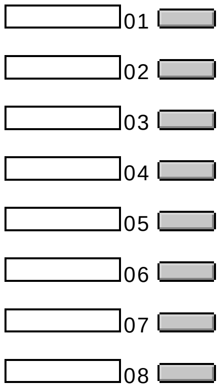
- 1616: system 4.2 Q4 2008+.
- 1616I: IP Office 6.0+.

7.2.9 20 Series

7.2.9.1 2030

This phone has 8 programmable buttons. They do not include any status indicator lamps.

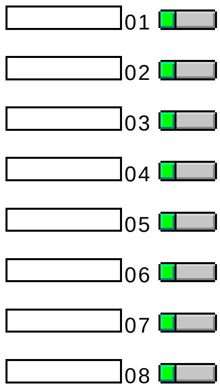
- 2030: Up to system 3.0DT.



7.2.9.2 2050 and 20CC

These phones have 8 programmable buttons. Each button includes a dual status indicator lamp.

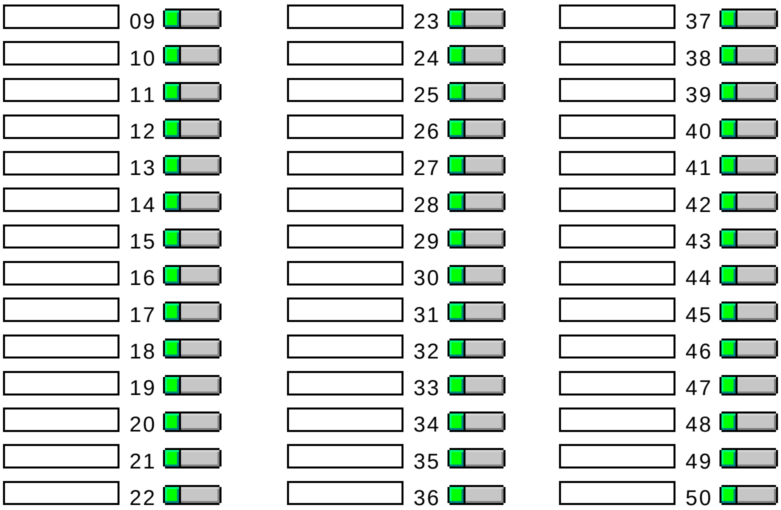
- 2050: Up to system 3.0DT only.
- 20CC: Up to system 3.0DT only.



7.2.9.3 20DS

This add-on is used with 2030, 2050 and 20CC phones. It provides an additional 42 programmable buttons with dual status indicator lamps. The unit requires a power supply and is connected to the next DT port on the same expansion module as the phone with which it is associated.

- 20DS: Up to system 3.0DT only.



7.2.10 2400 Series

7.2.10.1 2402D

These phones only have two physical programmable buttons with no display text labels. For systems running system 3.0 or higher (excluding 3.0DT) these buttons should only be used for call appearance buttons.

Display icons are used for status indication with a * used for current selected appearance button indication. For appearance functions, these only display active, alerting, held here and current selected button. They do not display in use elsewhere and on hold elsewhere.

Another 12 programmable buttons (buttons 4 to 15) are accessed by the user pressing FEATURE and then any key from 0 to 9, * and #.

- 2402D: system 3.0+.

1

2

4 = FEATURE + 1

5 = FEATURE + 2

6 = FEATURE + 3

7 = FEATURE + 4

8 = FEATURE + 5

9 = FEATURE + 6

10 = FEATURE + 7

11 = FEATURE + 8

12 = FEATURE + 9

13 = FEATURE + *

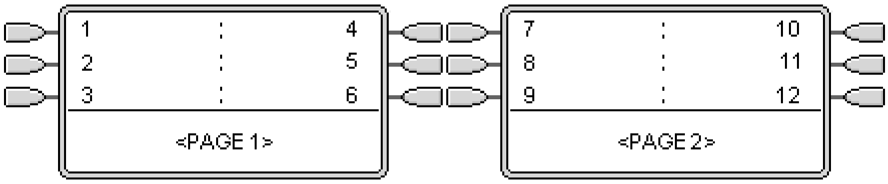
14 = FEATURE + 0

15 = FEATURE + #

7.2.10.2 2410

These phones have 6 physical display keys that provide 12 programmable buttons arranged in two pages. The ◄ and ► keys are used to switch the display between different button display pages as shown below.

- 2410: system 3.0+.



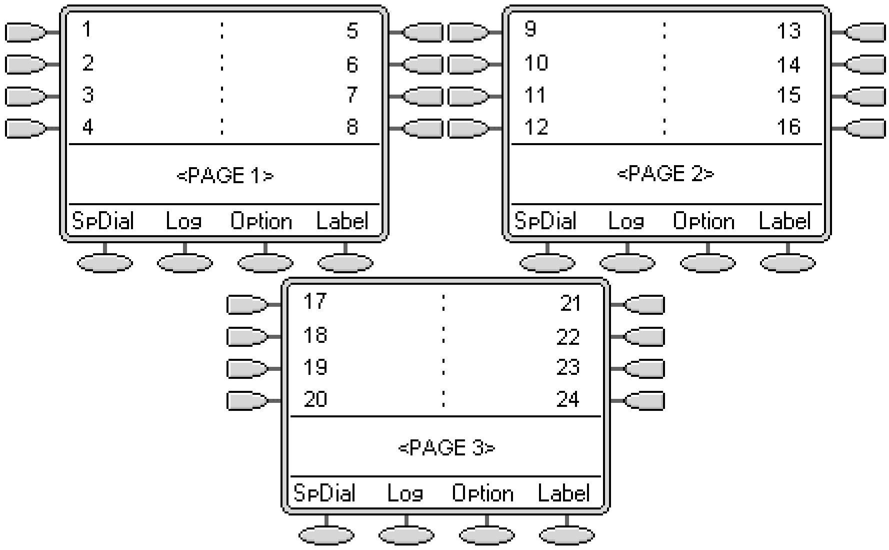
7.2.10.3 2420

These phones have 8 physical display keys and 24 programmable buttons. The ◀ and ▶ keys are used to switch the display between different button display pages as shown below.

These phones support two modes of button display, selected by the user through the phone (press Option -> Display Mode -> Call Center Mode).

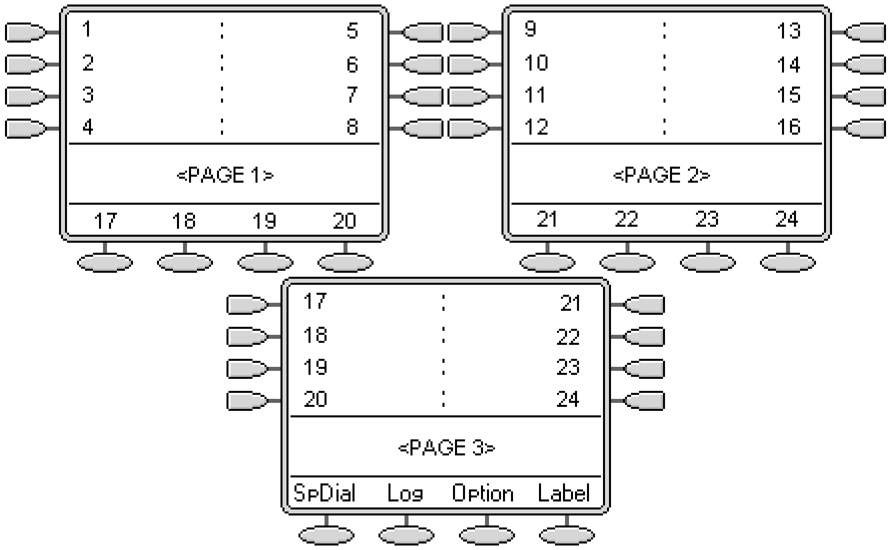
- 2420: system 2.1+.

Normal Mode



Call Center Mode

In this mode, several of the programmable button position are repeated and replace the normal functions on the base on the phone display.



7.2.11 3600 Series

7.2.11.1 3616, 3620, 3626

These phones support 6 programmable buttons. These are accessed by pressing LINE and then 1 to 6 when the phone is off hook. The FCN options are not re-programmable.

- 3616: system 2.1+.
- 3626: system 3.2+.
- 3620: system 3.2+.

system Button	3616/3626 Button
1	LINE + 1
2	LINE + 2
3	LINE + 3
4	LINE + 4
5	LINE + 5
6	LINE + 6

The following should be noted with these phones:

- 'In use' is indicated by the button number being shown. The same indication is used for 'In use elsewhere' and 'On Hold elsewhere' are not distinguished from 'In use'.
- 'Alerting' and 'Held here' are indicated by the button number flashing.
- No indication is shown when the 3616 or 3626 is idle. Only an alerting call will override idle.
- When off hook, the phones do not give any abbreviated ringing to indicate an alerting button.

7.2.11.2 3641, 3645

These phones support up to 12 programmable buttons (the system option within the phone menu must be selected, otherwise only 6 programmable buttons are provided). These are accessed by pressing LINE or FCN and then 1 to 6 when the phone is off hook.

- 3641/3645: system 4.0+.*
  *Supported from the system 4.0 Q2 2007 Maintenance release onwards.

system Button	3641/3645 Button
1	LINE 1
2	LINE 2
3	LINE 3
4	LINE 4
5	LINE 5
6	LINE 6
7	FCN 1
8	FCN 2
9	FCN 3
10	FCN 4
11	FCN 5
12	FCN 6

The following should be noted with these phones:

- 'In use' is indicated by the button number being shown. The same indication is used for 'In use elsewhere' and 'On Hold elsewhere' are not distinguished from 'In use'.
- 'Alerting' and 'Held here' are indicated by the button number flashing.
- No indication is shown when the 3616 or 3626 is idle. Only an alerting call will override idle.
- When off hook, the phones do not give any abbreviated ringing to indicate an alerting button.

7.2.12 3700 Series

The 3700 Series phones are a range of DECT handsets.

- The 3701 and 3711 are supported with the Avaya IP DECT system. This is supported by IP Office 3.1 and higher. They can also be used with the Avaya DECT R4 system but only as GAP compatible handsets.
- The 3720 and 3725 are supported with the Avaya DECT R4 system. This is supported by IP Office 5.0 and higher.

7.2.13 3800 Series

This phone supports 4 programmable buttons.

- 3810: system 2.1+.

system Button	3810 Button
8	1
9	2
1	3
2	4

Note that the correspondence between system button numbers and the phone buttons is not logical. If call appearance buttons are programmed, only the first two will be useable and will appear as buttons 3 and 4 on the actual phone.

Line, bridged and call appearance buttons would need to be programmed as system buttons 8 and 9 and would appear on buttons 1 and 2 on the phone. If all four buttons are required as call appearances, then buttons 1 to 9 must be programmed as call appearance buttons in the configuration.

7.2.14 3900 Series

This single station DECT phone is supported on an IP500v2 control unit running in IP Office Essential Edition - PARTNER® Version mode. The base station uses an ETR port for connection to the system. The 3910 is no longer available from Avaya and has been superseded by the 3920.

3910

3920

7.2.15 4400 Series

7.2.15.1 4406D+

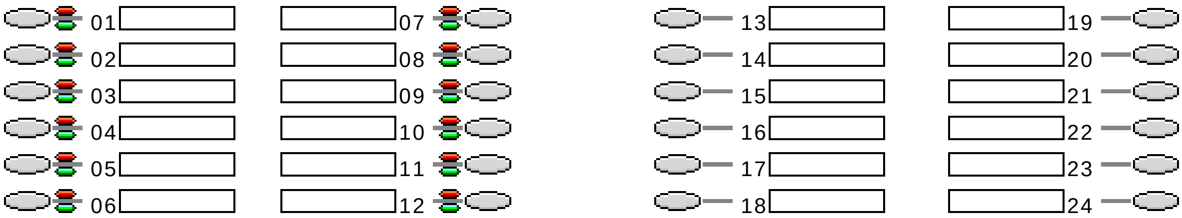
This phone has 6 programmable buttons with twin lamps; one green, one red.

- 4406D+: system 2.1+.

7.2.15.2 4412D+

This phone has 24 programmable buttons. The first 12 have twin lamps; one green, one red. The last 12 buttons do not include lamps.

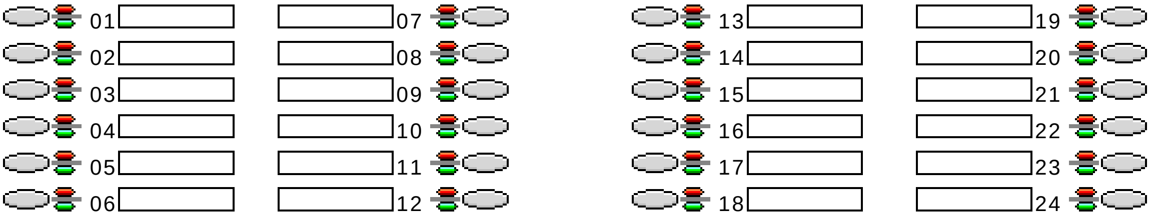
- 4412D+: system 2.1+.



7.2.15.3 4424D+

This phone has 24 programmable buttons with twin lamps; one green, one red. Note: No more than 10 4424D+ units are supported on any DS30 module. This restriction does not apply to DS16 V2 units.

- 4424D+: system 2.1+.



7.2.15.4 4450

Add-on for 4412D+ and 4424D+ phones. Provides an additional 60 programmable buttons with single lamps. Up to two 4450 units can be connected to an existing phone. Not recommended for appearance functions as the current selected button and on hold elsewhere are not indicated. A maximum of 2 unit as supported per DS module or control unit. A maximum of 10 units total are supported on a system. The 4450 buttons cannot indicate on hold elsewhere.

- 4450: system 1.0+.

In the button maps below, note that the bottom two rows break the normal pattern of button numbering.

4450x1

025	035	045	055	065
026	036	046	056	066
027	037	047	057	067
028	038	048	058	068
029	039	049	059	069
030	040	050	060	070
031	041	051	061	071
032	042	052	062	072
033	043	053	063	073
034	044	054	064	074
075	077	079	081	083
076	078	080	082	084

4450x2

085	095	105	115	125
086	096	106	116	126
087	097	107	117	127
088	098	108	118	128
089	099	109	119	129
090	100	110	120	130
091	101	111	121	131
092	102	112	122	132
093	103	113	123	133
094	104	114	124	134
135	137	139	141	143
136	138	140	142	144

7.2.16 4600 Series (Old Style)

7.2.16.1 4606

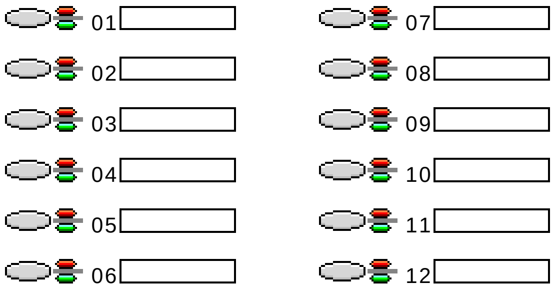
This phone has 6 programmable buttons with twin lamps; one green, one red.

- 4606: Up to system 3.2 only.

7.2.16.2 4612

This phone has 12 programmable buttons with twin lamps; one green, one red.

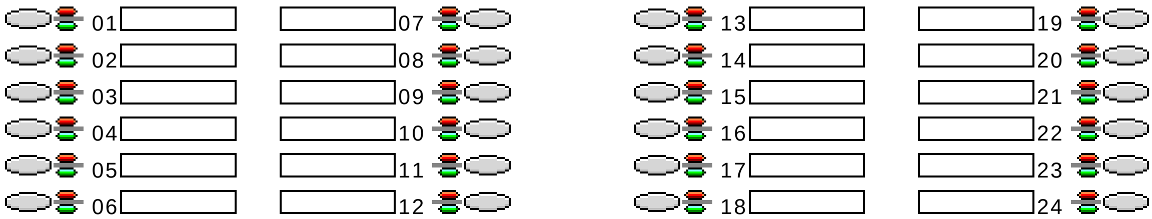
- 4612: Up to system 3.2 only.



7.2.16.3 4624

This phone has 24 programmable buttons with twin lamps; one green, one red.

- 4624: Up to system 3.2 only.



7.2.17 4600 Series (New Style)

7.2.17.1 4601

This phone has two programmable buttons, each with a single red lamp. For systems running system 3.0 or higher (excluding 3.0DT), these buttons should only be used for call appearance buttons.

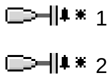
- 4601: system 3.0+.

7.2.17.2 4602, 4602SW

These phones only have two programmable buttons. For systems running system 3.0 or higher (excluding 3.0DT) these buttons should only be used for call appearance buttons.

Display icons are used for status indication with a * used for current selected appearance button indication. These only display active, alerting, held here and current selected button. They do not display in use elsewhere and on hold elsewhere.

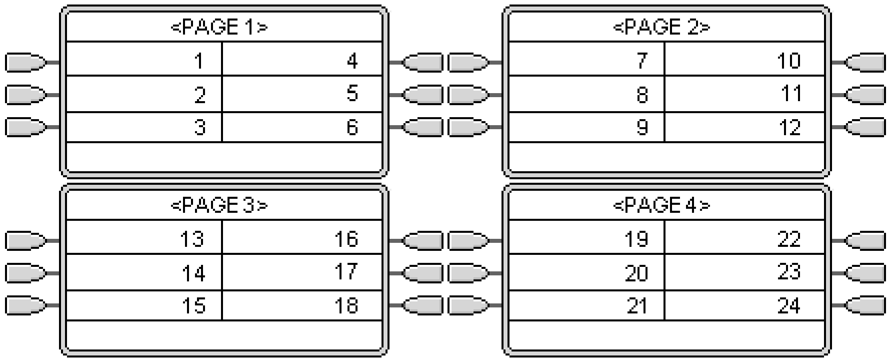
- 4602IP/4602SW: system 2.1+.



7.2.17.3 4610SW

This phone has 6 physical display keys that provide 24 programmable buttons arranged in 4 pages. The ◄ and ► keys are used to switch the display between different button display pages as shown below.

- 4610: system 3.1+.



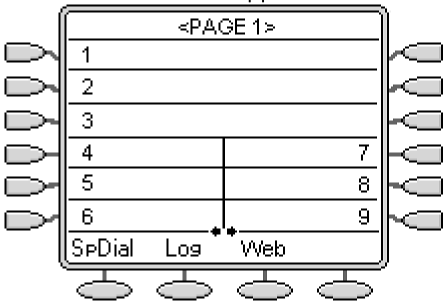
7.2.17.4 4620, 4620SW, 4621, 4625

These phones have 12 physical display keys that provide 24 programmable buttons arranged in two pages. The and keys are used to switch the display between different button display pages as shown below.

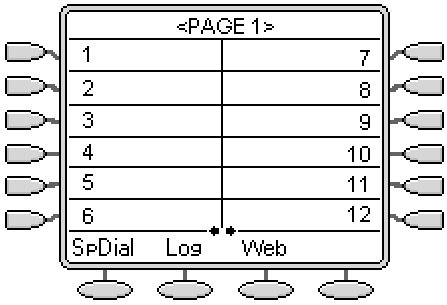
- 4620/4620SW: system 2.1+.
- 4621: system 3.1+.
- 4625SW: system 3.2+.

By default appearance actions use a full screen width display line and can be accessed by the display key on either side. The numbering of buttons and the number of display pages available is adjusted accordingly up 4 pages. The appearance buttons can be changed to half-width mode from the phone by pressing Options -> Application Options -> Call Appearance Width.

Button numbering with call appearance width set to full width mode and 3 appearance buttons.



Button numbering with call appearance width set to half width mode.



7.2.18 5400 Series

7.2.18.1 5402D

These phones only have two physical programmable buttons with no display text labels. For systems running system 3.0 or higher (excluding 3.0DT) these buttons should only be used for call appearance buttons.

Display icons are used for status indication with a * used for current selected appearance button indication. For appearance functions, these only display active, alerting, held here and current selected button. They do not display in use elsewhere and on hold elsewhere.

Another 12 programmable buttons (buttons 4 to 15) are accessed by the user pressing FEATURE and then any key from 0 to 9, * and #.

- 5402D: system 3.0+.

1

2

4 = FEATURE + 1

5 = FEATURE + 2

6 = FEATURE + 3

7 = FEATURE + 4

8 = FEATURE + 5

9 = FEATURE + 6

10 = FEATURE + 7

11 = FEATURE + 8

12 = FEATURE + 9

13 = FEATURE + *

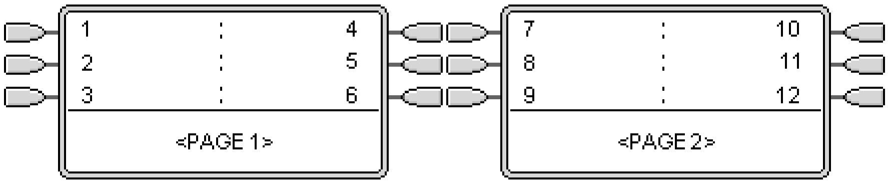
14 = FEATURE + 0

15 = FEATURE + #

7.2.18.2 5410

These phones have 6 physical display keys that provide 12 programmable buttons arranged in two pages. The and keys are used to switch the display between different button display pages as shown below.

- 5410: system 3.0+.



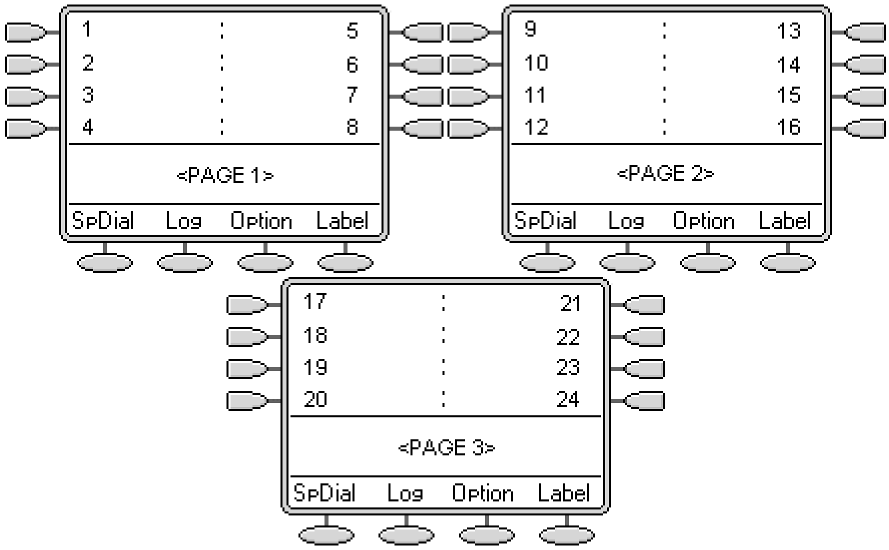
7.2.18.3 5420

These phones have 8 physical display keys and 24 programmable buttons. The ◀ and ▶ keys are used to switch the display between different button display pages as shown below.

These phones support two modes of button display, selected by the user through the phone (press Option -> Display Mode -> Call Center Mode).

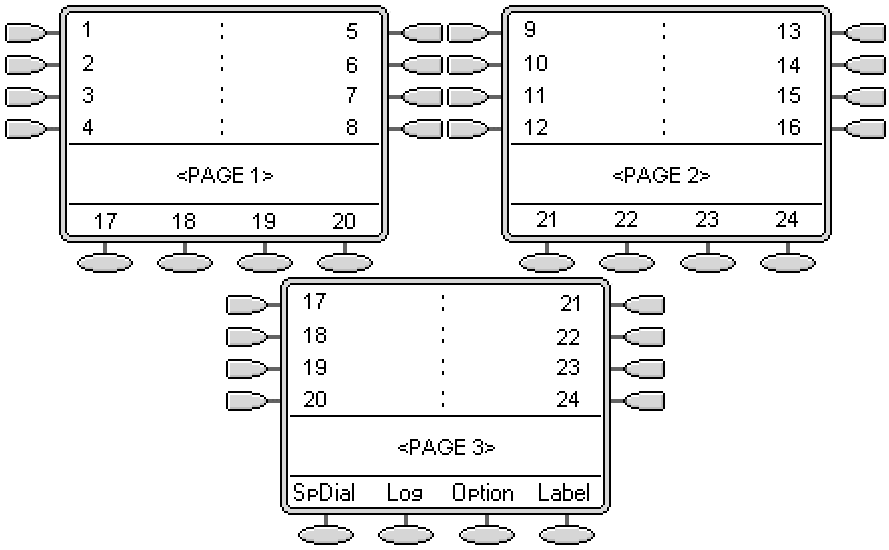
- 5420: system 3.0+.

Normal Mode



Call Center Mode

In this mode, several of the programmable button position are repeated and replace the normal functions on the base on the phone display.



7.2.19 5600 Series

7.2.19.1 5601

This phone has two programmable buttons, each with a single red lamp. For systems running system 3.0 or higher (excluding 3.0DT), these buttons should only be used for call appearance buttons.

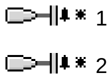
- 5601: system 3.0+.

7.2.19.2 5602, 5602SW

These phones only have two programmable buttons. For systems running system 3.0 or higher (excluding 3.0DT) these buttons should only be used for call appearance buttons.

Display icons are used for status indication with a * used for current selected appearance button indication. These only display active, alerting, held here and current selected button. They do not display in use elsewhere and on hold elsewhere.

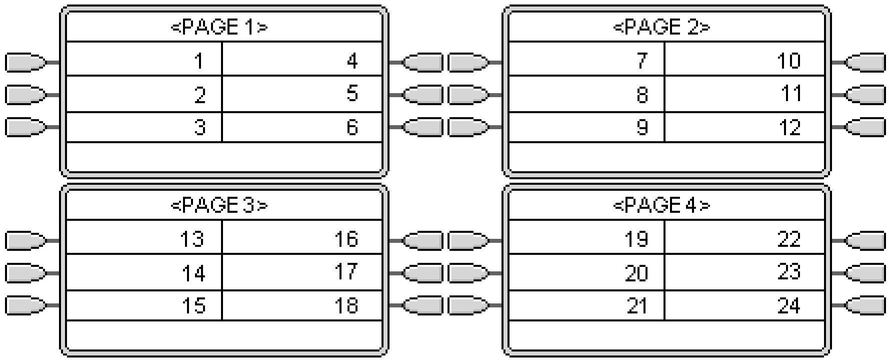
- 5602IP/5602SW: system 3.0+.



7.2.19.3 5610SW

This phone has 6 physical display keys that provide 24 programmable buttons arranged in 4 pages. The ◄ and ► keys are used to switch the display between different button display pages as shown below.

- 5610: system 3.1+.



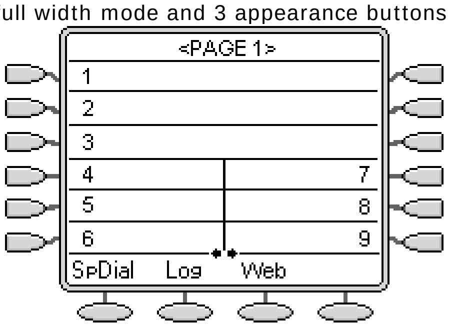
7.2.19.4 5620, 5620SW, 5621

These phones have 12 physical display keys that provide 24 programmable buttons arranged in two pages. The and keys are used to switch the display between different button display pages as shown below.

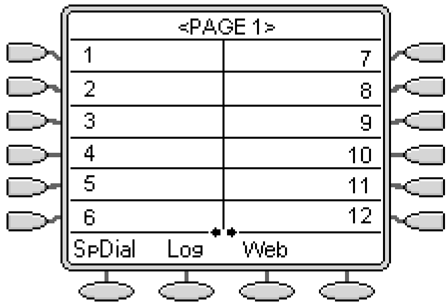
- 5620/5620SW: system 3.1+.
- 5621: system 3.2+.

By default appearance actions use a full screen width display line and can be accessed by the display key on either side. The numbering of buttons and the number of display pages available is adjusted accordingly up 4 pages. The appearance buttons can be changed to half-width mode from the phone by pressing Options -> Application Options -> Call Appearance Width.

Button numbering with call appearance width set to full width mode and 3 appearance buttons.



Button numbering with call appearance width set to half width mode.

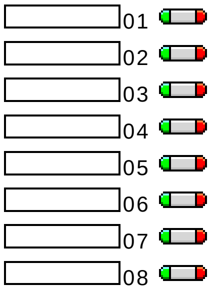


7.2.20 6400 Series

7.2.20.1 6408D

This phone has 8 programmable buttons. Each button includes twin lamps; one red, one green.

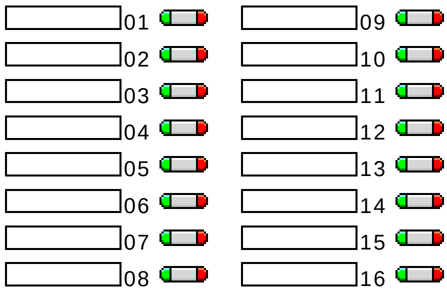
- 6408D: system 2.1+.



7.2.20.2 6416D

This phone has 16 programmable buttons. Each button includes twin lamps; one red, one green.

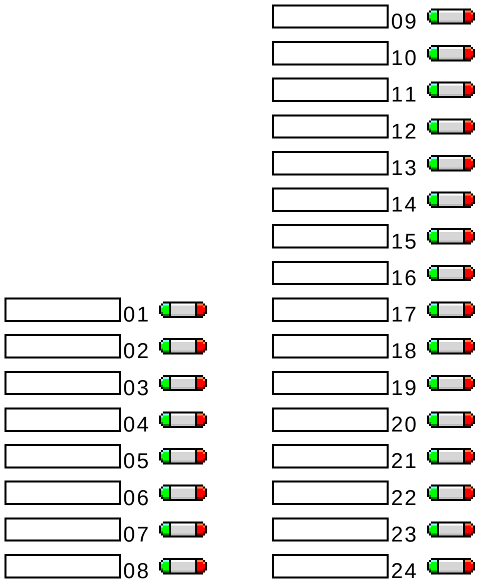
- 6416D: system 2.1+.



7.2.20.3 6424D

This phone has 24 programmable buttons. Each button includes twin lamps; one red, one green.

- 6424D: system 2.1+.



7.2.21 9000 Series

The 9040 has four display keys. These relate to two rows of display labels shown above, 1 to 4 and A to D. A < symbol is shown next to the currently selected row of labels, selection can be switched using the ^ shift key.

- 9040: Pre system 2.1 only.

9040 display buttons labeled 1 to 4 correspond with user buttons 1 to 4. 9040 display buttons labeled A to D correspond with user buttons 8 to 11.

It is only possible to use 9040 buttons 1, 2, A and B (buttons 1, 2, 8 and 9 in the system configuration) as appearance buttons. If all four of those buttons are required as call appearances, then buttons 1 to 9 must be programmed as call appearance buttons in the configuration. However button 3 and 4 on the handset will become disabled.

7.2.22 9600 Series

7.2.22.1 9620, 9620L

- These phones are supported by IP Office 6.0+ on IP500 and IP500v2 systems only.
- The voice activated dialing and USB features are not supported.

The 9620 variants supported are the 9620L and the 9620C. The phones support 12 programmable buttons but with no physical button keys. Button functions are accessed through the phone's display menu with button status only indicated when the button is one of the 3 that can be displayed at any time.

These phones do not support any button modules.

7.2.22.2 9630G

- These phones are supported by IP Office 6.0+ on IP500 and IP500v2 systems only.
- The voice activated dialing and USB features are not supported.

The phones support 24 programmable buttons. These can be accessed by scrolling the display and selecting the required button function using one of the 6 button to the right of the display.

This phone can be used with up to 3 x SBM24 or 3 x SBM12 button module. Each module provides an additional 24 programmable buttons. Note that the different types of button module cannot be mixed on the same phone. Attaching button modules may change the phones PoE class and may require a separate power supply.

7.2.22.3 9640, 9640G

- These phones are supported by IP Office 6.0+ on IP500 and IP500v2 systems only.
- The voice activated dialing and USB features are not supported.

The phones support 24 programmable buttons. These can be accessed by scrolling the display and selecting the required button function using one of the 6 button to the right of the display.

This phone can be used with up to 3 x SBM24 or 3 x SBM12 button module. Each module provides an additional 24 programmable buttons. Note that the different types of button module cannot be mixed on the same phone. Attaching button modules may change the phones PoE class and may require a separate power supply.

7.2.22.4 9650, 9650C

- These phones are supported by IP Office 6.0+ on IP500 and IP500v2 systems only.
- The voice activated dialing and USB features are not supported.

The phones support 24 programmable buttons. These can be accessed by scrolling the display and selecting the required button function using one of the 3 button to the right of the display. In addition buttons 4 to 19 can be accessed using the 8 buttons below the display acting in two pages of 8 buttons, the page being selected by the SHIFT button.

This phone can be used with up to 3 x SBM24 or 3 x SBM12 button module. Each module provides an additional 24 programmable buttons. Note that the different types of button module cannot be mixed on the same phone. Attaching button modules may change the phones PoE class and may require a separate power supply.

7.2.23 T3 Series

7.2.23.1 T3 Compact/T3 IP Compact

This phone has eight buttons on the right-hand edge, however only 4 (those numbered below) are programmable buttons, the remainder are fixed feature buttons. Note that some of the programmable buttons have default features and, if not programmed to a specific function in the system configuration, they will perform their default function.

- T3 Compact: system 3.1+.
- T3 IP Compact: system 3.2+.

	Handsfree/Monitoring
01	Call List
02	Voicemail
03	Do Not Disturb
04	Forward Unconditional
	Menu
	Redial
	Hold

7.2.23.2 T3 Classic/T3 IP Classic

This phone supports 10 programmable buttons (those numbered below). Note that some of the programmable buttons have default features and, if not programmed to a specific function in the system configuration, they will perform their default function.

- T3 Classic: system 3.1+.
- T3 IP Classic: system 3.2+.

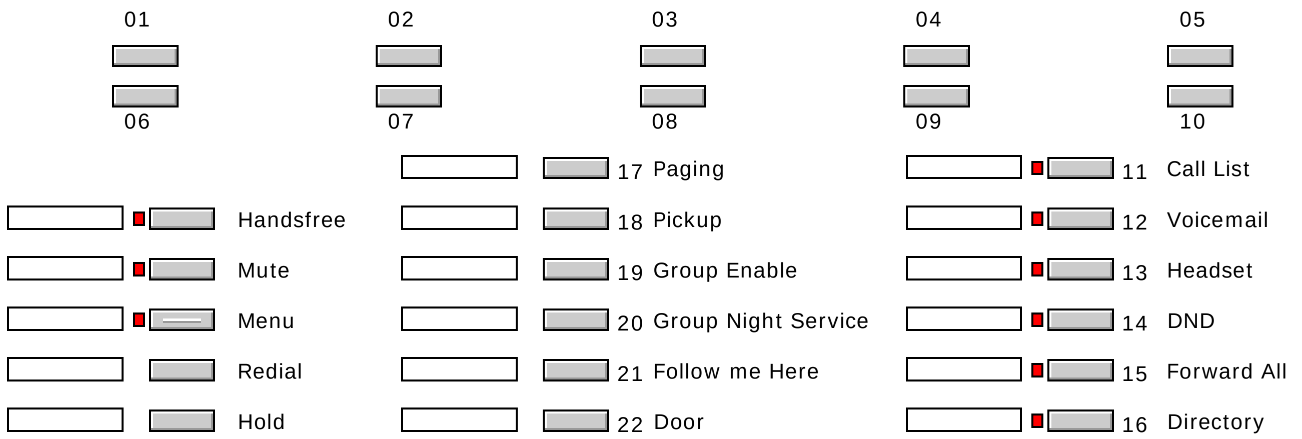
The four buttons below the display act as two pages of four using the display above. The four buttons on page one can be programmed and match system buttons 1 to 4. The other page cannot be programmed. In addition, some of the buttons in the two columns of buttons on the right can be programmed.

7.2.23.3 T3 Comfort/T3 IP Comfort

This phone supports 22 programmable buttons (those numbered below). Note that some of the programmable buttons have default features and, if not programmed to a specific function in the system configuration, they will perform their default function.

- T3 Comfort: system 3.1+.
- T3 IP Comfort: system 3.2+.

The ten buttons below the display act as eight pages of ten using the display above. The ten buttons on page one can be programmed and match system buttons 1 to 10. The other pages cannot be programmed. In addition, some of the buttons in the three columns of buttons on the right can be programmed.



7.3 Actions

The following sections provide details for each of the button actions supported by system. Note that this does not include buttons on phones on a system running in Partner Version mode.

For each the following details are listed:

- Software Level
  Indicates the software levels on which the action is supported.
- Action
  Indicates the selection path to the action from within the list of actions displayed in Manager.
- Action Data
  Indicates the type of data required by the action. For some actions no data is required while for others action data may be optional. The option to enter the data after pressing the button is not available for all phones, see Interactive Button Menus 515.
- Default Label
  This is the default text label displayed if possible on phones which provide a display area next to programmable buttons. Alternate labels can be specified in the system configuration or entered by the phone user (see Customizing Text Labels 514). Note that for buttons with action data set, the action data may also be displayed as part of the default label.
- Toggles
  Indicates whether the action toggles between two states, typically on or off.
- Status Indication
  Indicates whether the button provides status indication relevant to the feature if the button has status lamps or display. If the Status Indication is listed as *Required* it indicates that the button action is only supported on programmable buttons that can provide status indication.
- User Admin
  This item indicates that users with a Self-Administer 651 button can assign the action to other buttons themselves.
- Phone Support 516
  This is only a general indication of support or otherwise of an action by phones within particular series. On phones with 3 or less programmable buttons those button can only be used for the Call Appearance action. In addition some actions are only supported on phones where the programmable buttons provide status indication and or a display for data entry once the feature is invoked.

Table of Button Programming Actions

The following tables list the actions available for programmable buttons on system.

- Login Code Required
  Some function may require the user to enter their log in code. This typically applies when the action data is left blank for entry when the button is pressed.

General

Action	Action Data	Default Label
Dial 604	Any number.	Dial
Group 629	"Group name" in quote marks.	<Group name>
User 666	"User name" in quote marks.	<User name>

Appearance

Supported from IP Office 3.0 unless otherwise stated.

Action	Action Data	Default Label
Appearance 579	None.	a=
Bridged Appearance 583	User name and call appearance button number.	*<user name><appearance label>*
Coverage Appearance 603	User name.	*<user name>*
Line Appearance 639	Line appearance ID.	Line

Emulation

Action	Action Data	Default Label
Abbreviated Dial 573	Any number.	AD
Abbreviated Dial Pause 573	None.	Pause
Abbreviated Dial Program 574	None.	Prog
Abbreviated Dial Stop 574	None.	Stop
Account Code Entry 575	Account code or blank for entry when pressed.	Acct

Action	Action Data	Default Label
ACD Agent Statistics 576	None.	Stats
ACD Stroke Count 576	None.	Count
AD Special Function Mark 577	None.	Mark
AD Special Function Wait 578	None.	Wait
AD Special Functions 577	None.	Sfunc
AD Suppress 578	None.	Spres
Automatic Callback 580	None.	AutCB
Automatic Intercom 581	User number or name.	Iauto
Call Forwarding All 585	Any number or blank for entry when pressed.	CFrwd
Call Park 588	Park slot ID (alphanumeric) or blank for menu of slots in use.	CPark
Call Park To Other Extension 589	User number.	RPark
Call Pickup 590	None.	CpkUp
Cancel Leave Word Calling 596	None.	CnLWC
Consult 602	None.	Cnslt
Dial Intercom 608	User number or name or blank for entry when pressed.	Idial
Directed Call Pickup 614	User number or name or group number or name or or blank for entry when pressed..	DpkUp
Directory 615	None.	Dir
Drop 618	None.	Drop
Group Paging 631	User or group number or name or blank for entry when pressed.	GrpPg
Headset Toggle 632	None or FF *(IP Office 1.4+)*	HdSet
Inspect 636	None.	Inspt
Internal Auto-Answer 637	None.	HfAns
Leave Word Calling 638	None.	LWC
Manual Exclude 640	None.	Excl
Priority Calling 644	None.	Pcall
Ringer Off 650	None.	RngOf
Self-Administer 651	Blank or 1 or 2	Admin
Send All Calls 652	None.	SAC
Stored Number View 660	None.	BtnVu
Time of Day 662	None.	TmDay
Timer 662	None.	Timer
Twinning 663	None. *(IP Office 3.2+)*	Twinning
Visual Voice 668	None. *(IP Office 4.0+)*	Voice

Advanced

Supported from IP Office 1.1 unless otherwise stated.

Action	Action Data	Category	Default Label
Acquire Call 577	User number or blank for last call transferred.	Call	Acquire
After Call Work 579	*(IP Office 4.2+)*		
Break Out 581	System name or IP address or blank for selection when pressed. *(IP Office 4.0+)*	Dial	BkOut
Busy 584	None.	Busy	Busy
Busy On Held 584	0 (off) or 1 (on).	Busy	BusyH
Call Intrude 586	User number or blank for entry when pressed.	Call	Intru
Call List 586	None. *(IP Office 3.1+)*	Call	LIST
Call Listen 587	User number.	Call	Listn
Call Pickup Any 590	None.	Call	PickA
Call Pickup Group 591	None.	Call	PickG
Call Pickup Members 591	Group number or name.	Call	PickM
Call Queue 592	User number.	Call	Queue
Call Record 592	None.	Call	Recor
Call Steal 593	User number or blank for last call transferred.	Call	Steal

Action	Action Data	Category	Default Label
Call Waiting Off 594	None.	Call	CWOff
Call Waiting On 594	None.	Call	CWOn
Call Waiting Suspend 595	None.	Call	CWSus
Cancel All Forwarding 595	None.	Call	FwdOf
Cancel Ring Back When Free 596	None.	Miscellaneous	RBak-
Channel Monitor 597	Channel number.	Call	ChMon
Clear Call 598	None.	Call	Clear
Clear CW 598	None.	Call	ClrCW
Clear Hunt Group Night Service 599	Group number.	Call	HGNS-
Clear Hunt Group Out Of Service 599	Group number.	Call	HNOS-
Clear Quota 600	"Service name" within quote marks or "" for all services.	Call	Quota
Conference Add 600	None.	Call	Conf+
Conference Meet Me 601	Conference name or number.	Call	CnfRv
Dial 3K1 604	Any number.	Dial	D3K1
Dial 56K 605	Any number.	Dial	D56K
Dial 64K 605	Any number.	Dial	D64K
Dial CW 606	User number.	Dial	DCW
Dial Direct 606	User number or name or blank for entry when pressed.	Dial	Dirct
Dial Emergency 607	Any number.	Dial	Emrgy
Dial Inclusion 607	User number or name or blank for entry when pressed. *(IP Office 1.4+)*	Dial	Inclu
Dial Paging 609	User or group number or name or blank for entry when pressed.	Dial	Page
Dial Physical Extn by Number 610	Extension port Base Extension number. *(IP Office 1.4+)*	Dial	PhyEx
Dial Physical Extn by ID 610	Extension port ID number. *(IP Office 1.4+)*	Dial	DialP
Dial Speech 611	Any number.	Dial	DSpch
Dial V110 611	Any number.	Dial	DV110
Dial V120 612	Any number.	Dial	DV120
Dial Video 613	Any number.	Dial	Dvide
Display Msg 612	Command string.	Dial	Displ
Do Not Disturb Exception Add 616	Any number.	Do Not Disturb	DNDX+
Do Not Disturb Exception Delete 616	Any number.	Do Not Disturb	DNDX-
Do Not Disturb Off 617	None.	Do Not Disturb	DNDOf
Do Not Disturb On 617	None.	Do Not Disturb	DNDOn
Extn Login 619	None.	Extension	Login
Extn Logout 619	None.	Extension	Logof
Flash Hook 620	None. *(IP Office 1.4+)*	Miscellaneous	Flash
Follow Me Here 621	User number.	Follow Me	Here+
Follow Me Here Cancel 621	User number or blank for entry when pressed.	Follow Me	Here-
Follow Me To 622	User name or user number or blank for entry when pressed.	Follow Me	FolTo
Forward Hunt Group Calls On 623	None.	Forward	FwdH+
Forward Hunt Group Calls Off 623	None.	Forward	FwdH-
Forward Number 624	Any number or blank for entry when pressed.	Forward	FwdNo
Forward On Busy Number 624	Any number or blank for entry when pressed.	Forward	FwBNo
Forward On Busy Off 626	None.	Forward	FwBOf
Forward On Busy On 626	None.	Forward	FwBOn

Action	Action Data	Category	Default Label
Forward On No Answer Off 627	None.	Forward	FwNOf
Forward On No Answer On 627	None.	Forward	FwNOn
Forward Unconditional Off 628	None.	Forward	FwUOf
Forward Unconditional On 628	None.	Forward	FwUOn
Group Listen On 630	None. *(IP Office 4.1+)*	Extension	GroupListenOn
Hold Call 632	ISDN Exchange slot number.	Hold	Hold
Hold CW 633	None.	Hold	HoldCW
Hold Music 633	None.	Hold	Music
Hunt Group Disable 635	Group number or name or blank for all groups.	Hunt Group	HGDis
Hunt Group Enable 634	Group number or name or blank for all groups.	Hunt Group	HGEna
MCID Activate 640	None. *(IP Office 4.0+)*	Miscellaneous	MCID
Off Hook Station 641	None.	Miscellaneous	OHStn
Park Call 588	Park slot number. *(IP Office 1.0 to 3.2+)*	Call	Park
Priority Call 644	User number or name.	Call	PCall
Private Call 645	None. *(IP Office 4.0+)*	Call	PrivC
Relay Off 646	1 or 2.	Relay	Rely-
Relay On 646	1 or 2.	Relay	Rely+
Relay Pulse 647	1 or 2.	Relay	Relay
Resume Call 648	ISDN Exchange slot number.	Call	Resum
Retrieve Call 648	ISDN Exchange slot number.	Call	Retriv
Ring Back When Free 649	None.	Miscellaneous	RBak+
Set Absent Text 653	String for selected message and custom text.	Set	Absnt
Set Account Code 654	Blank or valid account code. *(IP Office 2.1+)*	Set	Acct
Set Hunt Group Night Service 654	Group number.	Set	HGNS+
Set Hunt Group Out Of Service 656	Group number.	Set	HGOS+
Set Inside Call Seq 657	Value 0 to 10.	Set	ICSeq
Set Night Service Group 657	Group number. *(IP Office 4.2+)*	Set	SetNSG
Set No Answer Time 659	Time in seconds (range 6 to 99999).	Set	NATim
Set Outside Call Seq 659	Value 0 to 10.	Set	OCSeq
Set Out of Service Group 658	Group number. *(IP Office 4.2+)*	Set	SetOOSG
Set Ringback Seq 659	Value 0 to 10.	Set	RBSeq
Set Wrap Up Time 660	Time in seconds (range 0 to 99999).	Set	WUTim
Suspend Call 661	ISDN Exchange slot number.	Suspend	Suspe
Suspend CW 661	ISDN Exchange slot number.	Suspend	SusCW
Toggle Calls 663	None.	Call	Toggl
Unpark Call 665	Park slot ID (alphanumeric).	Call	Ride
Voicemail Collect 669	See notes.	Voicemail	VMCol
Voicemail Off 670	None.	Voicemail	VMOff
Voicemail On 670	None.	Voicemail	VMOn
Voicemail Ringback Off 671	None.	Voicemail	VMRB-
Voicemail Ringback On 671	None.	Voicemail	VMRB+

7.3.1 Abbreviated Dial

This function allows quick dialing of a stored number.

- Software Level: 1.0+.
- Action: Emulation -> Abbreviated Dial.
- Action Data:
  - Full Number
    The number is dialled.
  - Partial Number
    The partial number is dialled and the user can then complete dialing the full number.
- Default Label: AD.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✓
  - 5400 Series: ✓[1]
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

  1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602 models.

7.3.2 Abbreviated Dial Pause

Not supported. Provided for CTI emulation only. Allows a user to enter a pause character when programming an abbreviated dial.

- Software Level: 1.0+.
- Action: Emulation -> Abbreviated Dial Pause.
- Action Data: *None.*
- Default Label: Pause.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.3 Abbreviated Dial Program

Not supported. Provided for CTI emulation only. Allows a user to program abbreviated dialing numbers against other programmable buttons. This function cannot be used to overwrite call appearance buttons.

- Software Level: 1.0+.
- Action: Emulation -> Abbreviated Dial Program.
- Action Data: *None*.
- Default Label: Prog.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✗
  - 1600 Series: ✗
  - 20 Series: ✗
  - 2400 Series: ✗
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✗
  - 5400 Series: ✗
  - 4600 Series: ✗
  - 5600 Series: ✗
  - 6400 Series: ✗
  - 9040: ✗
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗

7.3.4 Abbreviated Dial Stop

Not supported. Provided for CTI emulation only. Allows a user to enter a stop character when programming an abbreviated dial.

- Software Level: 1.0+.
- Action: Emulation -> Abbreviated Dial Stop.
- Action Data: *None*.
- Default Label: Stop.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.5 Account Code Entry

Enter an account code for a call. This button can be used before dialing a number or during a call.

- Software Level: 1.0+.
- Action: Emulation -> Account Code Entry.
- Action Data: Optional. If an code is set it must match an account code set in the account codes list. If no account code is set, the phone display will request entry of a valid code. This option is not supported on XX02 phones and the T7000 phone.
- Default Label: Acct.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays 1234.
  - DSS Link LED: None.

7.3.6 ACD Agent Statistics

Not supported. Provided for CTI emulation only.

- Software Level: 1.0+.
- Action: Emulation -> ACD Agent Statistics.
- Action Data: *None*.
- Default Label: Stats.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.7 ACD Stroke Count

Not supported. Provided for CTI emulation only.

- Software Level: 1.0+.
- Action: Emulation -> ACD Stroke Count.
- Action Data: *None*.
- Default Label: Count.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.8 Acquire Call

See Call Steal 593.

7.3.9 AD Special Functions

Not supported. Provided for CTI emulation only. Allows a user to enter a special character (mark, pause suppress, wait) when entering an abbreviated dial.

- Software Level: 1.0+.
- Action: Emulation -> AD Special Functions.
- Action Data: *None*.
- Default Label: Sfunc.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.10 AD Special Function Mark

Not supported. Provided for CTI emulation only. Allows a user to enter a mark character when programming abbreviated dial.

- Software Level: 1.0+.
- Action: Emulation -> AD Special Function Mark.
- Action Data: *None*.
- Default Label: Mark.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.11 AD Special Function Wait

Not supported. Provided for CTI emulation only. Allows a user to enter a Wait for Dial Tone character when programming an abbreviated dial.

- Software Level: 1.0+.
- Action: Emulation -> AD Special Function Wait.
- Action Data: *None*.
- Default Label: Wait.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.12 AD Suppress

Suppresses the display of dialed digits on the telephone display. Dialed digits are replaced with an s character.

- Software Level: 1.0+.
- Action: Emulation -> AD Suppress.
- Action Data: *None*.
- Default Label: Spres.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.13 After Call Work

This button is used by users configured as an CCR Agent [276] (User | Telephony | Supervisor Settings [276])and working with the IP Office IP Office Customer Call Reporter application. It shows the CCR agent their current After Call Work (ACW) status and allow them to manually change status. While in ACW state, the agent will not receive hunt group calls.

CCR Agents can be automatically put into and taken out of ACW by the system if the user is configured for Automatic After Call Work [276] (User | Telephony | Supervisor Settings [276]). Those users must have an After Call Work button.

- Software Level: 4.2+.
- Action: Advanced -> Miscellaneous -> After Call Work
- Action Data: None.
- Default Label: ACWrk.
- Toggles: ✓.
- Status Indication: ✓ *Required*.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: [651] ✗.
- Phone Support [516]
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓*
  - 1600 Series: ✓*
  - 20 Series: ✗
  - 2400 Series: ✓*
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✗
  - 5400 Series: ✓*
  - 4600 Series: ✓*
  - 5600 Series: ✓*
  - 6400 Series: ✗
  - 9040: ✗
  - 9600 Series: ✓
  - T3/T3 IP Series: ✓

    - **: Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602.*

7.3.14 Appearance

Creates a call appearance button [675]. This can be used to answer and make calls. Users with multiple call appearance buttons can handle multiple calls.

Call appearance functions, assigned to buttons that do not have status lamps or icons, are automatically disabled until the user logs in at a phone with suitable buttons.

IP Office 3.2+: Appearance buttons can be set with a ring delay if required or to not ring. This does not affect the visual alerting displayed next to the button. The delay uses the user's Ring Delay [278] (User | Telephony | Multi-line Options [278]) setting.

- Software Level: 3.0+.
- Action: Appearance -> Appearance.
- Action Data: Optional text label.
- Default Label: a=.
- Toggles: ✗.
- Status Indication: ✓ *Required.* See Call Appearance Button Indication [679].
- User Admin: [651] ✗.
- Phone Support [516]
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✗
  - 2400 Series: ✓
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓
  - 5600 Series: ✓
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

7.3.15 Automatic Callback

Sets a ringback on the extension being called. When the target extension ends its current call, the ringback user is rung (for their set No Answer Time) and if they answer, a new call is made to the target extension.

Ringback can also be cleared using the Cancel Ring Back When Free 596 function.

- Software Level: 1.0+.
- Action: Emulation -> Automatic Callback.
- Action Data: *None*.
- Default Label: AutCB.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.16 Automatic Intercom

Call an extension and have the call automatically answered on speaker phone after 3 beeps. The extension called must support a handsfree speaker. If the extension does not have a handsfree microphone then the user must use the handset if they want to talk. If the extension is not free when called, the call is presented as a normal call on a call appearance button if available.

- IP Office 4.2 Q4 maintenance release and higher: This feature can be used as part of handsfree announced transfers 784.

- Software Level: 1.0+.
- Action: Emulation -> Automatic Intercom.
- Action Data: User number or name.
  - IP Office 4.0+: This field can be left blank for number entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: Iauto.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays followed by the set number.
  - DSS Link LED: None.

7.3.17 Break Out

This feature is usable within a system Small Community Network 810. It allows a user on one system in the network to specify that the following dialing be processed by another system on the network as if the user dialed it locally on that other system.

On phones with a multi-line display, if the target system is not specified in the button settings, a menu of the available systems in the network is displayed from which a selection can be made.

Pre-IP Office 5.0: This feature requires the IP Offices to have *Advanced Small Community Networking* licenses.

- Software Level: 4.0+.
- Action: Advanced -> Dial -> Break Out.
- Action Data: Optional. The system name or IP address of the required system can be specified. If no system name or IP address is set, on display phones a list of systems within the Small Community Network is displayed when the button is pressed.
- Default Label: BkOut.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 2400 Series: ✓[1]
  - 3600 Series: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓[1]
  - 4600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗

- 20 Series: ✔
- 3700 Series: ✘
- 3810: ✘
- 5600 Series: ✔[1]

1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602 models.

7.3.18 Bridged Appearance

Creates an appearance button that follows the state of another user's call appearance button. The bridged appearance can be used to make and answer calls on behalf of the call appearance user.

The bridged appearance button user must also have at least one call appearance button programmed.

Bridged appearance functions, assigned to buttons that do not have status lamps or icons, are automatically disabled until the user logs in at a phone with suitable buttons.

IP Office 3.2+: Appearance buttons can be set with a ring delay if required or to not ring. This does not affect the visual alerting displayed next to the button. The delay uses the user's Ring Delay 278 (User | Telephony | Multi-line Options 278) setting.

- Software Level: 3.0+.
- Action: Appearance -> Bridged Appearance.
- Action Data: User name and call appearance button number.
- Default Label: *<user name><call appearance label>*.
- Toggles: ✗.
- Status Indication: ✓ *Required*. See Bridge Appearance Button Indication 685.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✗
  - 2400 Series: ✓
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓
  - 5600 Series: ✓
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

7.3.19 Busy

Not used.

7.3.20 Busy On Held

When on, busy on held returns busy to new calls while the user has an existing call on hold.

While this feature can be used by users with appearance keys, it is not recommended as this overrides the basic call handling intent of appearance keys.

- Software Level: 1.1+.
- Action: Advanced -> Busy -> Busy on Held.
- Action Data: 1 for on, 0 for off.
- Default Label: BusyH.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.21 Call Forwarding All

Switches forward unconditional on and sets the forward number to the number specified or prompts the user to enter a number if none is specified.

- Software Level: 1.0+.
- Action: Emulation -> Call Forwarding All.
- Action Data: Telephone number or blank for entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
  - If blank, the phone displays the user's current forward number and allows it to be changed if required. This option is not supported on XX02 phone models.
  - IP Office 4.0+: If blank, user's with a log in code will be prompted to enter that code to use this function.
- Default Label: CFrwd.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.22 Call Intrude

Intrudes on the existing call of the specified target extension. All call parties are put into a conference and can talk. Use of this feature is subject to the Can Intrude status of the intruder and the Cannot be Intruded status of the other call parties.

If the target is idle, the function is changed to a normal call.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Intrude.
- Action Data: User number or blank for entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: Intru.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.23 Call List

This function is only supported for T3 phones. It provides access to a list of received calls.

- Software Level: 3.1+.
- Action: Advanced -> Call -> Call List.
- Action Data: *None*.
- Default Label: LIST.
- Toggles: ✗.
- Status Indication: ✓ *Required*.
- User Admin: 651 ✗.
- Phone Support 516
  This function is only supported for T3 phones.
- T3 Phones:
  - Classic/Comfort icon: Displays LIST.
  - DSS Link LED: On when calls are in the list. Flashes when new calls are in the list.

7.3.24 Call Listen

This feature allows a user to monitor another conversation without being heard. It requires the user being monitored to be a member of the group set as the button user's Monitor Group 276 (User | Telephony Supervisor Settings 276) in the systemconfiguration. The use of call listen is also controlled by the Can Intrude setting of the user and the Cannot Be Intruded settings of the target. It is not affected by the settings of the third party to the call if they are internal.

- Warning: The use of monitoring may be subject to local and national restrictions. This feature should only be used in compliance with those restrictions.
- Note: On pre-4.0 systems, IP phone extensions can be used to monitor but cannot be monitored.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Listen.
- Action Data: User number.
- Default Label: Listn.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.25 Call Park

Allows the user to park and unpark calls. The button can be used in two ways, either associated with a specified park slot number or unspecified.

- When associated with a specific park slot number, the button will park and unpark calls from that park slot and indicate when a call is parked in that park slot. Similarly the Park buttons within IP Office application (for example Phone Manager, SoftConsole and one-X Portal for IP Office) can be used to park, retrieve and indicate parked calls.
- When not associated with a specific park slot number, the button will park calls by assigning them a park slot number based on the users extension number. For example, for extension *XXX*, the first parked call is assigned to park slot *XXX0*, the next to *XXX1* and so on up to *XXX9*. The button will indicate when there are parked calls in any of those slots. On the T7000 phone, only a single automatic part slot *XXX0* is supported.
  - With a call connected, pressing the button will park that call using a park slot number assigned by the system based on the extension number.
  - With no call connected, pressing the button will display details of any calls parked by the extension and allow their retrieval.

- Software Level: 1.0+.
- Action: Emulation -> Call Park.
- Action Data: Optional. Either blank or a specific park slot number. Name ca
- Park slot IDs can be up to 9 digits in length. Names can also be used for application park slots.
- Default Label: CPark.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- Calls parked by extension	CPark◆	CPark	Green flash.
- Call Parked by other extension	CPark	CPark	Red flash.
- No parked calls	CPark	CPark	Off.

- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓[1]
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

  1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602.

7.3.26 Call Park To Other Extension

Allows the user to park their current call against another user's extension. The parked call indication on that extension is then activated according to the telephone type.

If the target extension has a Call Park button with no specific park slot number, the parked call will be indicated by that button and can be unparked from the list of parked calls shown when that button is pressed.

- Pre-IP Office 4.0: The park slot number assigned to the parked call is based on the number of the extension against which the call is being parked. For example, calls parked against extension 203 are assigned park slot ID 2030, 2031 and so on up to 2039 depending on the number of calls parked.
- IP Office 4.0+: The park slot number assigned to the parked call is based on the number of the extension parking the call. For example, calls parked by extension 201 are assigned the park slot ID 2010, 2011 and so on up to 2019 depending on the number of calls parked.
  - Software Level: 1.0+.
  - Action: Emulation -> Call Park to Other Extension.
  - Action Data: User number. For system 4.0+ this field can be left blank for number entry when pressed 515.
    - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
  - Default Label: RPark *("Park" in pre-4.0 system)*.
  - Toggles: ✗ Pre-4.0/✓ 4.0+.
  - Status Indication: ✗ Pre-4.0/✓ 4.0+.
    This is the status indication on the extension parking the call.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- Parked Call	RPark◆	RPark	Green flash.
- No parked call	RPark	RPark	Off.

  - User Admin: 651 ✓.
  - Phone Support 516
    Note that support for particular phone models is also dependant on the system software level.
    - Analog: ✗
    - 1400 Series: ✓[1]
    - 1600 Series: ✓[1]
    - 20 Series: ✓
    - 2400 Series: ✓[1]
    - 3600 Series: ✗
    - 3700 Series: ✗
    - 3810: ✓
    - 4400 Series: ✓
    - 5400 Series: ✓[1]
    - 4600 Series: ✓[1]
    - 5600 Series: ✓[1]
    - 6400 Series: ✓
    - 9040: ✓
    - 9600 Series: ✓
    - T3/T3 IP Series: ✗

      1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602.

7.3.27 Call Pickup

Answer an alerting call on the system.

- Software Level: 1.0+.
- Action: Emulation -> Call Pickup.
- Action Data: *None*.
- Default Label: CpkUp.
- Toggles: ✗.
- Status Indication: ✗.
- [User Admin:](651) ✓.
- [Phone Support](516)
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones: Displays a menu for entry of the extension number from which to pickup a call.
  - Classic/Comfort icon: Displays .
  - DSS Link LED: None.

7.3.28 Call Pickup Any

Pick up the first available ringing call on the system.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Pickup Any.
- Action Data: *None*.
- Default Label: PickA.
- Toggles: ✗.
- Status Indication: ✗.
- [User Admin:](651) ✗.
- [Phone Support](516)
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones: Displays a list of call ringing from which the user can select a call to answer.
  - Classic/Comfort icon: Displays .
  - DSS Link LED: None.

7.3.29 Call Pickup Group

Pick up a call ringing any hunt group of which the user is a member.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Pickup Group.
- Action Data: *None*.
- Default Label: PickG.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones: Displays a list of calls ringing the hunt group from which the user can select which call to answer.
  - Classic/Comfort icon: Displays followed by group name.
  - DSS Link LED: None.

7.3.30 Call Pickup Members

This feature can be used to pick up any call to an extension that is a member of the hunt group specified. The call picked up does not have to be a hunt group call.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Pickup Members.
- Action Data: Group number or name.
- Default Label: PickM.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones: Displays a list of calls ringing the hunt group from which the user can select which call to answer.
  - Classic/Comfort icon: Displays followed by group name.
  - DSS Link LED: None.

7.3.31 Call Queue

Transfer the call to the target extension if free or busy. If busy, the call is queued to wait for the phone to become free. This is similar to transfer except it allows you to transfer calls to a busy phone.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Queue.
- Action Data: User number.
- Default Label: Queue.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.32 Call Record

This feature allows you to record a conversation and requires Voicemail Pro to be installed. An advice of recording warning will be given if configured on the voicemail system. The recording is placed in the mailbox specified by the user's Manual Recording Mailbox setting. Call recording also requires available conference resources similar to a three-party conference.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Record.
- Action Data: None.
- Default Label: Recor.
- Toggles: ✗.
- Status Indication: ✓.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.33 Call Steal

This function can be used with or without a specified user target.

- If the specified target has alerting calls, the function will connect to the longest waiting call.
- If the specified target has no alerting calls but does have a connected call, the function will take over the connected call, disconnecting the original user. This usage is subject to the Can Intrude setting of the Call Steal user and the Cannot Be Intruded setting of the target. The feature is independent the intrude settings of the third party to the call.
- If no target is specified, the function attempts to reclaim the users last ringing or transferred call if it has not been answered or has been answered by voicemail.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Steal.
- Action Data: User number or blank for last call transferred.
- Default Label: Aquir
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.34 Call Waiting Off

Switches call waiting off for the user. This button function is obsolete. The Call Waiting On 594 button function toggles on/off and indicates current status.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Waiting Off.
- Action Data: *None*.
- Default Label: CWOff.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.35 Call Waiting On

Enables call waiting on the user's extension. When the user is on a call and another call arrives, they will hear a call waiting tone. Note: Call waiting does not operate for user's with call appearance buttons. See Call Waiting 758.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Waiting On.
- Action Data: *None*.
- Default Label: CWOn.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.36 Call Waiting Suspend

Disables call waiting, if on, for the duration of the extension's next call.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Call Waiting Suspend.
- Action Data: *None*.
- Default Label: CWSus.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.37 Cancel All Forwarding

Cancels forward unconditional, forward on busy, forward on no answer, follow me and do not disturb if any of those are active on the user's extension.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Cancel All Forwarding.
- Action Data: *None*.
- Default Label: FwdOf.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.38 Cancel Leave Word Calling

Not supported. Provided for CTI emulation only. Cancels the last Leave Word Calling message originated by the user.

- Software Level: 1.0+.
- Action: Emulation -> Cancel Leave Word Calling.
- Action Data: *None*.
- Default Label: CnLWC.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.39 Cancel Ring Back When Free

Cancels any existing ringback set by the user, see Ring Back When Free 649. Note that the Ring Back When Free button toggles to set or cancel ringback when free and also indicates the current status.

- Software Level: 1.1+.
- Action: Advanced -> Miscellaneous -> Cancel Ring Back When Free.
- Action Data: *None*.
- Default Label: RBak-.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.40 Channel Monitor

For Avaya use only.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Channel Monitor.
- Action Data: *Channel*.
- Default Label: ChMon.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.41 Clear Call

This feature can be used to end the last call put on hold. This can be used in scenarios where a first call is already on hold and simply ending the second call will cause an unsupervised transfer of the first call.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Clear Call.
- Action Data: *None*.
- Default Label: Clear.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.42 Clear CW

End the user's current call and answer any call waiting. Requires the user to also have call waiting indication on. This function does not work for users with multiple call appearance buttons.

- Software Level: 1.1+.
- Action: Advanced -> Call ->Clear CW.
- Action Data: *None*.
- Default Label: ClrCW.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.43 Clear Hunt Group Night Service

Changes the specified hunt group from 'Night Service' mode to 'In Service' mode. This button function is obsolete. The Set Hunt Group Night Service 654 function can be used to toggle a group in/out of service and provides lamp status indication.

Note: If the hunt group has been placed into night service mode by an associated time profile, this function cannot override that night service mode.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Clear Hunt Group Night Service.
- Action Data: Group number.
    - IP Office 4.0+: If left blank, the button will affect all hunt groups of which the user is a member.
- Default Label: HGNS-.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
    - Analog: ✗
    - 1400 Series: ✓[1]
    - 1600 Series: ✓[1]
    - 20 Series: ✓
    - 2400 Series: ✓[1]
    - 3600 Series: ✗
    - 3700 Series: ✗
    - 3810: ✓
    - 4400 Series: ✓
    - 5400 Series: ✓
    - 4600 Series: ✓[1]
    - 5600 Series: ✓[1]
    - 6400 Series: ✓
    - 9040: ✓
    - 9600 Series: ✗
    - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.44 Clear Hunt Group Out Of Service

Changes the specified hunt groups status from 'Out of Service' mode to 'In Service' mode. This button function is obsolete. The Set Hunt Group Out Of Service 656 function can be used to toggle a group in/out of service and provides lamp status indication.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Clear Hunt Group Out of Service.
- Action Data: Group number.
    - IP Office 4.0+: If left blank, the button will affect all hunt groups of which the user is a member.
- Default Label: HGOS-.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
    - Analog: ✗
    - 1400 Series: ✓[1]
    - 1600 Series: ✓[1]
    - 20 Series: ✓
    - 2400 Series: ✓[1]
    - 3600 Series: ✗
    - 3700 Series: ✗
    - 3810: ✓
    - 4400 Series: ✓
    - 5400 Series: ✓
    - 4600 Series: ✓[1]
    - 5600 Series: ✓[1]
    - 6400 Series: ✓
    - 9040: ✓
    - 9600 Series: ✗
    - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.45 Clear Quota

Quotas can be assigned on outgoing calls to data services such as internet connections. The quota defines the number of minutes available for the service within a time frame set within the service, for example each day, each week or each month.

The Clear Quota function can be used to reset the quota for a specific service or for all services.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Clear Quota.
- Action Data: Service name" or "" (all services).
- Default Label: Quota.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.46 Conference Add

Places all the calls the user has on hold into a conference with the user.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Conference Add.
- Action Data: *None*.
- Default Label: Conf+.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.47 Conference Meet Me

This feature allows a user to join a specific numbered conference. By default, ad hoc conferences are assigned numbers starting from 100 for the first conference in progress. Therefore specifying a number away from this range ensure that the conference joined is not an ad hoc conference started by other users.

IP Office 4.1+: A currently connected caller can be transferred into the conference by pressing TRANSFER, then the Conference Meet Me button and TRANSFER again to complete the transfer. This allows the user to place callers into the conference specified by the button without being part of the conference call themselves. This option is only support on Avaya phones with a fixed TRANSFER button (excluding T3 and T3 IP phones).

Pre-IP Office 5.0: This function is not supported on IP500 systems without an *IP500 Upgrade Standard to Professional* license.

IP Office 6.0+: IP500 and IP500v2 systems require a Preferred Edition license.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Conference Meet Me.
- Action Data: Conference number. This can be an alphanumeric value up to 15 characters.
- Default Label: CnfMM.
- Toggles: ✗.
- Status Indication: ✓ IP Office 4.1+

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- Conference In Use.	CnfMM◀	CnfMM	Green on.
- Conference Idle.	CnfMM	CnfMM	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.48 Consult

Not supported. Provided for CTI emulation only.

- Software Level: 1.0+.
- Action: Emulation -> Consult.
- Action Data: *None*.
- Default Label: Cnslt.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.49 Coverage Appearance

Creates a button that alerts when a call to the specified covered user is unanswered after that users Individual Coverage Timer expires.

The call coverage appearance button user must also have at least one call appearance button programmed. The covered user does not need to be using call appearance buttons.

Coverage appearance functions, assigned to buttons that do not have status lamps or icons, are automatically disabled until the user logs in at a phone with suitable buttons.

IP Office: Appearance buttons can be set with a ring delay if required or to not ring. This does not affect the visual alerting displayed next to the button. The delay uses the user's Ring Delay 278 (User | Telephony | Multi-line Options 278) setting.

- Software Level: 3.0+.
- Action: Appearance -> Coverage Appearance.
- Action Data: User name.
- Default Label: *<user name>*.
- Toggles: ✗.
- Status Indication: ✓. See Coverage Button Indication 689.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✗
  - 2400 Series: ✓
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓
  - 5600 Series: ✓
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

7.3.50 Dial

This action is used to dial the number contained in the Telephone Number field. A partial number can be enter for the user to complete. On buttons with a text label area, Dial followed by the number is shown.

- Software Level: 1.0+.
- Action: Dial.
- Action Data: Telephone number or partial telephone number.
- Default Label: Dial.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays the telephone number set.
  - DSS Link LED: None.

7.3.51 Dial 3K1

The call is presented to local exchange as a "3K1 Speech Call". Useful in some where voice calls cost less than data calls.

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial 3K1.
- Action Data: Telephone number.
- Default Label: D3K1.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.52 Dial 56K

The call presented to local exchange as a "Data Call".

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial 56K.
- Action Data: Telephone number.
- Default Label: D56K.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.53 Dial 64K

The call is presented to local exchange as a "Data Call".

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial 64K.
- Action Data: Telephone number.
- Default Label: D64K.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.54 Dial CW

Call the specified extension number and force call waiting indication on if the extension is already on a call. The call waiting indication will not work if the extension called has multiple call appearance buttons in use.

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial CW.
- Action Data: User number.
- Default Label: DCW.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
Note that support for particular phone models is also dependant on the system software level.
    - Analog: ✗
    - 1400 Series: ✓[1]
    - 1600 Series: ✓[1]
    - 20 Series: ✓
    - 2400 Series: ✓[1]
    - 3600 Series: ✗
    - 3700 Series: ✗
    - 3810: ✓
    - 4400 Series: ✓
    - 5400 Series: ✓
    - 4600 Series: ✓[1]
    - 5600 Series: ✓[1]
    - 6400 Series: ✓
    - 9040: ✓
    - 9600 Series: ✓
    - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.55 Dial Direct

Call an extension and have the call automatically answered on speaker phone after 3 beeps. The extension called must support a handsfree speaker. If the extension does not have a handsfree microphone then the user must use the handset if they want to talk. If the extension is not free when called, the call is presented as a normal call on a call appearance button if available.

- IP Office 4.2 Q4 maintenance release and higher: This feature can be used as part of handsfree announced transfers 784.

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial Direct.
- Action Data: User number or name or blank for entry when pressed.
    - If left blank, the Dial Direct button can be used with User 666 buttons to specify the target.
- Default Label: Dirct.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
Note that support for particular phone models is also dependant on the system software level.
    - Analog: ✗
    - 1400 Series: ✓[1]
    - 1600 Series: ✓[1]
    - 20 Series: ✓
    - 2400 Series: ✓[1]
    - 3600 Series: ✗
    - 3700 Series: ✗
    - 3810: ✓
    - 4400 Series: ✓
    - 5400 Series: ✓
    - 4600 Series: ✓[1]
    - 5600 Series: ✓[1]
    - 6400 Series: ✓
    - 9040: ✓
    - 9600 Series: ✗
    - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.56 Dial Emergency

Dials the number specified regardless of any outgoing call barring applicable to the user.

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial Emergency.
- Action Data: Telephone number.
- Default Label: Emrgy.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.57 Dial Inclusion

Intrudes on the existing connected call of the specified target extension. The intruder and the target extension can then talk but cannot be heard by the other party. If the target extension is free the button makes a normal call.

During the intrusion all parties hear a repeated intrusion tone. When the intruder hangs-up the original call parties are reconnected.

Use of this feature is subject to the Can Intrude status of the intruder and the Cannot be Intruded status of the other call parties if internal.

- Software Level: 1.4+.
- Action: Advanced -> Dial -> Dial Inclusion.
- Action Data: User number or name or blank for user selection when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: Inclu.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.58 Dial Intercom

Call an extension and have the call automatically answered on speaker phone after 3 beeps. The extension called must support a handsfree speaker. If the extension does not have a handsfree microphone then the user must use the handset if they want to talk. If the extension is not free when called, the call is presented as a normal call on a call appearance button if available.

- IP Office 4.2 Q4 maintenance release and higher: This feature can be used as part of handsfree announced transfers 784.

- Software Level: 1.0+.
- Action: Emulation -> Dial Intercom.
- Action Data: User number or name or blank for number entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: Idial.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays 🔊 followed by the set number.
  - DSS Link LED: None.

7.3.59 Dial Paging

Makes a paging call to an extension or group specified. If no number is specified, this can be dialed after pressing the button. The target extension or group members must be free and must support handsfree auto-answer in order to hear the page.

On Avaya phones with a CONFERENCE button, a paged user can convert the page call into a normal call by pressing that button.

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial Paging.
- Action Data: User number or name or group number or name or blank for number entry when pressed.
- Default Label: Page.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

  - T3 Phones:
    - Classic/Comfort icon: Displays followed by target number if set.
    - DSS Link LED: None.

Paging Limits
The table below lists the maximum recommended size for paging groups.

Control Unit	Software Level	
	4.0+	3.2
IP500/IP500v2	64	–
IP412	48	24
IP406 V2	32	16
Small Office Edition	16	16
IP406 V1	–	16
IP403	–	16

7.3.60 Dial Physical Extn by Number

Call the specified extension using its Base Extension number setting. This is regardless of the current user logged in at that extension and any forwarding, follow me or do not disturb settings applied by the extension user. This function requires the extension to be assigned a default extension number in the system configuration. If the extension does not have a default extension number, Dial Physical Extn by ID 610 should be used.

- Software Level: 1.4+.
- Action: Advanced -> Dial -> Dial Physical Extn By Number.
- Action Data: Extension port base extension number.
- Default Label: PhyEx.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.61 Dial Physical Extn by ID

Call the specified extension, if free, regardless of the current user logged in at that extension and any forwarding, follow me or do not disturb settings applied by the extension user. This function uses the port ID shown in the system configuration.

- Software Level: 1.4+.
- Action: Advanced -> Dial -> Dial Physical Extn By ID.
- Action Data: Extension port ID number.
- Default Label: DialP.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.62 Dial Speech

This feature allows a short code to be created to force the outgoing call to use the Speech bearer capability.

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial Speech.
- Action Data: Telephone number.
- Default Label: DSpch.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.63 Dial V110

The call is presented to local exchange as a "Data Call".

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial V110.
- Action Data: Telephone number.
- Default Label: DV110.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.64 Dial V120

The call is presented to local exchange as a "Data Call".

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial V120.
- Action Data: Telephone number.
- Default Label: DV120.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.65 Display Msg

Allows the sending of text messages to digital phones on the local system.

- Software Level: 1.0+.
- Action: Advanced -> Dial -> Display Msg.
- Action Data: The telephone number takes the format *N";T"* where:
  - *N* is the target extension.
  - *T* is the text message. Note that the *";* before the text and the *"* after the text are required.
- Default Label: Displ.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.66 Dial Video

The call is presented to the local exchange as a "Video Call".

- Software Level: 1.1+.
- Action: Advanced -> Dial -> Dial Video.
- Action Data: Telephone number.
- Default Label: Dvide.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.67 Directed Call Pickup

Pickup a call ringing at a specific extension or hunt group.

- Software Level: 1.0+.
- Action: Emulation -> Directed Pickup.
- Action Data: User number or name or group number or name or blank for number entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: DpkUp.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

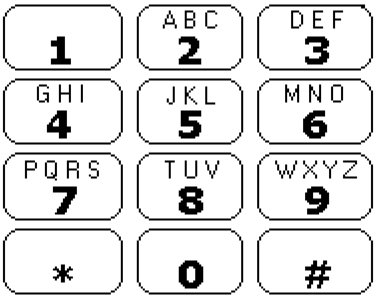
7.3.68 Directory

A Dir button provides access to various directories and allows telephone number selection by dialed name matching. The directories available for searching depend on the phone type, see User Directory Access 729. Once they user has selected a directory, dialing on the dial pad letter keys is used to display matching names, with controls for scrolling through the matching names and for calling the currently displayed name.

The method of name matching is controlled by the Dial by Name 160 (System | Telephony | Telephony) 160 setting in the system configuration:

- With Dial By Name on
  Matching is done against all the dial keys pressed. For example, dialing 527 matches names starting with JAS (for example "Jason") and KAR (for example "Karl"). Only the first 50 matches are displayed.
- With Dial By Name off
  Matching is done against the first letter only. For example pressing 5 displays names beginning with J. Press 5 again displays names beginning with K. Only the first 50 matches are displayed. This mode is not supported by IP Office Release 5+.

Name dialing functions on the system assume that the phone is using the standard ITU keypad as follows:



- Dialing Spaces
  For names that include spaces, the method of indicating a space has changed in IP Office 5.
  - Pre-IP Office 5
    To enter a name with a space, nothing is dialed for the space. For example "John S..." is dialed as 56467.
  - IP Office 5+
    To enter a name with a space, the 0 key is used for the space. For example "John S..." is dialed as 5646<u>0</u>7.

- Software Level: 1.0+.
- Action: Emulation -> Directory.
- Action Data: *None*.
- Default Label: Dir.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✗
  - 1600 Series: ✓*
  - 20 Series: ✓
  - 2400 Series: ✓*
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓*
  - 4600 Series: ✓*
  - 5600 Series: ✓*
  - 6400 Series: ✓
  - 9040: ✗
  - 9600 Series: ✗
  - T3/T3 IP Series: ✓
    - *✗ *Not 1603, 2402, 4601, 4602, 5402, 5601, 5602 and T7100 models.*
- T3 Phones:
  - Classic/Comfort icon: Displays .
  - DSS Link LED: None.

7.3.69 Do Not Disturb Exception Add

Adds a number to the user's "Do Not Disturb Exception List". This can be the number of an internal user or a number to match the CLI of a particular external caller.Calls from that number, except hunt group calls, will ignore the user's Do Not Disturb setting. For further details see Do Not Disturb (DND) 769 in the Telephone Features section.

- Software Level: 1.1+.
- Action: Advanced -> Do Not Disturb -> Do Not Disturb Exception Add.
- Action Data: Telephone number or CLI. Up to 31 characters. For CLI numbers any prefix added by the system must also be included.
- Default Label: DNDX+.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.70 Do Not Disturb Exception Delete

Removes a number from the user's "Do Not Disturb Exception List". This can be the number of an internal user or a number to match the CLI of a particular external caller.

- Software Level: 1.1+.
- Action: Advanced -> Do Not Disturb -> Do Not Disturb Exception Delete.
- Action Data: Telephone number or CLI.
- Default Label: DNDX-.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.71 Do Not Disturb Off

Cancels the user's 'do not disturb' mode if set. This button function is largely obsolete as the do not disturb on 617 function toggles on/off and indicates the button status.

- Software Level: 1.1+.
- Action: Advanced -> Do Not Disturb -> Do Not Disturb Off.
- Action Data: *None*.
- Default Label: DNDOf.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.72 Do Not Disturb On

Enables the user's 'do not disturb' 769 mode.

For CCR Agents, using this function button on the following phones will be requested the user to select a reason code - 1400, 1600, 2400, 4600, 5400, 5600 and 9600 Series phones with available programmable buttons.

- Software Level: 1.1+.
- Action: Advanced -> Do Not Disturb -> Do Not Disturb On.
- Action Data: *None*.
- Default Label: DNDOn.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.73 Drop

This action is supported on phones which do not have a permanent Drop button.

- For a currently connected call, pressing Drop disconnects the call. When drop is used to end a call, silence is returned to the user rather than dial tone. This is intended operation, reflecting that Drop is mainly intended for use by call center headset users.
- If the user has no currently connected call, pressing Drop will redirect a ringing call using the user's Forward on No Answer setting if set or otherwise to voicemail if available.
- For a conference call, on phones with a suitable display, Drop can be used to display the conference parties and allow selection of which party to drop from the conference.

- Software Level: 1.0+.
- Action: Emulation -> Drop.
- Action Data: *None*.
- Default Label: Drop.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✗
  - 1600 Series: ✗
  - 20 Series: ✓
  - 2400 Series: ✗
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓
  - 5600 Series: ✓
  - 6400 Series: ✗
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

7.3.74 Extn Login

This feature allows user configured with a log in code to take over ownership of an physical extension. That user's associated extension number becomes the number of the extension while they are logged in along with all their user settings (if appropriate to the phone type).

If the user logging in was already logged in or associated with another phone, they will be automatically logged out that phone.

When used, the user will be prompted to enter their extension number and then their log in code. Login codes of up to 15 digits are supported with Extn Login buttons. Login codes of up to 31 digits are supported with Extn Login short codes.

- Software Level: 1.1+.
- Action: Advanced -> Extension -> Extn Login.
- Action Data: *None*.
- Default Label: Login.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.75 Extn Logout

Logs out a user from the phone. The phone will return to its normal default user, if an extension number is set against the physical extension settings in the configuration. Otherwise it takes the setting of the NoUser user. This action is obsolete as Extn Login 619 can be used to log out an existing logged in user.

If the user who logged out was the default user for an extension, dialing *36 will associate the extension with the user unless they are set to forced log in.

IP Office 4.0+: This feature cannot be used by a user who does not have a log in code.

- Software Level: 1.1+.
- Action: Advanced -> Extension -> Extn Logout.
- Action Data: *None*.
- Default Label: Logof.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.76 Flash Hook

Sends a hook flash signal to the currently connected line if that line is an analog line.

- Software Level: 1.4+.
- Action: Advanced -> Miscellaneous -> Flash Hook.
- Action Data: *Optional*.
  Normally this field is left blank. IP Office 4.0+: It can contain the destination number for a Centrex Transfer for external calls on a line from a Centrex service provider.
- Default Label: Flash.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.77 Follow Me Here

Causes calls to the extension number specified, to be redirected to this user's extension.

IP Office 4.0+: User's with a log in code will be prompted to enter that code when using this function.

- Software Level: 1.1+.
- Action: Advanced -> Follow Me -> Follow Me Here.
- Action Data: User name or user number.
  - IP Office 4.0+: This field can be left blank for number entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: Here+.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

  - T3 Phones:
    - Classic/Comfort icon: Displays followed by the user name.
    - DSS Link LED: On when active.

7.3.78 Follow Me Here Cancel

Cancels any 'Follow Me Here' set on the specified extension. Only works if entered at the extension to which the extension's calls are being sent by the follow me action.

- Software Level: 1.1+.
- Action: Advanced -> Follow Me -> Follow Me Here Cancel.
- Action Data: User number or blank for number entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: Here-.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.79 Follow Me To

Leaving the extension blank prompts the user to enter the extension to which their calls should be redirected.

IP Office 4.0+: User's with a log in code will be prompted to enter that code when using this function.

- Software Level: 1.1+.
- Action: Advanced -> Follow Me -> Follow Me To.
- Action Data: User name or user number or blank for number entry when pressed 515.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: FolTo.
- Toggles: ✓.
- Status Indication: ✓.
  On/off status indication is provided if the button is programmed with a user name or number.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.80 Forward Hunt Group Calls Off

Cancels the forwarding of the user's hunt group calls. This function is largely obsolete since the button function Forward Hunt Group Calls On [623] toggles on/off and indicates status.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward Hunt Group Calls Off.
- Action Data: *None*.
- Default Label: FwdH-.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: [651] ✗.
- Phone Support [516]
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.81 Forward Hunt Group Calls On

Forward the user's hunt group calls (internal and external). This function only works when forward unconditional is also on and uses the same forwarding number as forward unconditional.

This option is only applied for calls to *Sequential* and *Rotary* type hunt groups. Calls from other hunt group types are not presented to the user when they have Forward Unconditional active. Note also that hunt group calls cannot be forwarded to another hunt group.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward Hunt Group Calls On.
- Action Data: *None*.
- Default Label: FwdH+.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: [651] ✗.
- Phone Support [516]
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.82 Forward Number

Sets the number to which calls are forwarded when the user has forwarding on. Used for all forwarding options unless a separate Forward On Busy Number is also set. Forwarding to an external number is blocked if Inhibit Off-Switch Transfers is selected within the system configuration.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward Number.
- Action Data: Telephone number.
  - IP Office 4.0+: The field to be left blank to prompt the user for entry when the button is pressed 515. If blank, users with a log in code will be prompted to enter that code.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: FwdNo.
- Toggles: ✗.
- Status Indication: ✓ *for system 4.0+.*
  For a button with a prefixed number, status indication will indicate when that number matches the users current set number. For a button with a no number, status indication will show when a number has been set.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	FwdNo◀	FwdNo	Green on.
- Off.	FwdNo	FwdNo	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.83 Forward On Busy Number

Sets the number to which calls are forwarded when using 'Forward on Busy' and/or 'Forward on No Answer'. Forwarding to an external number is blocked if Inhibit Off-Switch Transfers is selected within the system configuration.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward on Busy Number.
- Action Data: Telephone number.
  - IP Office 4.0+: The field to be left blank to prompt the user for entry when the button is pressed 515. If blank, users with a log in code will be prompted to enter that code.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: FwBNo.
- Toggles: ✗.
- Status Indication: ✓ *for system 4.0+.*
  For a button with a prefixed number, status indication will indicate when that number matches the users current set number. For a button with a no number, status indication will show when a number has been set.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	FwBNo◀	FwBNo	Green on.
- Off.	FwBNo	FwBNo	Off.

- User Admin: 651 ✗.

- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.

  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.84 Forward On Busy Off

Switches forward on busy off. This button function is largely obsolete, as Forward On Busy On 626 can be used to switch forward on busy on/off and provides status indication.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward on Busy Off.
- Action Data: *None*.
- Default Label: FwBOf.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.85 Forward On Busy On

Enables forwarding when the user's extension is busy. For users with call appearance buttons, they will only return busy when all call appearance buttons are in use. Uses the **Forward Number** as its destination unless a separate **Forward on Busy Number** is set.

IP Office 3.2+: **Forward Internal** (**User | Forwarding**) can also be used to control whether internal calls are forwarded.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward on Busy On.
- Action Data: *None*.
- Default Label: FwBOn.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.86 Forward On No Answer Off

Switches forward on no answer off. This button function is largely obsolete, as Forward On No Answer On 627 can be used to switch forward on no answer on/off and provides status indication.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward on No Answer Off.
- Action Data: *None*.
- Default Label: FwNOf.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.87 Forward On No Answer On

Switches forward on no answer on/off. The time used to determine the call as unanswered is the user's no answer time. Uses the Forward Number as its destination unless a separate Forward on Busy Number is set.

IP Office 3.2+: Forward Internal (User | Forwarding) can also be used to control whether internal calls are forwarded.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward on No Answer On.
- Action Data: *None*.
- Default Label: FwNOn.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.88 Forward Unconditional Off

Switch 'forward all calls' off. This does not affect 'Forward on No Answer' and/or 'Forward on Busy' if also on. This function is largely obsolete as a button set to Forward Unconditional On 628 toggles on/off and indicates when on.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward Unconditional Off.
- Action Data: *None*.
- Default Label: FwUOf.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.89 Forward Unconditional On

This function is also know as 'divert all' and 'forward all'. It forwards all calls, except hunt group and page calls, to the forward number set for the user's extension. To also forward hunt group calls to the same number 'Forward Hunt Group Calls On 623' must also be used.

IP Office 3.2+: Forward Internal (User | Forwarding) can also be used to control whether internal calls are forwarded.

In addition to the lamp indication shown below, most phones display D when forward unconditional is on.

- Software Level: 1.1+.
- Action: Advanced -> Forward -> Forward Unconditional On.
- Action Data: *None*.
- Default Label: FwUOn.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

  - T3 Phones:
    - Classic/Comfort icon: Displays followed by the user name.
    - DSS Link LED: On when active.

7.3.90 Group

Monitors the status of a hunt group queue. This option is only supported for hunt groups with queuing enabled. The user does not have to be a member of the group.

Depending on the users button type, indication is given for when the group has alerting calls and queued calls (queued in this case is defined as more calls waiting than there are available group members).

- Pre-4.0 IP Office: Pressing a Group button displayed information about the longest waiting call in the queue and options to answer, drop or ignore the call.
- IP Office 4.0+: Pressing a Group button answers the longest waiting call.
- IP Office 4.0+: The definition of queued calls include group calls that are ringing. However, for operation of the Group button, ringing calls are separate from other queued calls.

- Software Level: 1.0+.
- Action: Group.
- Action Data: Group name enclosed in " " double-quotes or group number.
- Default Label: <*group name*>.
- Toggles: ✗.
- Status Indication: ✓ *Required*.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- No calls	Main	Main	Not lit.
- Call alerting.	Main◆	Main◆	Green flash.
- Calls queued.	Main	Main	Red flash.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.91 Group Listen On

Using group listen allows callers to be heard through the phone's handsfree speaker but to only hear the phone's handset microphone. When group listen is enabled, it modifies the handsfree functionality of the user's phone in the following manner

- When the user's phone is placed in handsfree / speaker mode, the speech path from the connected party is broadcast on the phone speaker but the phone's base microphone is disabled.
- The connected party can only hear speech delivered via the phone's handset microphone.
- Group listen is not supported for IP phones or when using a phone's HEADSET button.
- Group listen is automatically turned off when the call is ended.

This enables listeners at the user's phone to hear the connected party whilst limiting the connected party to hear only what is communicated via the phone handset.

- Software Level: 4.1+.
- Action: Advanced -> Extension -> Group Listen On.
- Action Data: *None*.
- Default Label: Group Listen On.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✗
  - 1600 Series: ✗
  - 20 Series: ✗
  - 2400 Series: ✓*
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✗
  - 4600 Series: ✗
  - 5400 Series: ✓*
  - 5600 Series: ✗
  - 6400 Series: ✓*
  - 9040: ✗
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗

7.3.92 Group Paging

Makes a paging call to an extension or group specified. If no number is specified, this can be dialed after pressing the button. The target extension or group members must be free and must support handsfree auto-answer in order to hear the page.

On Avaya phones, a paged user can convert the page call into a normal call by pressing the Conference button.

- Software Level: 1.0+.
- Action: Emulation -> Group Paging.
- Action Data: User number or name or group number or name.
  - IP Office 4.0+: On large display phones, if configured without a preset target, this type of button will display an interactive button menu 515 for target selection.
- Default Label: Page.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays followed by target number if set.
  - DSS Link LED: None.

Paging Limits
The table below lists the maximum recommended size for paging groups.

Control Unit	Software Level	
	4.0+	3.2
IP500/IP500v2	64	–
IP412	48	24
IP406 V2	32	16
Small Office Edition	16	16
IP406 V1	–	16
IP403	–	16

7.3.93 Headset Toggle

This function is intended for use with Avaya phones that have separate handset and headset sockets but do not provide a dedicated Headset button, for example older style 4400 Series and 4600 Series phones. On phones without a headset socket or with a dedicated headset button this control will have no effect.

- Software Level: 1.4+.
- Action: Miscellaneous -> Headset Toggle.
- Action Data: *None*.
- Default Label: HdSet.
- Toggles: ✓.
- Status Indication: ✓.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✗
  - 1600 Series: ✗
  - 20 Series: ✗
  - 2400 Series: ✗
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✓
  - 4600 Series: ✓*
  - 5400 Series: ✗
  - 5600 Series: ✗
  - 6400 Series: ✗
  - 9040: ✗
  - 9600 Series: ✗
  - T3/T3 IP Series: ✓
    - **4606, 4612 and 4624 only.*
- T3 Phones:
  - Classic/Comfort icon: Displays HdSet.
  - DSS Link LED: On when active.

7.3.94 Hold Call

This uses the Q.931 Hold facility, and "holds" the incoming call at the ISDN exchange, freeing up the ISDN B channel. The Hold Call feature "holds" the current call to a slot. The current call is always automatically placed into slot 0 if it has not been placed in a specified slot. Only available if supported by the ISDN exchange.

- Software Level: 1.1+.
- Action: Advanced -> Hold -> Hold Call.
- Action Data: ISDN Exchange hold slot number or blank (slot 0).
- Default Label: Hold.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.95 Hold CW

Place the user's current call on hold and answers the waiting call. This function is not supported on phones which have multiple call appearance buttons set.

- Software Level: 1.1+.
- Action: Advanced -> Hold -> Hold CW.
- Action Data: *None*.
- Default Label: HoldCW.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
    - Analog: ✗
    - 1400 Series: ✓[1]
    - 1600 Series: ✓[1]
    - 20 Series: ✓
    - 2400 Series: ✓[1]
    - 3600 Series: ✗
    - 3700 Series: ✗
    - 3810: ✓
    - 4400 Series: ✓
    - 5400 Series: ✓
    - 4600 Series: ✓[1]
    - 5600 Series: ✓[1]
    - 6400 Series: ✓
    - 9040: ✓
    - 9600 Series: ✗
    - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.96 Hold Music

This feature allows the user to listen to the system's music on hold. See Music On Hold 763 for more information.

- Software Level: 1.1+.
- Action: Advanced -> Hold -> Hold Music.
- Action Data: *Optional*.
  Systems can support multiple hold music sources. However only the system source is supported for Hold Music buttons.
- Default Label: Music.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
    - Analog: ✗
    - 1400 Series: ✓[1]
    - 1600 Series: ✓[1]
    - 20 Series: ✓
    - 2400 Series: ✓[1]
    - 3600 Series: ✗
    - 3700 Series: ✗
    - 3810: ✓
    - 4400 Series: ✓
    - 5400 Series: ✓
    - 4600 Series: ✓[1]
    - 5600 Series: ✓[1]
    - 6400 Series: ✓
    - 9040: ✓
    - 9600 Series: ✓
    - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.97 Hunt Group Enable

An individual users membership of any particular hunt groups is programmed through the system configuration. This control allows the user to enable or disable that membership. While enabled, the user can receive hunt group calls when logged in.

In addition to the lamp indication below, phones display G when any group membership is enabled.

- Software Level: 1.1+.
- Action: Advanced -> Hunt Group -> Hunt Group Enable.
- Action Data: Group number or name or blank for all groups of which the user is a member.
- Default Label: HGEna.
- Toggles: ✓.
- Status Indication: ✓ *Required*.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

  - T3 Phones:
    - Classic/Comfort icon: Displays [♣] followed by the group number or * for all if programmed with no specific group number.
    - DSS Link LED: On when active.

7.3.98 Hunt Group Disable

This function is obsolete, the Hunt Group Enable 634 function being able to toggle membership between enabled and disabled and providing lamp indication of when membership is enabled.

An individual user's membership of any particular hunt groups is programmed through the system configuration. This control allows the user to disable that membership. They will no longer receive calls to that hunt group until their membership is enabled again.

- Software Level: 1.1+.
- Action: Advanced -> Hunt Group -> Hunt Group Disable.
- Action Data: Group number or blank for all groups of which the user is a member.
- Default Label: HGDis.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.99 Inspect

Not supported. Provided for CTI emulation only. Allows users on display phones to determine the identification of held calls. Allows users on an active call to display the identification of incoming calls.

- Software Level: 1.0+.
- Action: Emulation -> Inspect.
- Action Data: None.
- Default Label: Inspt.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.100 Internal Auto-Answer

This function is also know as handsfree auto-answer. It sets the user's extension to automatically connect internal calls after a single tone. This function should only be used on phones that support handsfree operation.

- Software Level: 1.0+.
- Action: Emulation -> Internal Auto-Answer.
- Action Data: *Optional*.
  - If left blank this function acts as described above for internal auto-answer.
  - IP Office 4.1+: *FF* can be entered. In that case the button will enable/disable headset force feed operation for external calls. In this mode, when headset mode is selected but the phone is idle, an incoming external call will cause a single tone and then be automatically connected. This operation is only supported on Avaya phones with a fixed HEADSET button. Ring delay is applied if set on the call or line appearance button receiving the call before the call is auto-connected.
- Default Label: HfAns.
- Toggles: ✓.
- Status Indication: ✓ *Required*.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays HfAns.
  - DSS Link LED: On when active.

7.3.101 Leave Word Calling

Not supported. Provided for CTI emulation only. Leaves a message for the user associated with the last number dialed to call the originator.

- Software Level: 1.0+.
- Action: Emulation -> Leave Word Calling.
- Action Data: *None*.
- Default Label: LWC.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.102 Line Appearance

Creates an line appearance button linked to the activity of a specified line appearance ID number. The button can then be used to answer and make calls on that line.

The line appearance button user must also have at least one call appearance button programmed before line appearance buttons can be programmed.

Line appearance functions, assigned to buttons that do not have status lamps or icons, are automatically disabled until the user logs in at a phone with suitable buttons.

- IP Office 3.2+: Appearance buttons can be set with a ring delay if required or to not ring. This does not affect the visual alerting displayed next to the button. The delay uses the user's Ring Delay 278 (User | Telephony | Multi-line Options 278) setting.
- IP Office 4.2+: Lline appearances are supported on T3 and T3 IP phones. These phones do not require (or support) call appearance buttons in order to use line appearances.

- Software Level: 3.0+.
- Action: Appearance -> Line Appearance.
- Action Data: Line ID number.
- Default Label: Line *<Line ID number>*.
- Toggles: ✗.
- Status Indication: ✓. See Line Appearance Button Indication 694.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✗
  - 2400 Series: ✓
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓
  - 5600 Series: ✓
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

7.3.103 Manual Exclude

Not supported. Provided for CTI emulation only.

- Software Level: 1.0+.
- Action: Emulation -> Manual Exclude.
- Action Data: *None*.
- Default Label: Excl.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.104 MCID Activate

This action is used with ISDN Malicious Caller ID call tracing. It is used to trigger a call trace at the ISDN exchange. The call trace information is then provided to the appropriate legal authorities.

This option requires the line to the ISDN to have MCID enabled at both the ISDN exchange and on the system. The user must also be configured with Can Trace Calls 276 (User | Telephony | Supervisor Settings 276) enabled.

- Software Level: 4.0+.
- Action: Advanced -> Miscellaneous -> MCID Activate.
- Action Data: *None*.
- Default Label: MCID.
- Toggles: ✗.
- Status Indication: ✓.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✗
  - 2400 Series: ✓
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 4600 Series: ✓*
  - 5400 Series: ✓
  - 5600 Series: ✓*
  - 6400 Series: ✗
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

    - **Not 1603, 4601, 4602, 5601 and 5602.*

7.3.105 Off Hook Station

Enables the user's extension to be controlled by an application, for example Phone Manager or SoftConsole. Calls can then be answered and cleared through the application without having to manually go off or on hook. Requires the phone to support full handsfree operation.

- Software Level: 1.1+.
- Action: Advanced -> Miscellaneous -> Off Hook Station.
- Action Data: None.
- Default Label: OHStn.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✓
  - 2400 Series: ✓*
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 4600 Series: ✓*
  - 5400 Series: ✓*
  - 5600 Series: ✓*
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

    - *✗ *Not 2402, 4601, 4602, 5402, 5601 and 5602 models.*

7.3.106 Park

Monitors the status of a system park slot. The user can use the button to park a call into that slot and to also retrieve a call parked in that slot including calls parked by other users.

Park buttons with indication will indicate when the park slot is in use. Similarly the Park buttons within IP Office application (for example Phone Manager, SoftConsole and one-X Portal for IP Office) can be used to park, retrieve and indicate parked calls.

- Software Level: 1.0 to 3.2 only.
  For IP Office 4.0 and higher this function has been replaced by Emulation | Call Park 588.
- Action: Park Call.
- Action Data: Park slot number.
- Default Label: Park.
- Toggles: ✓.
- Status Indication: ✓ *Required*.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- No parked call	PARK1	PARK1	Off
- Parked here.	PARK1◆	PARK1◆	Green flash.
- Parked elsewhere.	PARK1	PARK1	Red flash.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.

  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.107 Park Call

IP Office 4.0+: This function has been replaced by Emulation | Call Park 588.

- Software Level: 1.0 to 3.2 only.
- Action: Advanced -> Call -> Park Call.
- Action Data: Park slot number.
- Default Label: Park.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- No parked call	PARK1	PARK1	Off
- Parked here.	PARK1◆	PARK1◆	Green flash.
- Parked elsewhere.	PARK1	PARK1	Red flash.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.108 Priority Call

This feature allows the user to call another user even if they are set to 'do not disturb'. A priority call will follow forward and follow me settings but will not go to voicemail.

Software Level: 1.1+.

- Action: Advanced -> Call -> Priority Call.
- Action Data: User number or name.
- Default Label: PCall.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.109 Priority Calling

Not supported. Provided for CTI emulation only.

- Software Level: 1.0+.
- Action: Emulation -> Priority Calling.
- Action Data: *None*.
- Default Label: Pcall.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.110 Private Call

When on, any subsequent calls cannot be intruded on, bridged into or silently monitored until the user's private call status is switched off.

Note that use of private calls is separate from the user's intrusion settings. If a user is set to Cannot be Intruded, switching private calls off does not affect that status. To allow private calls to be used to fully control the user status, Cannot be Intruded should be disabled for that user.

If enabled during a call, any current recording, intrusion or monitoring is ended.

- Software Level: 4.0+.
- Action: Advanced -> Call -> Private Call.
- Action Data: *None*.
- Default Label: PrivC.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✗
  - 2400 Series: ✓
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 4600 Series: ✓*
  - 5400 Series: ✓
  - 5600 Series: ✓*
  - 6400 Series: ✗
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗
    - **Not 1603, 4601, 4602, 5601 and 5602.*

7.3.111 Relay Off

Opens the specified switch in the system's external output port (EXT O/P).

- Software Level: 1.1+.
- Action: Advanced -> Relay -> Relay Off.
- Action Data: Switch number (1 or 2).
- Default Label: Rely-.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.112 Relay On

Closes the specified switch in the system's external output port (EXT O/P).

- Software Level: 1.1+.
- Action: Advanced -> Relay -> Relay On.
- Action Data: Switch number (1 or 2).
- Default Label: Rely+.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.113 Relay Pulse

Closes the specified switch in the system's external output port (EXT O/P) for 5 seconds and then opens the switch.

- Software Level: 1.1+.
- Action: Advanced -> Relay -> Relay Pulse.
- Action Data: Switch number (1 or 2).
- Default Label: Relay.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

  - T3 Phones:
    - Classic/Comfort icon: Displays S1 or S2 dependant on switch number.
    - DSS Link LED: None.

7.3.114 Resume Call

Resume a call previously suspended to the specified ISDN exchange slot. The suspended call may be resumed from another phone/ISDN Control Unit on the same line.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Resume Call.
- Action Data: ISDN Exchange suspend slot number.
- Default Label: Resum.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.115 Retrieve Call

Retrieves a call previously held to a specific ISDN exchange slot. Only available when supported by the ISDN exchange.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Retrieve Call.
- Action Data: Exchange hold slot number.
- Default Label: Retriv.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.116 Ring Back When Free

Sets a ringback on the extension being called. When the target extension ends its current call, the ringback users is rung (for their set No Answer Time/Allocated Answer Interval) and if they answer, a new call is made to the target extension.

Ringback can be cleared using the Cancel Ring Back When Free 596 function.

- Software Level: 1.1+.
- Action: Advanced -> Miscellaneous -> Ring Back When Free.
- Action Data: None.
- Default Label: AutCB.
- Toggles: ✗.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.117 Ringer Off

Switches the phone's call alerting ring on/off.

- Software Level: 1.0+.
- Action: Emulation -> Ringer Off.
- Action Data: *None*.
- Default Label: RngOf.
- Toggles: ✓.
- Status Indication: ✓ *Required*.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.118 Self-Administer

Allows a user to program features against other programmable buttons themselves.

- IP Office 3.0+: Appearance 579 can no longer be used to create call appearance buttons. Similarly, existing call appearance button cannot be overwritten using any of the other Admin button functions.
- IP Office 4.0+: User's with a log in code will be prompted to enter that code when they use this button action.
- IP Office 4.2+: T3 phone users can access a similar set of functions for button programming, see T3 Phone Self-Administration 512.
- On 4412D+, 4424D+, 4612IP, 4624IP, 6408D, 6416D, 6424D phones:
  - Admin can be permanently accessed via Menu, ▶, ▶, Admin. See Using a Menu Key 509.
  - Admin1 can be permanently accessed via Menu, Menu, ▶, ProgA, ▶, DSS.

- Software Level: 1.0+.
- Action: Emulation -> Self-Administer.
- Action Data: *See below.*

Value	Other Phones
None	If no value is set, the button allows user programming of the following emulation actions. Abbreviated Dial. 573 Abbreviated Dial Program. 574 Account Code Entry. 575 AD Suppress. 578 Automatic Callback. 580 Break Out (4.0+). 581 Call Forwarding All. 585 Call Park (4.0+). 588 Call Park to Other Extension (4.0+). 589 Call Pickup. 590 Directed Call Pickup. 614 Directory. 615 Drop. 618 Group Paging. 631 Headset Toggle. 632 Hook Flash (4.0+). 620 Internal Auto-Answer. 637 Park Call (pre-4.0) 642 Ringer Off. 650 Self-Administer. 651 Send All Calls. 652 Set Hunt Group Night Service. 654 Time of Day. 662 Timer. 662
1	If *1* is entered as the telephone number, allows user programming of the following system functions. Dial (pre-4.0). 604 Abbreviated Dial (4.0+). 573 Group. 629 Park (pre-4.0). 642 CPark (4.0+). 588 User. 666 Flash Hook. 620
2	If *2* is entered, the button can be used for viewing details of the control unit type and its software version. This option is available in IP Office 4.1+. If the user has a log in code set, they will be prompted to enter that code. System phone 765 users can also use the button to manually set the system's date and time 727.
3	Not used.

Default Label: Admin.

- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✓
  - 2400 Series: ✓
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 4600 Series: ✓*
  - 5400 Series: ✓*
  - 5600 Series: ✓*
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✓*2

    - *Not 1403, 1603, 2402, 5402, 4601, 4602, 5601 and 5602.
    - *2: IP Office 4.2+: See T3 Phone Self-Administration 512.

7.3.119 Send All Calls

Sets the user's extension into 'Do Not Disturb' mode. Callers, other than those on the user's do not disturb exception list, receive busy or are diverted to the users voicemail mailbox. Note that with a call already connected and other calls already alerting, enabling Do Not Disturb will not affect those calls already existing. For full details of see Do Not Disturb 769.

When on, most phones display an N on the display. This function and the Do Not Disturb On 617 function work in parallel, ie. setting one sets the other.

- Software Level: 1.0+.
- Action: Emulation -> Send All Call.
- Action Data: *None*.
- Default Label: SAC.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays [♣].
  - DSS Link LED: On when active.

7.3.120 Set Absent Text

This feature can be used select the user's current absence text. Note: The user still has to select Set or Clear on their phone to display or hide the text. This text is then displayed to internal callers who have suitable display phones or applications.

The text is displayed to callers even if the user has forwarded their calls or is using follow me. Absence text is supported across a Small Community Network (SCN).

The absence text message is limited to 128 characters. Note however that the amount displayed will depend on the caller's telephone device or application.

- Software Level: 1.1+.
- Action: Advanced -> Set -> Set Absent Text.
- Action Data: The telephone number should take the format "*y,n,text*" where:
  - *y* = 0 or 1 to turn this feature off or on repesctively.
  - *n* = the number of the absent statement to use:
    - 0 = None.
    - 1 = On vacation until.
    - 2 = Will be back.
    - 3 = At lunch until.
    - 4 = Meeting until.
    - 5 = Please call.
    - 6 = Don't disturb until.
    - 7 = With visitors until.
    - 8 = With cust. til.
    - 9 = Back soon.
    - 10 = Back tomorrow.
    - 11 = Custom.
  - *text* = any text to follow the absent statement..
  - Default Label: Absnt.
  - Toggles: ✗.
  - Status Indication: ✗.
  - User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓[1]
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

    1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602.

7.3.121 Set Account Code

Dials an account code and then returns dial tone for the user to dial a number. Can also be used to enter an account code after a call has been connected.

- Software Level: 2.1+.
- Action: Advanced -> Set -> Set Account Code..
- Action Data: Account code or blank.
  If blank, the user is prompted to dial an account code after pressing the button. This option is not supported on XX02 phone modules.
- Default Label: Acct.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

  - T3 Phones:
    - Classic/Comfort icon: Displays 1234.
    - DSS Link LED: None.

7.3.122 Set Hunt Group Night Service

Puts the specified hunt group into 'Night Service' mode. Calls to a group set to night service, receive busy or are diverted to voicemail if available or are diverted to the group's night service fallback group if set.

This function cannot be used to override hunt groups already set to night service mode by an associated time profile.

- Software Level: 1.1+.
- Action: Advanced -> Set -> Set Hunt Group Night Service.
- Action Data: Hunt group extension number.
  - IP Office 4.0+: If left blank, the button will affect all hunt groups of which the user is a member.
- Default Label: HGNS+.
- Toggles: ✓.
- Status Indication: ✓ *Required*.
  If the button is blank (no specific hunt group) it will indicate on if any one of the hunt groups of which the user is a member is set to night service. If the button is set for multiple hunt groups it will indicate on if any one of those groups is set to night service.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◀	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.

2. May have limited support on some specific T3 phone models if detailed below.

- T3 Phones: Supported on Avaya T3 Classic, T3 Comfort phones and DSS Link units only.
  - T3 Classic/T3 Comfort icon: Displays ☽ followed by the group number. The background uses the same settings as the LED below.
  - DSS Link LED: On when all related groups are in night service. Slow flash if related hunt groups are in mixed states.

7.3.123 Set Hunt Group Out Of Service

Puts the specified hunt group into 'Out of Service' mode. Calls to a group set to out of service receive busy or are diverted to voicemail if available or are diverted to the group's out of service fallback group if set.

Pre-IP Office 4.0: This function cannot be used to override hunt groups already set to night service mode by an associated time profile.

IP Office 4.0+: This function can be used to used to override hunt groups already set to night service mode by an associated time profile.

- Software Level: 1.1+.
- Action: Advanced -> Set -> Set Hunt Group Out of Service.
- Action Data: Hunt group extension number.
  - IP Office 4.0+: If left blank, the button will affect all hunt groups of which the user is a member.
- Default Label: HGOS+.
- Toggles: ✓.
- Status Indication: ✓ *Required*.
  If the button is blank (no specific hunt group) it will indicate on if any one of the hunt groups of which the user is a member is set out of service. If the button is set for multiple hunt groups it will indicate on if any one of those groups is set out of service.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays – followed by the group number. The background uses the same settings as the LED below.
  - DSS Link LED: On when set. On when all related groups are out of service. Slow flash if related hunt groups are in mixed states.

7.3.124 Set Inside Call Seq

Only supported for analog extensions. The distinctive ringing pattern 762 used for other phones is set by the phone type. See the Set Inside Call Seq 494 short code.

7.3.125 Set Night Service Group

This button allows the user to change the Night Service target of a hunt group. The button user does not have to be a member of the hunt group. For Small Community Networks this function can be used for hunt groups on remote systems.

Changing the destination does not affect calls already ringing at the hunt groups previous night service destination.

- Software Level: 4.2+.
- Action: Advanced -> Set -> Set Night Service Group.
- Action Data: Hunt group extension number. This is the group for which the night service destination is being set.
- Default Label: SetNSG.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

    - *On T3 phones this option is accessible through the phone's menus.*

7.3.126 Set Out of Service Group

This button allows the user to change the Out of Service target of a hunt group. The button user does not have to be a member of the hunt group. For Small Community Networks this function can be used for hunt groups on remote systems.

Changing the destination does not affect calls already ringing at the hunt groups previous Out of Service destination.

- Software Level: 4.2+.
- Action: Advanced -> Set -> Set Out of Service Group.
- Action Data: Hunt group extension number. This is the group for which the night service destination is being set.
- Default Label: SetOOSG.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

    - *On T3 phones this option is accessible through the phone's menus.*

7.3.127 Set No Answer Time

Allows the user to change their no answer time setting. This is the time calls will ring before going to voicemail or following the user's divert on no answer setting if set on.

In situations where call coverage is also being used, the user's no answer time must be greater than their individual coverage time for coverage to occur.

- Software Level: 1.1+.
- Action: Advanced -> Set -> Set No Answer Time.
- Action Data: Time in seconds.
- Default Label: NATim.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.128 Set Outside Call Seq

Only supported for analog extensions. The distinctive ringing pattern 762 used for other phones is set by the phone type. See the Set Outside Call Seq 496 short code.

7.3.129 Set Ringback Seq

Only supported for analog extensions. The distinctive ringing pattern 762 used for other phones is set by the phone type. See the Set Ringback Seq 497 short code.

7.3.130 Set Wrap Up Time

Allows the user to change their Wrap-up Time setting. Specifies the amount of time after ending one call before another call can ring. During this interval the user is treated as still being on a call. It is recommended that this option is not set to less than the default of 2 seconds. 0 is used to allow immediate ringing.

- Software Level: 1.1+.
- Action: Advanced -> Set -> Set Wrap Up Time.
- Action Data: Time in seconds. Range 0 to 99999 seconds.
- Default Label: WUTim
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.131 Stored Number View

Not supported. Provided for CTI emulation only. Allows a user to view the contents of any programmed feature button.

- Software Level: 1.0+.
- Action: Emulation -> Stored Number View.
- Action Data: *None*.
- Default Label: BtnVu.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.132 Suspend Call

Uses the Q.931 Suspend facility. Suspends the incoming call at the ISDN exchange, freeing up the ISDN B channel. The call is placed in exchange slot 0 if a slot number is not specified. Only available when supported by the ISDN exchange.

- Software Level: 1.1+.
- Action: Advanced -> Suspend -> Suspend.
- Action Data: Exchange slot number or blank (slot 0).
- Default Label: Suspe.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.133 Suspend CW

Uses the Q.931 Suspend facility. Suspends the incoming call at the ISDN exchange and answer the call waiting. The call is placed in exchange slot 0 if a slot number is not specified. Only available when supported by the ISDN exchange.

- Software Level: 1.1+.
- Action: Advanced -> Suspend -> Suspend CW.
- Action Data: Exchange slot number or blank (slot 0).
- Default Label: SusCW.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.134 Time of Day

Displays the time and date on the user's telephone. This function is ignored on those Avaya phones that display the date/time by default.

- Software Level: 1.0+.
- Action: Emulation -> Time of Day.
- Action Data: *None*.
- Default Label: TmDay.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✓
  - 5400 Series: ✓[1]
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗

  1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602 models.

7.3.135 Timer

Starts a timer running on the display of the user's extension. The timer disappears when the user ends a call.

- For pre-6.1: This function is not supported on Avaya phones that display a call timer next to each call appearance through the phone's own settings.
- For 6.1: This function can be used on Avaya phones (except 9600 Series) that display a call timer next to each call appearance. The button will temporarily turn the call timer on or off for the currently selected call appearance. The change only applies for the duration of the current call.

- Software Level: 1.0+.
- Action: Emulation -> Timer.
- Action Data: *None*.
- Default Label: Timer.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✓.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✓
  - 5400 Series: ✓[1]
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗

  1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602 models.

7.3.136 Toggle Calls

Cycle between the user's current call and any held calls.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Toggle Calls..
- Action Data: *None*.
- Default Label: Toggl.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.137 Twinning

This action can be used by user's setup for mobile twinning. This action is not used for internal twinning.

- While the phone is idle, the button allows the user to set and change the destination for their twinned calls. It can also be used to switch mobile twinning on/off and indicates the status of that setting.
- When a call has been routed by the system to the user's twinned destination, the Twinning button can be used to retrieve the call at the user's primary extension.
- In configurations where the call arrives over an IP trunk and the outbound call is on an IP trunk, SCN may optimise the routing and in this case the button may not be useable to retrieve the call.
  - For user's setup for one-X Mobile Client, changes to their Mobile Twinning status made through the system configuration or using a Twinning button are not reflected in the status of the Extension to Cellular icon on their mobile client. However, changes to the Extension to Cellular status made from the mobile client are reflected by the Mobile Twinning field in the system configuration. Therefore, for one-X Mobile Client users, it is recommended that they control their Mobile Twinning status through the one-X Mobile Client rather than through a Twinning button.
- Mobile Twinning Handover (IP Office Release 6.1)
  When on a call on the primary extension, pressing the Twinning button will make an unassisted transfer to the twinning destination. This feature can be used even if the user's Mobile Twinning setting was not enabled.
  - During the transfer process the button will wink.
  - Pressing the twinning button again will halt the transfer attempt and reconnect the call at the primary extension.
  - The transfer may return if it cannot connect to the twinning destination or is unanswered within the user's configured Transfer Return Time (if the user has no Transfer Return Time configured, a enforced time of 15 seconds is used).

- Software Level: 3.2+.
- Action: Emulation -> Twinning.
- Action Data: *None*.
- Default Label: Twinning.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	Twinning ◄	Twinning	Green on.

- Off.	Twinning	Twinning	Off.
- Twinned call at secondary	Twinning	Twinning	Red on.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✓
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✓
  - 5400 Series: ✓[1]
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗

  1. Not 1403, 1603, 2402, 4601, 4602, 5402, 5601 and 5602 models.

7.3.138 Unpark Call

This function is obsolete, since the Call Park 588 function can be used to both park and retrieve calls and provides visual indication of when calls are parked. Retrieve a parked call from a specified system park slot.

- Software Level: 1.1+.
- Action: Advanced -> Call -> Unpark Call.
- Action Data: System park slot number. This must match a park slot ID used to park the call.
- Default Label: Ride.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.139 User

Monitors whether another user's phone is idle or in use. The **Telephone Number** field should contain the users name enclosed in double quotes. The button can be used to make calls to the user or pickup their longest waiting call when ringing. On buttons with a text label, the user name is shown.

The actions performed when the button is pressed will depend on the state of the target user and the type of phone being used. It also depend on whether the user is local or on a remote SCN system.

<table>
<tr><th>Phone</th><th>Large display 1400, 1600, 2400, 4600, 5400, 5600, 9600 Phones</th><th>Other Phones or across a Small Community Network</th></tr>
<tr><td>Idle</td><td colspan="2">Call the user.</td></tr>
<tr><td>Ringing</td><td>Displays an option to pickup the call.</td><td>Picks up the call.</td></tr>
<tr><td rowspan="2">On a Call</td><td>For IP Office 4.0+ the following options are displayed (name lengths may vary depending on the phone display):
• CALL
Initiates a call to the users.
• MESSAGE
Cause a single burst of ringing on the target phone. On some phones, when they end their current call their phone will then display PLEASE CALL and your extension number.
• VOICEMAIL
Call the user's voicemail mailbox.
• CALLBACK
Set an automatic callback.</td><td rowspan="2">No action.
For 1400, 1600 and 9600 Series phones, the Call, Voicemail and Callback options are supported.</td></tr>
<tr><td>For 1400, 1600 and 9600 Series phones the following additional options are displayed:
• Drop
Disconnect the user's current call.
If configured to be able to intrude on the user:
• Acquire
Take control of the call.
• Intrude
Intrude into the call, turning it into a 3-way conference.
If configured to be able to listen 443 to the user:
• Listen
Start silent monitoring of the user's call.</td></tr>
</table>

- A User button can be used in conjunction with other buttons to indicate the target user when those buttons have been configured with no pre-set user target. In cases where the other button uses the phone display for target selection 515 this is only possible using User buttons on an associate button module.
- For IP Office 4.2+: The following changes have been made to the indication of user status via BLF (busy lamp field) indicators such as a User button:
  - The status shown for a logged out user without mobile twinning will depend on whether they have **Forward Unconditional** enabled.
    - If they have **Forward Unconditional** enabled the user is shown as idle.
    - If they do not have **Forward Unconditional** enabled they will show as if on DND.
  - The status shown for a logged out user with mobile twinning will be as follows:
    - If there are any calls alerting or in progress through the system to the twinned destination, the user status is shown as alerting or in-use as appropriate. This includes the user showing as busy/in-use if they have such a call on hold and they have **Busy on Held** enabled.
    - If the user enables DND through Mobile Call Control or one-X Mobile client, their status will show as DND.
    - Calls from the system direct to the user's twinned destination number rather than redirected by twinning will not change the user's status.

- **Software Level**: 1.0+.
- **Action**: User.

- Action Data: User name enclosed in "double-quotes".
- Default Label: <*the user name*>.
- Toggles: ✗.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	44XX/64XX	16XX
- Idle.	Extn221	Extn221	Not lit.	Not lit.
- Alerting.	Extn221	Extn221	Green flash.	Red flash.
- In Use/Busy.	Extn221	Extn221	Green on.	Red wink (on with brief flashes).
- DND	Extn221	Extn221	Green on.	Red on.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones:
  - Classic/Comfort icon: Displays the user name.
  - DSS Link LED: On when busy, flashing when call alerting user.

7.3.140 Visual Voice

This action provides the user with a display menu for access to their mailbox. It can be used with Voicemail Pro and Embedded Voicemail. The menu provide the user with options to listening to messages, leaving messages and managing the mailbox.

If pressed when a call is connected, the button allows entry of an extension number for direct to voicemail transfer of the connected call.

On phones that have a display but do not support full visual voice operation as indicated below, use of the button for user mailbox access using voice prompts and for direct to voicemail transfer during a call is supported (does not include T3 and T3 IP phones).

Access to Visual Voice on supported phones can be triggered by the phone's MESSAGES button rather than requiring a separate Visual Voice programmable button. This is done using the System | Voicemail 157 option Messages button goes to Visual Voice.

- Software Level: 4.0+.
- Action: Emulation -> Visual Voice.
- Action Data: *None*.
- Default Label: Voice.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓
  - 1600 Series: ✓
  - 20 Series: ✗
  - 2400 Series: ✓*
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✗
  - 4400 Series: ✓
  - 4600 Series: ✓*
  - 5400 Series: ✓*
  - 5600 Series: ✓*
  - 6400 Series: ✗
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✓*2
  - *Not 1403, 1603, 2402, 5402, 4601, 4602, 5601 and 5602.
  - *2: Takes the user direct to the listen part of Visual Voice. For the full Visual Voice menu options the user should use *Menu -> Settings -> Voicemail Settings*.

Visual Voice Controls

The arrangement of options on the screen will vary depending on the phone type and display size.

- Listen
  Access your own voicemail mailbox. When pressed the screen will show the number of New, Old and Saved messages. Select one of those options to start playback of messages in that category. Use the ▲ up arrow and ▼ arrow keys to move through the message. Use the options below
  - Listen
    Play the message.
  - Pause
    Pause the message playback.
  - Delete
    Delete the message.
  - Save
    Mark the message as a saved message.
  - Call
    Call the message sender if a caller ID is available.
  - Copy
    Copy the message to another mailbox. When pressed a number of additional options are displayed.
- Message
  Record and send a voicemail message to another mailbox or mailboxes.
- Greeting
  Change the main greeting used for callers to your mailbox. If no greeting has been recorded then the default system mailbox greeting is used.
- Email
  This option is only shown if you have been configured with an email address for voicemail email usage in the system configuration. This control allows you to see and change the current voicemail email mode being used for new messages received by your voicemail mailbox. Use Change to change the selected mode. Press Done when the required mode is displayed. Possible modes are:
- Password
  Change the voicemail mailbox password. To do this requires entry of the existing password.

- Voicemail
  Switch voicemail coverage on/off.

Using the Visual Voice Button for Voicemail Transfer

If pressed when you have a call is connected, the MESSAGE button allows entry of an extension number for direct to voicemail transfer of the connected call.

7.3.141 Voicemail Collect

Connects to the voicemail server. The telephone number must indicate the name of the Voicemail box to be accessed, eg. "?Extn201" or "#Extn201". The ? indicates "collect Voicemail" and the # indicates "deposit Voicemail". This action is not supported by voicemail using Intuity emulation mode.

When used with Voicemail Pro, names of specific call flow start points can also be used to directly access those start points via a short code. In these cases ? is not used and # is only used if ringing is required before the start points call flow begins.

- Software Level: 1.1+.
- Action: Advanced -> Voicemail -> Voicemail Collect.
- Action Data: *See above*.
- Default Label: VMCol.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

    1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
    2. May have limited support on some specific T3 phone models if detailed below.

7.3.142 Voicemail Off

Disables the user's voicemail box from answering calls that ring unanswered at the users extension. This button function is largely obsolete as the Voicemail On 670 function toggles on/off.

This does not disable the user's mailbox and other methods of placing messages into their mailbox.

- Software Level: 1.1+.
- Action: Advanced -> Voicemail -> Voicemail Off.
- Action Data: *None*.
- Default Label: VMOff.
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.143 Voicemail On

Enables the user's voicemail mailbox to answer calls which ring unanswered or arrive when the user is busy.

- Software Level: 1.1+.
- Action: Advanced -> Voicemail -> Voicemail On.
- Action Data: *None*.
- Default Label: VMOn.
- Toggles: ✗.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.
- T3 Phones: Supported on Avaya T3 Classic, Comfort and Compact phones for IP Office 4.2+.
  - Classic/Comfort icon: Displays 🔔→✉. The background uses the same settings as the LED below.
  - DSS Link LED: On when set.

7.3.144 Voicemail Ringback Off

This button function is largely obsolete as the Voicemail Ringback On 671 function toggles on/off. Disables voicemail ringback to the user's extension.

- Software Level: 1.1+.
- Action: Advanced -> Voicemail -> Voicemail Ringback Off.
- Action Data: *None.*
- Default Label: VMRB-
- Toggles: ✗.
- Status Indication: ✗.
- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✗
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

7.3.145 Voicemail Ringback On

Enables voicemail ringback to the user's extension. Voicemail ringback is used to call the user when they have new voicemail messages in their own mailbox or a hunt group mailbox for which they have been configured with message waiting indication.

The ringback takes place when the user's phone returns to idle after any call is ended.

- Software Level: 1.1+.
- Action: Advanced -> Voicemail -> Voicemail Ringback On.
- Action Data: None.
- Default Label: VMRB+.
- Toggles: ✓.
- Status Indication: ✓.

Status	24XX/54XX	46XX/56XX	16XX/44XX/64XX
- On.	<Label>◄	<Label>	Green on.
- Off.	<Label>	<Label>	Off.

- User Admin: 651 ✗.
- Phone Support 516
  Note that support for particular phone models is also dependant on the system software level.
  - Analog: ✗
  - 1400 Series: ✓[1]
  - 1600 Series: ✓[1]
  - 20 Series: ✓
  - 2400 Series: ✓[1]
  - 3600 Series: ✗
  - 3700 Series: ✗
  - 3810: ✓
  - 4400 Series: ✓
  - 5400 Series: ✓
  - 4600 Series: ✓[1]
  - 5600 Series: ✓[1]
  - 6400 Series: ✓
  - 9040: ✓
  - 9600 Series: ✓
  - T3/T3 IP Series: ✗[2]

  1. Not 1403, 1603, 4601, 4602, 5601 and 5602 except where 4602 is supported on IP Office 2.1 and 3.0DT software.
  2. May have limited support on some specific T3 phone models if detailed below.

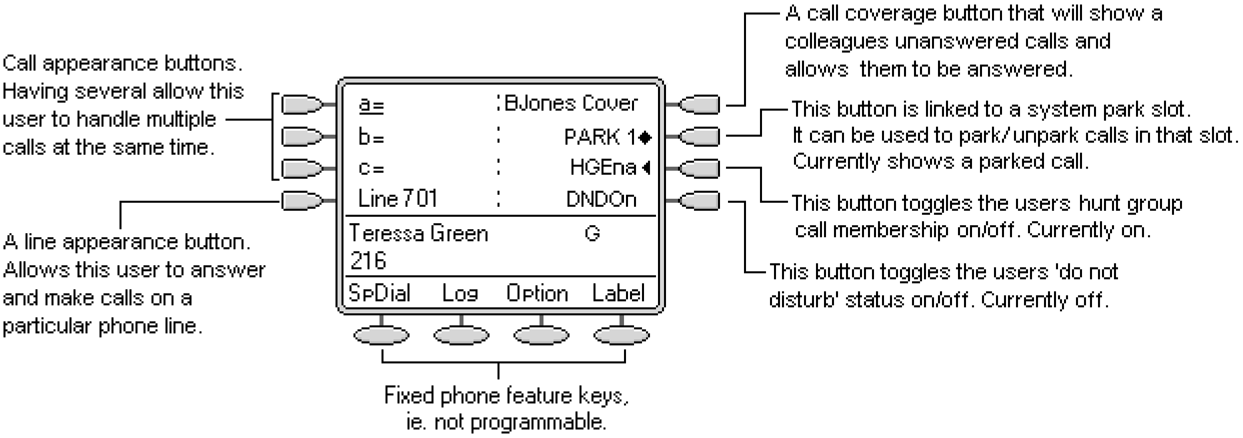
Chapter 8.
Key and Lamp Operation

8. Key and Lamp Operation

IP Office 3.0*. Many Avaya phones supported on system have a programmable keys or buttons (the terms 'key' and 'button' mean the same thing in this context). Various actions can be assigned to each of these keys, allowing the phone user to access that action.

Many of the phones also have indicator lamps next to the programmable buttons. These lamps are used to indicate the status of the button, for example 'on' or 'off'. On other phones the programmable buttons use an adjacent area of the phones display to show status icons and text labels for the buttons.

- Example
  The example below shows the display and programmable buttons on an Avaya 5421 phone where a number of programmable features have been assigned to the user.



- This type of phone displays text labels for the programmed features. On other phones a paper label may have to be updated to indicate the programmed feature.

The system supports four 'appearance' actions - Call Appearance 675, Bridged Appearance 680, Line Appearance 690 and Call Coverage Appearance 686. These actions can be assigned to the programmable buttons on a user's phone. Those 'appearance' buttons can then be used to answer, share, switch between and in some case make calls. This type of call handling is often called 'key and lamp mode'.

This document covers the programming and operation of phones using the appearance functions. Details of the other actions that can be assigned to programmable keys are covered in Button Programming 506.

Note

- *IP Office 3.0DT does not support appearance buttons.
- For all the examples within this documentation, it is assumed that Auto Hold is on and Answer Pre-Select (4.0+) is off unless otherwise stated.
- The text shown on phone displays are typical and may vary between phone types, locales and system software releases.

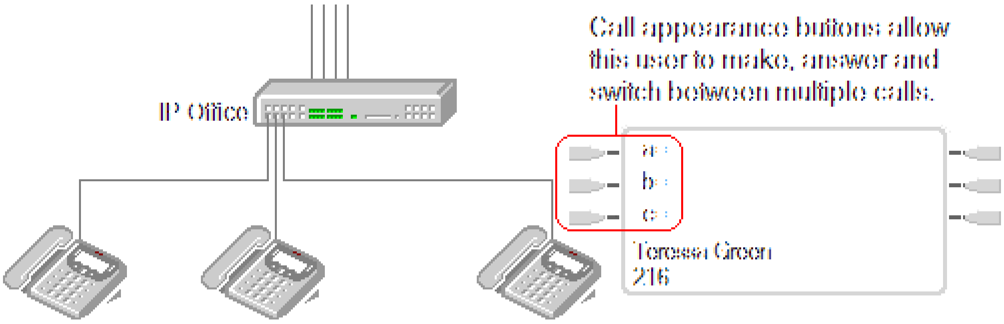
8.1 Appearance Button Features

Appearance functions are only supported on Avaya phones which have programmable buttons and also support multiple calls. Appearance functions are also only supported on those buttons that have suitable adjacent indicator lamps or a display area. Appearance buttons are not supported across Small Community Networks (SCN).

8.2 Call Appearance Buttons

Call appearance buttons are used to display alerts for incoming calls directed to a user's extension number or to a hunt group of which they are a member. Call appearance buttons are also used to make outgoing calls.

By having several call appearance buttons, a user is able to be alerted about several calls, select which call to answer, switch between calls and take other actions.



When all the user's call appearance buttons are in use or alerting, any further calls to their extension number receive busy treatment. Instead of busy tone, the user's forward on busy is used if enabled or otherwise voicemail if available.

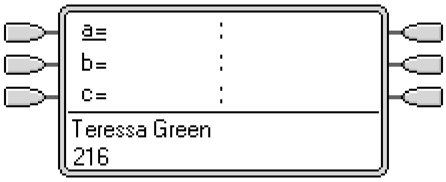
Call appearance buttons are the primary feature of key and lamp operation. None of the other appearance button features can be used until a user has some call appearance button programmed[1].

There are also addition requirements to programming call appearance buttons:

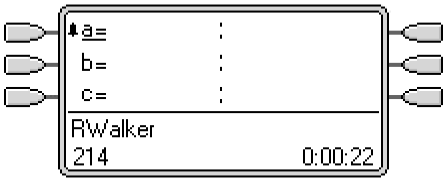
- Call appearance buttons must be the first button programmed for the user.
  - It must be followed by any further call appearance buttons in a continuous block.
  - IP Office 5+: Line appearance buttons can be programmed before call appearance buttons for 1400, 1600 and 9600 Series phone users.
- Programming a single call appearance button for a user is not supported. The normal default is 3 call appearances per user except on phones where only two physical buttons are available.
- [1] For IP Office 4.2+, T3 phones support the use of Line Appearance buttons. These can be programmed against buttons on T3 phones without requiring call appearance buttons. See T3 Phone Line Appearances 695.

8.2.1 Example 1

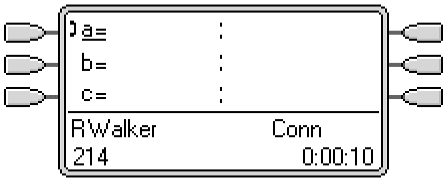
In this example, the user has multiple call appearance buttons.



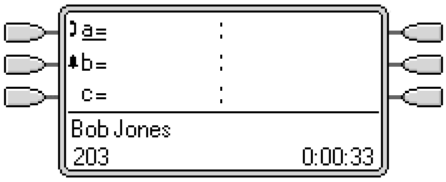
1. Phone Idle
   The phone is currently idle.



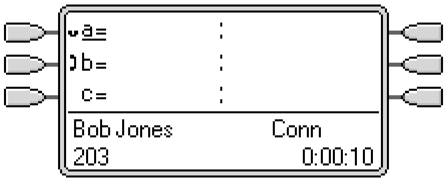
2. First Call Alerts
   A call arrives. It alerts against the first available call appearance button. Pressing that button will answer the call.



3. Call Answered
   The call is now connected.



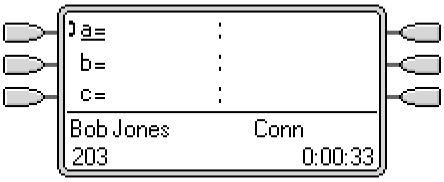
4. Second Call Alerts
   A second call arrives whilst the first is still connected. It alerts against the next available call appearance button. As the user has a call in progress, the alert gives just a single ring and briefly display details of the caller.



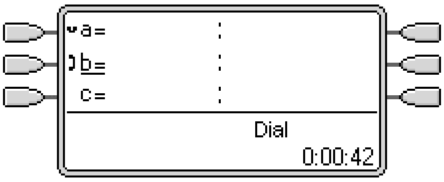
5. Pressing the Second Call Appearance
   Pressing the second call appearance button will hold the first call and answer the second.

8.2.2 Example 2

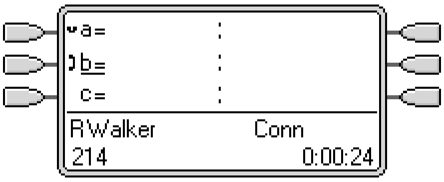
In this example, the user will use their call appearances to make two calls and start a conference between those calls.



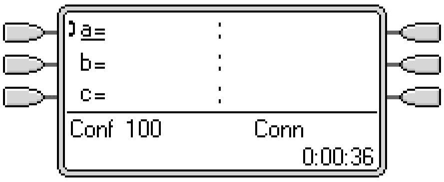
1. Initial Call
   The user has a call in progress, shown on their first call appearance button. It is decided to conference another user into the call.



2. Make Conference Enquiry
   Pressing the CONFERENCE button on the users phone automatically places the current call on hold and takes the phone off hook on the next available call appearance.



3. Enquiry in Progress
   The other extension has been dialed and invited to join a conference call. The user presses the CONFERENCE button on their phone again.



4. Conference Starts
   The conference call has started. The separate call appearances have collapsed to a single appearance that represents the conference.

8.2.3 How are Call Appearance Buttons Treated?

For incoming calls

- Call Waiting settings are ignored except for hunt group call waiting where the call waiting tone is replaced by an alert on a call appearance button if available.
- Follow Me, Forward Unconditional and Forward Hunt Group Calls are used when set.
- If Do Not Disturb is set, only calls from numbers in the user's Do Not Disturb Exception list will alert if a call appearance is available.

Busy status

- For calls direct to the user's extension number
  The user is busy when all their available call appearances are in use. Instead of busy tone, the user's forward on busy is used if enabled or otherwise voicemail if available.
- For calls to a hunt group of which the user is a member
  The user is busy to further hunt group calls when they have any appearance button in use on their phone. The only exception is calls to a collective hunt group with call waiting.
- In both cases above, even when busy, the user may still receive alerts on other appearance buttons; for example call coverage, line appearance and bridged appearance buttons.

For outgoing calls

- Outgoing calls are treated exactly the same as calls made by non-appearance button users.
- External Calls made on a call appearance, which route out on a line for which the user also has a line appearance, will remain on the call appearance. The line appearance will indicate 'in use elsewhere'.

For call appearance buttons matched by a bridged appearance button

- If the bridged appearance is used to make or answer calls, the state of the call appearance will match that of the bridged appearance.
- If the call is put on hold by the bridged appearance user, the call appearance will show 'on hold elsewhere'.

Other

- Held/Parked Call Timeout
  If the user has parked a call, the parked call timer only starts running when the user is idle rather than on another call.
- Incoming calls routed directly to the user as the incoming call routes destination on a line for which the user also has a line appearance, will only alert on the line appearance. These calls do not follow any forwarding set but can be covered.

8.2.4 Button Indication

On phones with a text display area next to the button, by default a=, b= and so on is displayed. This can be replaced by another label if required.

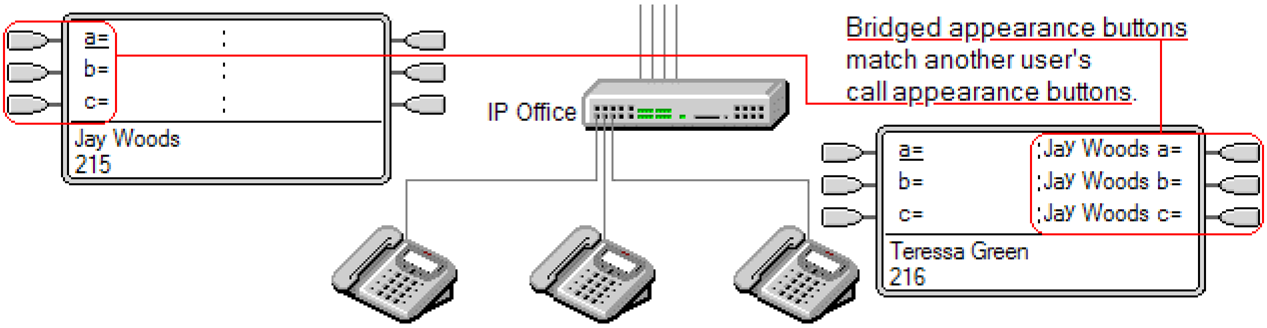
When the user is not connected to a call, the button indicated as selected is the button that will be used if the user goes off hook without pressing an appearance button. When a user is connected to a call, that call is the selected button.

The following table shows how the different states of call appearance buttons (alerting, held, etc) are indicated. This is a general table, not all phone button types are covered. The ring that accompanies the visual indication can be delayed or switched off. See Ring Delay 705.

5410/5420	4400 Series	Call Appearance Button State
CA1	Red off, Green off.	Idle The call appearance is not in use and is not currently selected.
CA1	Red on, Green off.	Idle + Selected The call appearance is not in use but is the current selected button that will be used if the user goes off hook.
CA1 Flashing icon.	Red off, Green steady flash.	Alerting The matching call appearance is alerting for an incoming call. This is accompanied by ringing. If the user is already on a call, only a single ring is given.
CA1 Flashing icon.	Red on, Green steady flash.	Alerting + Selected As above but Ringing Line Preference has made this the user's current selected button.
CA1	Red on, Green on.	In Use Here The user has a call connected on the call appearance or is dialing.
CA1	Red off, Green on.	In Use Elsewhere The call appearance button is in use on a bridged appearance.
CA1	Red off, Green fast flash.	On Hold Here The call has been put on hold by this user.
CA1	Red off, Green intermittent flash.	On Hold Elsewhere A call on a bridged appearance button matched to the call appearance has been put on hold. Calls on a call appearance that are put on hold by another user will continue to show connected lamp status, though the phone display will indicate a held call.
CA1 Icon flashes off.	Red off, Green broken flash.	Inaccessible The button pressed is not accessible. The call is still dialing, ringing or cannot be bridged into.

8.3 Bridged Appearance Buttons

A bridged appearance button shows the state of one of another user's call appearance buttons. It can be used to answer or join calls on that user's call appearance button. It can also be used to make a call that the call appearance user can then join or retrieve from hold.



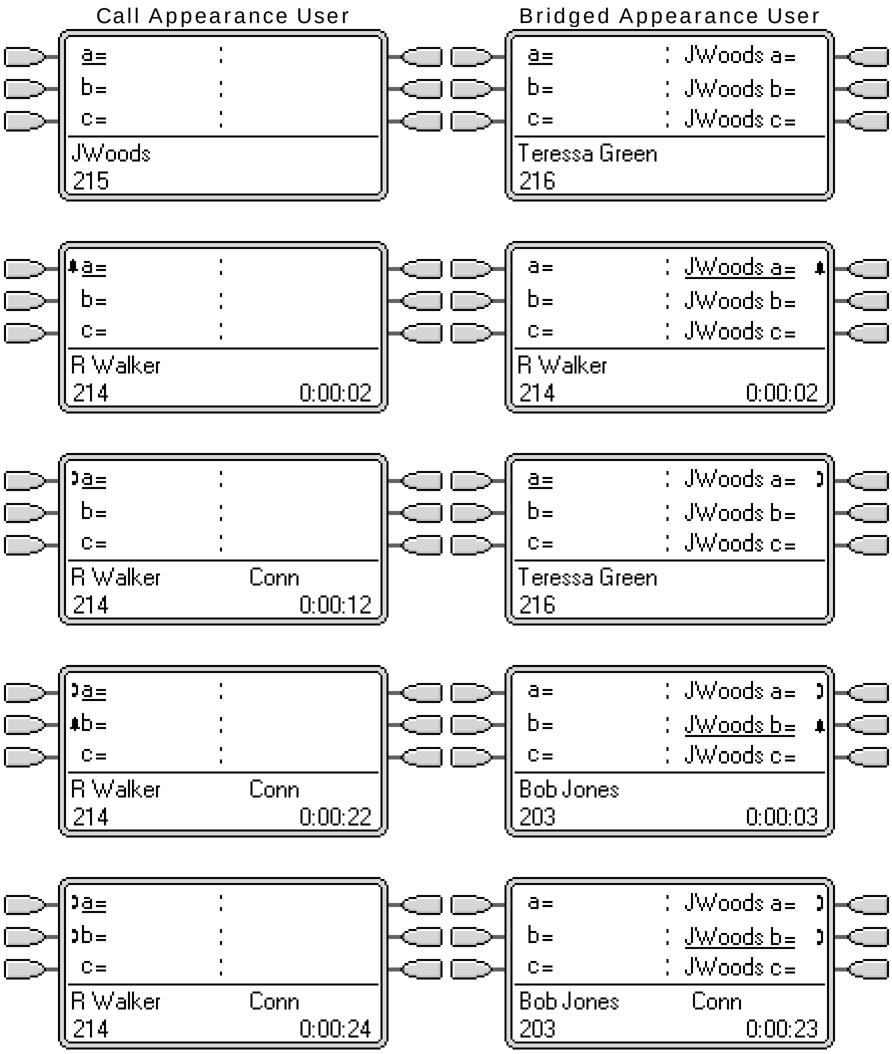
- When the user's call appearance button alerts, any associated bridged appearance buttons on other user's phones also alert. The bridged appearance buttons can be used to answer the call on the call appearance button user's behalf.
- When the call appearance button user answers or makes a call, any associated bridged appearance buttons on other users' phones show the status of the call, ie. active, on hold, etc. The bridged appearance button can be used to retrieve the call if on hold or to join the call if active (subject to intrusion permissions).

- Note
  Bridged appearance buttons are different from the action of bridging into a call (joining a call). See Joining Other Calls (Bridging) 710.

Bridged appearance buttons are not supported between users on different systems in a Small Community Network.

8.3.1 Example 1

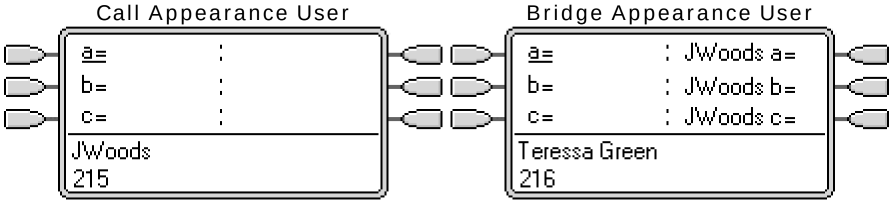
In this example, one user is able to see the status of the other user's call appearances, and when necessary answer calls for the other user. Both users have Ringing Line Preference and Auto Hold on.



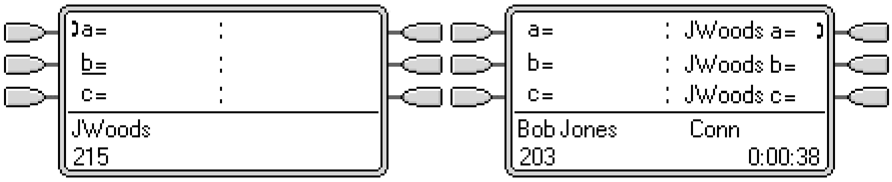
1. Both Phone Idle
   Our user has bridged appearance buttons that match a colleague's call appearances buttons.
2. First Call
   The colleague has a call alerting on their first call appearance button. It also alerts on our user's first bridged appearance button.
3. Call Answered
   The colleague has answered the call. The bridged appearance indicates 'in use elsewhere'.
4. Second Call
   Another call alerts at the colleagues phone and again is mirrored on our user's second bridged call appearance button.
5. Call Answered
   Our user has gone off hook and answered the incoming call alerting on the bridged call appearance.

8.3.2 Example 2

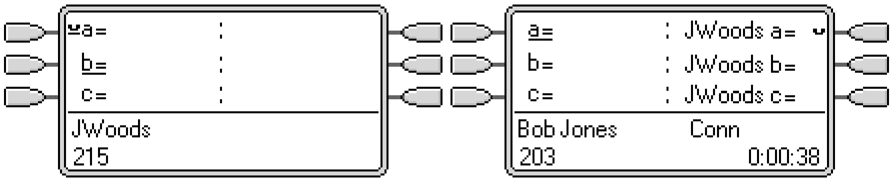
In this example, the bridged appearance user makes a call on behalf of the call appearance user. Once the call is connected, they put it on hold. The call appearance user is able to take the call off hold using their call appearance button. Both users have Ringing Line Preference and Auto Hold on.



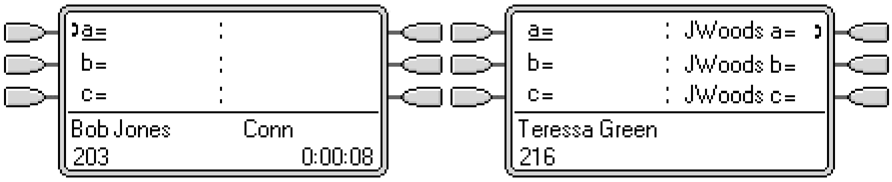
1. Both Phones Idle
   Our user has bridged appearance buttons that match a colleague's call appearances buttons.



2. Bridged User Makes Call
   Our user has pressed a bridged appearance and made a call on it. The matching call appearance shows 'in use elsewhere'.



3. Call Put on Hold
   Having made the call, the bridged user puts it on hold. The matching call appearance indicates 'on hold elsewhere'.

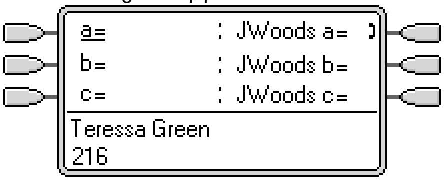


4. Call Taken Off Hold
   By pressing the call appearance, the first user has answered the held call. The bridged appearance user returns to idle.

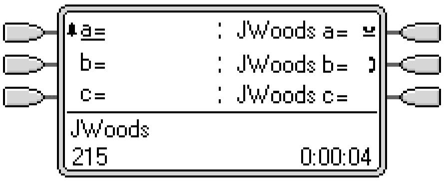
8.3.3 Example 3

In this example, a call is passed from the call appearance user to the bridged appearance user. Both users have Ringing Line Preference and Auto Hold on.

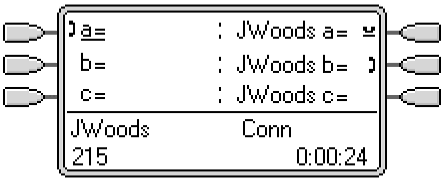
Bridged Appearance User



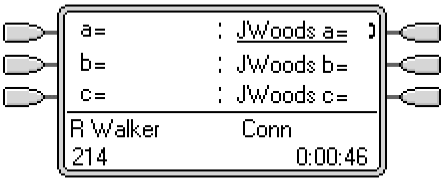
1. Call on Colleague's Phone
   The call appearance user has answered a call on one of their call appearances. The bridged appearance user's matching bridged appearance shows 'in use elsewhere'.



2. Call Held by Colleague
   The call appearance user has put the call on hold and called the bridged appearance user. The first bridged call appearance shows a call 'on hold elsewhere' whilst the second matches the call between users.



3. Enquiry Call Between Colleagues
   By going off hook, the bridged appearance user has answered the call from the call appearance user. They are asked to pickup the call on the colleagues first call appearance.



4. Call Taken Off Hold
   Pressing the first bridged appearance button takes that call off hold and connects it to the bridged appearance user.

   - In this example, Auto Hold is not set for the system, so pressing the bridged appearance button disconnected the call from the colleague.
   - If Auto Hold had been set, the colleague's call would have been put on hold until they hung up.

8.3.4 How are Bridged Appearances Treated?

Bridged appearance buttons operate in parallel with their matching call appearance button.

- Whose user settings control the call?
  Until answered on a bridged appearance button, calls alerting on a bridged appearance button follow the settings of the user or hunt group to which the call was originally directed.
- If the call appearance is in use, any matching bridged appearance will indicate the same.
- If a bridged appearance is in use, the call appearance it matches will indicate the same.
- The bridge appearance will only alert if the call appearance is alerting. For example, direct intercom and paging call to the call appearance will show on the bridged appearance but will not give any audible alert.
- If the bridged appearance user put the call on hold, the call appearance will indicate 'on hold elsewhere'.
- Bridged appearances to a user who has logged out, or has logged into a non-multi line phone, will not operate.
- If the bridged appearance user has 'do not disturb' (DND) enabled, the bridge appearance button icon or lamps will still operate but alerting and ringing line preference selection are not applied unless the caller is in their DND exception list.

Bridged appearance buttons are not supported between users on different systems in a Small Community Network.

8.3.5 Button Indication

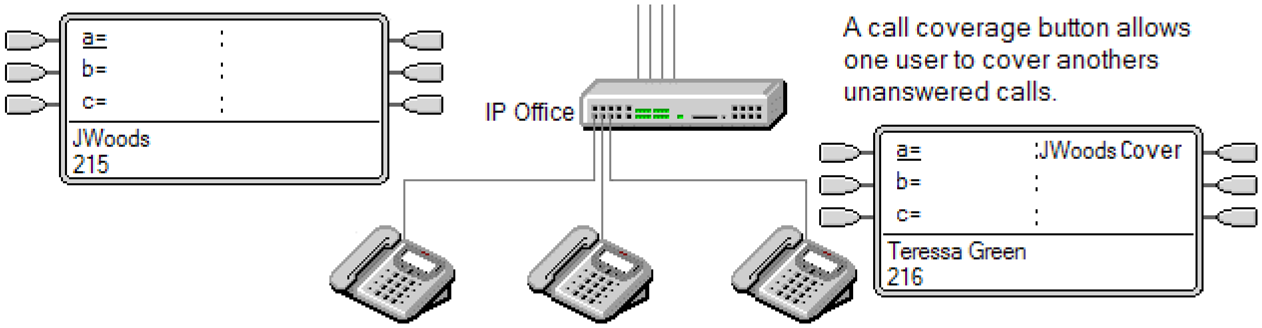
On phones with a text display area next to the button, the name of the bridged user and the label from the bridged user's call appearance key are displayed.

The following table shows how the different states of bridged appearance buttons (alerting, held, etc) are indicated. This is a general table, not all phone button types are covered. The ring that accompanies the visual indication can be delayed or switched off. See Ring Delay 705.

5410/5420	4400 Series	Bridge Appearance Button State
JWoods CA1	Red off, Green off.	**Idle** The bridged appearance is not in use.
JWoods CA1 Flashing icon.	Red off, Green steady flash.	**Alerting** The matching call appearance is alerting for an incoming call. This is accompanied by ringing. If the user is already on a call, only a single ring is given.
JWoods CA1 Flashing icon.	Red on, Green steady flash.	**Alerting + Selected** As above but Ringing Line Preference has made this the user's current selected button.
JWoods CA1	Red off, Green on.	**In Use Elsewhere** The matching call appearance button is in use.
JWoods CA1	Red on, Green on.	**In Use Here** The user has made a call or answered a call on the bridged appearance, or bridged into it.
JWoods CA1	Red off, Green fast flash.	**On Hold Here** The call has been put on hold by this user.
JWoods CA1	Red off, Green intermittent flash.	**On Hold Elsewhere** The call on that call appearance has been put on hold by another user.
JWoods CA1 Icon flashes off.	Red off, Green broken flash.	**Inaccessible** The button pressed is not useable. The call is still dialing, ringing or cannot be bridged into.

8.4 Call Coverage Buttons

Call coverage allows a user to be alerted when another user has an unanswered call.



The user being covered does not necessarily have to be a key and lamp user or have any programmed appearance buttons. Their Individual Coverage Time setting (default 10 seconds) sets how long calls will alert at their extension before also alerting on call coverage buttons set to that user.

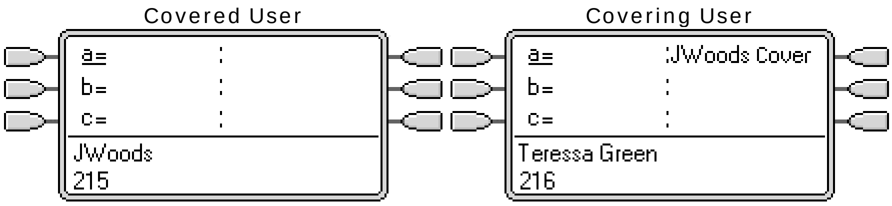
The user doing the covering must have appearance buttons including a call coverage appearance button programmed to the covered users name.

- Note: Call coverage has been supported from IP Office 1.3. However the method of programming and operation has changed with IP Office 3.0. Users of pre-3.0 systems that used call coverage should refer to "Upgrading from Pre-IP Office 3.0" 721.

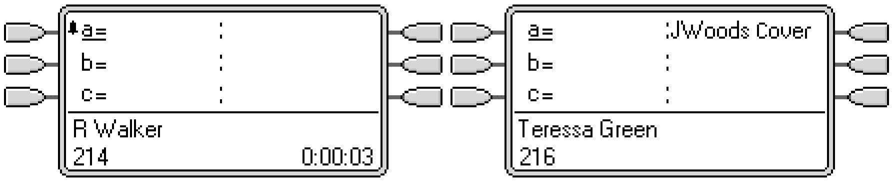
Call coverage appearance buttons are not supported between users on different systems in a Small Community Network.

8.4.1 Example 1

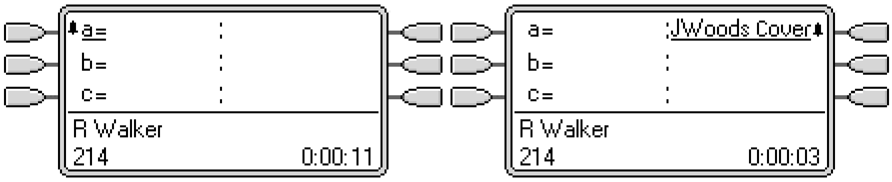
In this example, the covering user is able to answer their colleagues call when it rings unanswered. Both users have Ringing Line Preference and Auto Hold on.



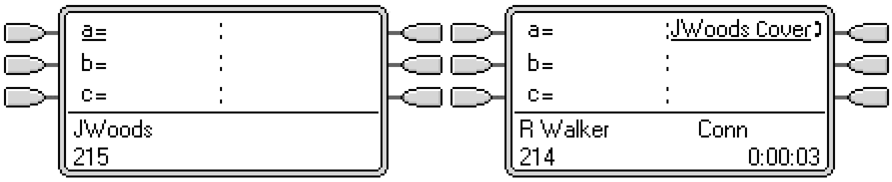
1. Both Phones Idle
   Our user has a call coverage button to cover their colleague.



2. Call to Covered User
   A call arrives for the covered user.



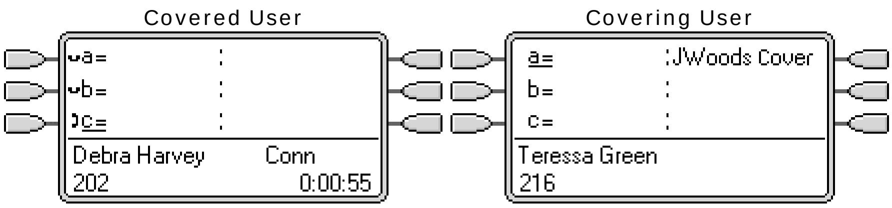
3. Call Alerts to Coverage
   After ringing for the covered user's Individual Coverage Time, the call also begins alerting on the call coverage button .



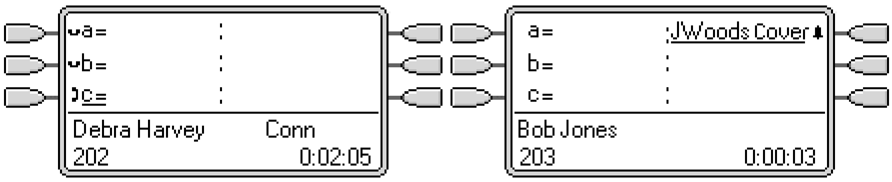
4. Covering User Answers
   By going off hook or pressing the alerting button, the covering user has answered the call.

8.4.2 Example 2

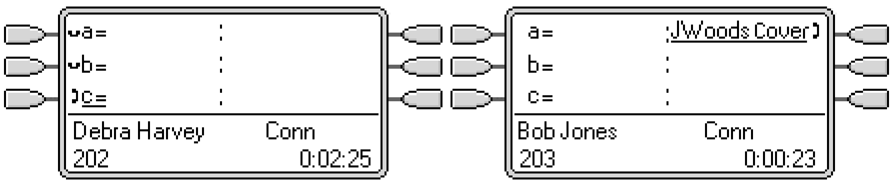
In this example, the covered user has calls on all their available call appearances. Both users have Ringing Line Preference and Auto Hold on.



1. Calls in Progress
   The covered user already has a number of calls in progress on all their call appearance keys.



2. Call Alerts to Coverage
   The covered user is treated as busy, so their next call goes immediately to call coverage.



3. Covering User Answers
   The covering user has answered the call.

8.4.3 How is Call Coverage Treated?

Whose user settings control the call?

- Until answered, calls alerting on a call coverage button follow the settings of the user to which the call was originally directed.
- Once answered, the call follows the user settings of the user who answered it.

Coverage is applied to:

- Internal calls dialed to the covered user's extension number.
- External calls routed to the covered user by a incoming call route.
- Calls forwarded internally by the covered user or on follow me from the covered user.

Coverage is not applied to:

- Hunt group calls to a hunt group of which the covered user is a member.
- Calls forwarded to the covered user using forward or follow me functions.
- Calls alerting on the covered user's bridged appearance and call coverage buttons.
- Coverage is only applied to calls alerting on a line appearance if the call was also routed to that user by an incoming call route.
- Page and intercom calls.
- Parked, transferred and held calls ringing back to the user.
- Automatic callback calls set by the covered user.
- Voicemail ringback calls.
- Call coverage appearance buttons are not supported between users on different systems in a Small Community Network.

Coverage is applied:

- If the covered user's phone is available, call coverage is applied only after the covered user's Individual Coverage Time has expired.
- If the covered user's phone is busy, call coverage is applied immediately.
- If the covered user is using follow me or forward all to an internal number to divert their calls, call coverage is still applied.
- If the covered user has 'do not disturb' on, call coverage is applied immediately except for calls from numbers in the covered user's do not disturb exceptions list.

Other items:

- If the call is not answered after the covered user's **No Answer Time** it will go to the covered user's voicemail if available or follow their forward on no answer settings.
- If the covered user has several alerting calls, the call answered by the call coverage button is the covered user's longest ringing call.
- Calls will not alert at a covering user who has 'do not disturb' enabled, except when the calling number is in the covering user's do not disturb exception list.

8.4.4 Button Indication

On phones with a text display area next to the button, the name of the covered user is displayed followed by the word *Cover*.

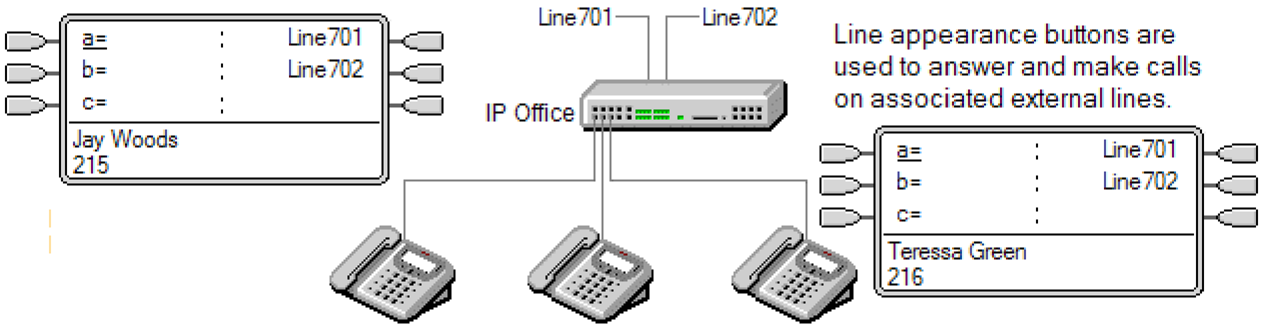
When the user is not connected to a call, the button indicated as selected is the button that will be used if the user goes off hook without pressing an appearance button. When a user is connected to a call, that call is the selected button.

The following table shows how the different states of call coverage appearance buttons (alerting, held, etc) are indicated. This is a general table, not all phone button types are covered. The ring that accompanies the visual indication can be delayed or switched off. See Ring Delay 705.

5410/5420	4400 Series	Call Coverage Button State
JWoods Cover	Red off, Green off.	**Idle** The button is not in use.
JWoods Cover Flashing icon.	Red off, Green steady flash.	**Alerting** The call coverage is alerting for an unanswered call at the covered user's phone. This is accompanied by ringing. If the user is already on a call, only a single ring is given.
JWoods Cover Flashing icon.	Red on, Green steady flash.	**Alerting + Selected** As above but Ringing Line Preference has made this the user's current selected button.
JWoods Cover	Red on, Green on.	**In Use Here** The user has answered the call requiring coverage.
JWoods Cover	Red off, Green fast flash.	**On Hold Here** The covered call has been put on hold by the call coverage button user.

8.5 Line Appearance Buttons

Line appearance buttons allow specific individual line to be used when making calls or answered when they have an incoming call. It also allows users to bridge into calls on a particular line.



Incoming call routing is still used to determine the destination of all incoming calls. Line appearance buttons allow a call on a specific line to alert the button user as well as the intended call destination. When these are one and the same, the call will only alert on the line appearance but can still receive call coverage.

When alerting on suitable phones, details of the caller and the call destination are shown during the initial alert.

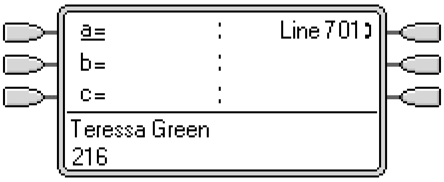
Individual line appearance ID numbers to be assigned to selected lines on a system. Line appearance buttons are only supported for analog, E1 PRI, T1, T1 PRI, and BRI PSTN trunks; they are not supported for other trunks including E1R2, QSIG and IP trunks.

Line appearance buttons are not supported for lines on remote systems in a Small Community Network.

- Using Line Appearances for Outgoing Calls
  In order to use a line appearance to make outgoing calls, changes to the normal external dialing short codes are required. For full details see Outgoing Line Programming 720.
- Private Lines
  Special behaviour is applied to calls where the user has both a line appearance for the line involved and is also the Incoming Call Route destination of that call. Such calls will alert only on the Line Appearance button and not on any other buttons. These calls will also not follow any forwarding.
- T3 Phone Line Appearances
  For IP Office 4.2+, line appearances are supported on T3 and T3 IP phones, see T3 Phone Line Appearances 695.

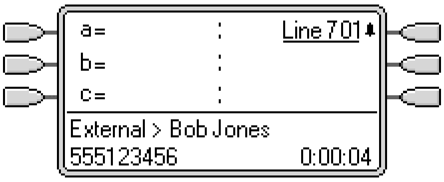
8.5.1 Example 1

In this example, the user is able to answer a call alerting on a particular line.

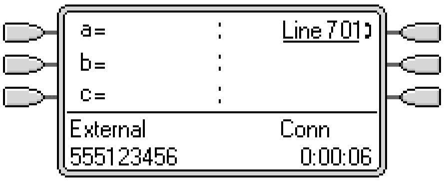


1. Line Goes Active
   A call is active on the line with line ID number 601. This is indicated as 'in use elsewhere'.

   - For an incoming call, the line will show active but will not alert until call routing has been determined. On analog ICLID lines, alerting is delayed until the ICLID that might be used to do the call routing has been received.



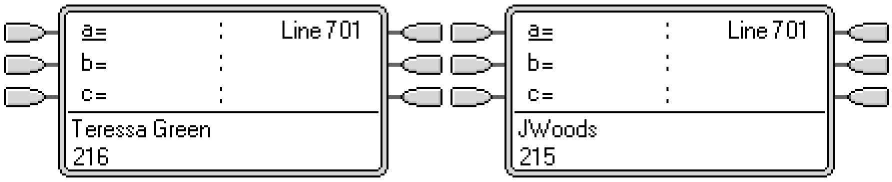
2. Line Appearance Alerting
   The routing of the call has been complete and it is ringing against its destination. On our user's phone the line appearance also alerts and ringing line preference has made it the current selected button.



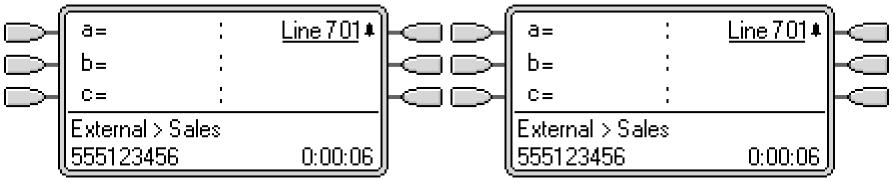
3. Answer Call
   By going off hook or pressing the line appearance, our user has answered the call on that line.

8.5.2 Example 2

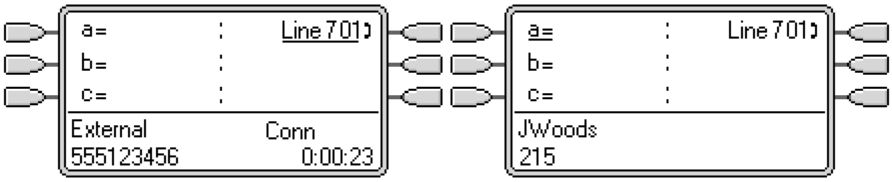
In this example, two users exchange a call using line appearance buttons set to the same line. Note that this requires that the user who first answers the call to have **Cannot be Intruded** off. Both users have **Ringing Line Preference** and **Auto Hold** on.



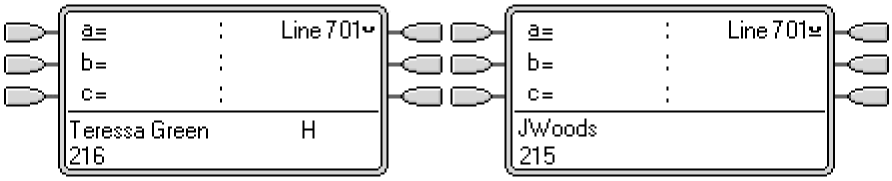
1. **Idle**
   The two users has line appearances for the same line.



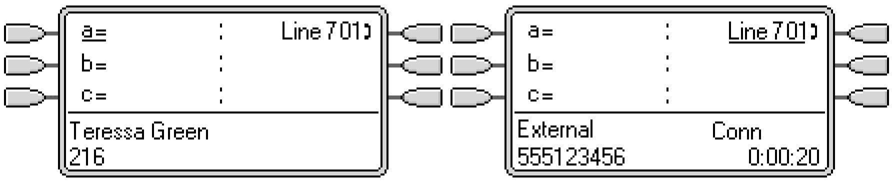
2. **Call Alerts**
   A call arrives. Either user can answer it by pressing the alerting line appearance



3. **Call Answered**
   The first user has answer the call.



4. **Line Held**
   The first user has put the call on hold.



5. **Line Retrieved**
   The second user has retrieved the held call by pressing the line appearance.

8.5.3 How are Line Appearances Treated?

Incoming Calls

- Until answered using a line appearance button, incoming calls alerting on a line appearance, follow the settings of the incoming call route's destination group or user. They do not follow the settings of any line appearance user.
- If an incoming calls destination is voicemail, or once the incoming call has passed from its destination to voicemail, it cannot be answered or bridged into using a line appearance button.
- If the line appearance user is also the incoming call route destination for the call, the call will alert on their line appearance only. In this case:
  - It will alert on the line appearance even if all call appearances are in use.
  - The call will not follow any of the user's forwarding settings .
  - The call will receive call coverage from other user's with call coverage buttons set to the line appearance user.
  - The ring delay used is that of the first free call appearance.
- For analog lines set to ICLID, any line appearances show active while the system waits for ICLID information. During this time the line has not been routed and cannot be answered using a line appearance button.
- Calls alerting on a line appearance can also alert on a call coverage appearance on the same phone. If Ringing Line Preference is set, the current selected button will change from the line appearance to the call coverage appearance.
- If the line appearance user has do not disturb (DND) enabled, the line appearance button icon or lamps will still operate but alerting and ringing line preference selection are not applied unless the caller is in their DND exception list.

Outgoing Calls

- In order to be used for making outgoing calls, some additional system programming may be required. See Outgoing Line Programming 720.
- Calls made on a call appearance, which are routed out on a line for which the user also has a line appearance, will remain on the call appearance. The line appearance will indicate 'in use elsewhere'.

Additional Notes

- Calls alerting on a line appearance do not receive call coverage or go to a users voicemail unless the user was the call's original incoming call route destination.
- If a call indicated by a line appearance is parked, it cannot be joined or unparked by using another line appearance.
- Where a line appearance button is used to answer a call for which automatic call recording is invoked, the recording will go to the automatic recording mailbox setting of the original call destination.
- Line appearance buttons are not supported for lines on remote systems in a Small Community Network.

8.5.4 Button Indication

On phones with a text display area next to the button, the label Line and the line number are displayed.

When the user is not connected to a call, the button indicated as selected is the button that will be used if the user goes off hook without pressing an appearance button. When a user is connected to a call, that call is the selected button.

The following table shows how the different states of line appearance buttons (alerting, held, etc) are indicated. This is a general table, not all phone button types are covered. The ring that accompanies the visual indication can be delayed or switched off. See Ring Delay 705.

5410/5420	4400 Series	Line Appearance Button State
Line 601	All off.	**Idle** The associated line is not in use.
Line 601		**Idle + Selected** The associated line is not in use but the button is the user currently selected button.
Line 601 Flashing icon.	Red off, Green steady flash.	**Alerting** The line is ringing at it incoming call route destination. This is accompanied by ringing. If the user is already on a call, only a single ring is given.
Line 601 Flashing icon.	Red on, Green steady flash.	**Alerting + Selected** As above but Ringing Line Preference has made this the user's current selected button.
Line 601	Red off, Green on.	**In Use Elsewhere** The line is in use.
Line 601	Red on, Green on.	**In Use Here** The user has answered the line, made a call on it or bridged into the call on the line.
Line 601	Red off, Green fast flash.	**On Hold Here** The call on the line has been put on hold by this user.
Line 601	Red off, Green intermittent flash.	**On Hold Elsewhere** The call on the line has been put on hold by another appearance button user.
Line 601 Icon flashes off.	Red off, Green broken flash.	**Inaccessible** The button pressed is not accessible. The call is still dialing, ringing, routing or cannot be bridged into.

8.5.5 T3 Phone Line Appearances

For IP Office 4.2+, line appearances are supported on T3 and T3 IP phones. As these phones do not support support call appearance, bridge appearance or call coverage appearance buttons the user can be programmed with just line appearance buttons.

Soft Key	LED Key	Line Appearance Button State
L601	Off	Idle The associated line is not in use.
✓601	Off	Idle + Selected The associated line is not in use but the button is the user currently selected button.
L601 alternating with bell symbol.	Fast flashing	Alerting The line is ringing at it incoming call route destination. This is accompanied by ringing. If the user is already on a call, only a single ring is given.
L601 alternating with bell symbol.	Fast flashing	Alerting + Selected As above but Ringing Line Preference has made this the user's current selected button.
L601	On	In Use Elsewhere The line is in use.
✓601	On	In Use Here The user has answered the line, made a call on it or bridged into the call on the line.
L601 Slow flash	Slow flash	On Hold Here The call on the line has been put on hold by this user.
L601 Slow flash	Slow flash	On Hold Elsewhere The call on the line has been put on hold by another appearance button user.
-601	Off	Inaccessible The button pressed is not accessible. The call is still dialing, ringing, routing or cannot be bridged into. A single tone is also given.

Notes

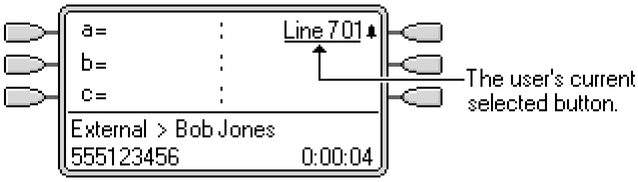
- Hot Desking
  The following applies to appearance button programmed for a user on a system with T3 phones.
  - From a T3 Phone
    If a T3 user with programmed line appearances but no programmed call appearances hot desks onto a phone type that requires call appearances, the phone will not operate correctly. This configuration is not supported by Avaya.
  - To a T3 Phone
    If appearance buttons other than line appearance are programmed for a user, when that user in on a T3 phone those other appearance buttons will be treated as blank. Depending on the button and type of T3 phone the button may assume its default T3 phone function. See T3 Compact 566, T3 Classic 567 and T3 Comfort 568.
- Call Waiting
  Line appearances will ignore the T3 phones user selected call waiting setting. So with a call connected and call waiting off, calls can still alert on line appearances.
- Multiple Calls
  T3 phones are limited to a maximum of 6 associated calls at any time, including calls connected, on hold and alerting.
- Delayed Ringing
  The only Ring Delay options supported are Immediate or No Ring. Any other delayed
- Preference
  Idle line preference is always used, however T3 phones will never default to using a line appearance for an outbound call.
- Joining/Bridging
  Joining a call active on a line appearance is supported. This is subject to the intrusion settings of the users involved. The call then becomes a conference call.

8.6 Other Appearance Controls

8.6.1 Selected Button Indication

During appearance button usage, one of the user's appearance buttons may be indicated as the user's current selected button. This is the appearance button already in use, or if idle, the appearance button that will be used if the user goes off hook by lifting the handset.

- On phones with a display area next to each button, the current selected button is indicated by either an _ underscore of the button label, a * star or a shaded background.



- On phones with twin LED lamps, the current selected button is indicated by the red lamp being on .
- On Transtalk 9040 phones, the current selected button is indicated by a ◄ icon.

The system sets which appearance button is the current selected button using the following methods:

- Idle Line Preference 697
  This feature can be set on or off for each individual user, the default is on. When on, it sets the current selected button as the first available idle call appearance or line appearance button. See Idle Line Preference 697.
- Ringing Line Preference 700
  This feature can be set on or off for each individual user, the default is on. When on, it sets the current selected button as the button which has been alerting at the users phone for the longest. Ringing Line Preference overrides Idle Line Preference. See Ringing Line Preference 700.
- Delayed Ring Preference: 707 *(IP Office 4.0+)*
  This setting is used in conjunction with ringing line preference and appearance buttons set to delayed or no ring. It sets whether ringing line preference should observe or ignore the delayed ring applied to the user's appearance buttons when determining which button should have current selected button status.
- User Selection
  The phone user can override both Idle Line Preference and Ringing Line Preference by pressing the appearance button they want to use or answer. That button will then remain the current selected button whilst active.
  - If the user currently has a call connected, pressing another appearance button will either hold or disconnect that call. The action is determined by the system's Auto Hold 704 setting.
- Answer Pre-Select: 703 *(IP Office 4.0+)*
  Normally when a user has multiple alerting calls, only the details of the call on current selected button are shown. Pressing any of the alerting buttons will answer the call on that button, going off-hook will answer the current selected button. Enabling the user telephony setting Answer Pre-Select allows the user to press any alerting button to make it the current selected button and displaying its call details without answering that call. To answer a call when the user has Answer Pre-Select enabled, the user must press the alerting button to display the call details and then either press the button again or go off-hook.

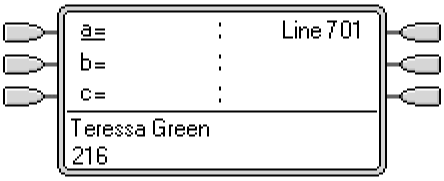
8.6.2 Idle Line Preference

Idle Line Preference determines the user's currently selected button as the first available idle call appearance or line appearance button. Selected button indication 696 is applied to that button and if the user goes off-hook, for example by lifting their handset, an outgoing call is started on that button.

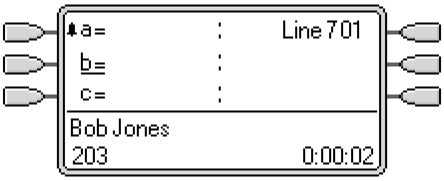
- Idle Line Preference is overridden by Ringing Line Preference if also on for the user.
- By default Idle Line Preference is on for all users.
- For appearance button users with Idle Line Preference off, going off-hook (lifting the handset or pressing SPEAKER, HEADSET, etc) will have no effect until an appearance button is pressed.
- If all the available call appearance and line appearance buttons are in use, no current selected button choice is made by Idle Line Preference. In this case, going off hook will have no effect.
- ?Why Would I Use Just Idle Line Preference
  In environments that are focused on making outgoing calls, for example telemarketing, incoming calls are infrequent and user's go off-hook expecting to make a call. Using Idle Line Preference without Ringing Line Preference ensures that the user doesn't inadvertently answer a call when expecting to make a call.

Idle Line Preference Example 1

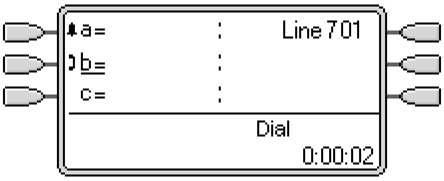
In this example, only Idle Line Preference has been programmed for the user. Ringing Line Preference has not been programmed. The user has three call appearance buttons and one line appearance button programmed.



1. Phone Idle
   The phone is idle. The current selected button determined by Idle Line Preference is the first available idle call appearance button. This is shown by the _ underscore of the button text.



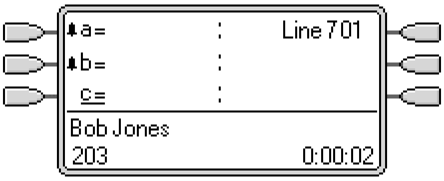
2. First Call to User
   A call for the user arrives. It alerts on the first available call appearance button. Idle Line Preference has changed the current selected button to the next available idle call appearance.



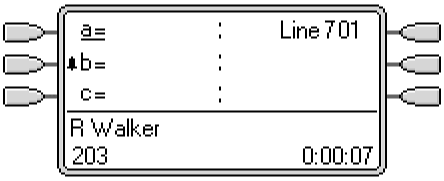
3. User Goes Off Hook
   With the call still alerting, if the user goes off hook, it will be interpreted as making a call using the currently selected button, not as answering the alerting button.
4. To answer the alerting call, the user should press the alerting button.

Idle Line Preference Example 2

In this example, only Idle Line Preference has been programmed for the user. Ringing Line Preference has not been not programmed. The user has three call appearance buttons and one line appearance button programmed.



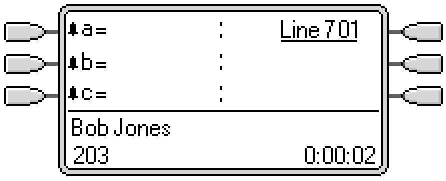
1. Two Calls Alerting
   The users has two incoming calls alerting. Idle Line Preference has set the currently selected button to their third call appearance.



2. First Caller Abandons
   If the first incoming caller disconnects, the currently selected button changes to the first call appearance as this is now the first available idle call or line appearance button.

Idle Line Preference Example 3

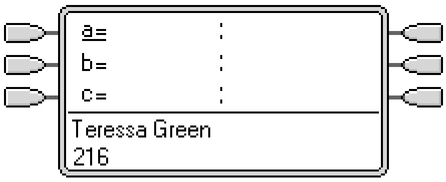
In this example, only Idle Line Preference has been programmed for the user. Ringing Line Preference has not been programmed. The user has three call appearance buttons and one line appearance button programmed.



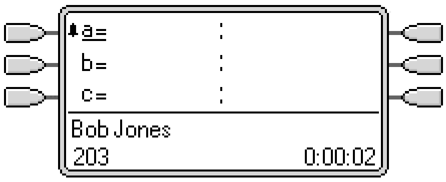
1. All Call Appearances Alerting
   In this case, all the users call appearance buttons are alerting incoming calls. Idle Line Preference has changed the currently selected button to the first available line appearance.

Idle Line Preference Example 4

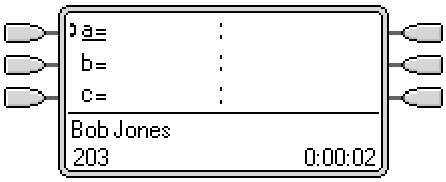
In this example, both Idle Line Preference and Ringing Line Preference are set for the user.



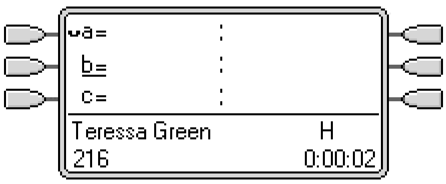
1. Phone Idle
   The phone is idle and *Idle Line Preference* has assigned current selected button to the first call appearance.



2. Call Alerting
   A call has arrived and *Ringing Line Preference* keeps the current selected button at the first call appearance.



3. Call Answered
   With the call answered it retains current selected button status.



4. Call Held
   When the call is put on hold, *Idle Line Preference* assigns current selected button status to the next available call appearance button.

8.6.3 Ringing Line Preference

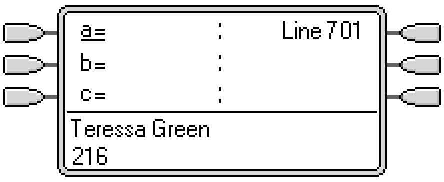
Ringing Line Preference determines the user's currently selected button as the button which has been alerting the longest. Selected button indication 696 is applied to that button and if the user goes off-hook, for example by lifting their handset, the alerting call on that button is answered.

Ringing Line Preference includes calls alerting on call appearance, line appearance, bridged appearance and call coverage buttons.

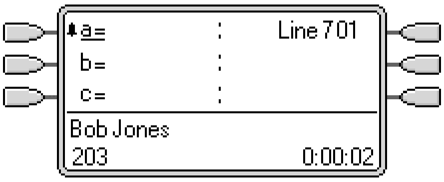
- Ringing Line Preference overrides Idle Line Preference.
- By default Ringing Line Preference is on for all users.
- Ringing Line Preference Order
  When a user's longest waiting call alerts on several of the user's appearance buttons and Ringing Line Preference is set for the user, the order used for current selected button assignment is;
  - Call appearance.
  - Bridged appearance.
  - Call coverage.
  - Line appearance.
- Example:
  A user has a call to a covered user alerting initially on a line appearance button. Ringing Line Preference assigns current selected button status to the line appearance. When the same call also begins to alert on the call coverage appearance button, current selected button status switches to the call coverage appearance button.
- Ring Delay and Ringing Line Preference *(IP Office 3.2+)*
  Appearance buttons can be set to *Delayed Ring* or *No Ring*. These buttons still alert visually but do not give an audible ring or tone. Ringing line preference is still applied to alerting buttons even if set to *Delayed Ring* or *No Ring*.
- Delayed Ring Preference *(IP Office 4.0+)*
  For users with Ringing Line Preference selected, their Delayed Ring Preference setting sets whether ringing line preference is used or ignores buttons that are visually alerting but have *Delayed Ring* or *No Ring* set. The default is off, ie. ignore ring delay.

Ringing Line Preference Example 1

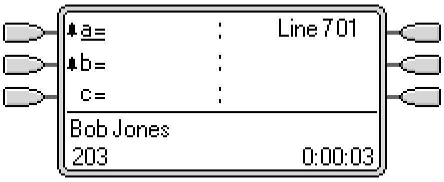
In this example, both Ring Line Preference and Idle Line Preference have been set for the user. The user has three call appearance buttons and one line appearance button programmed. They also have Ringing Line Preference on and Auto Hold is on. Answer Pre-Select is off.



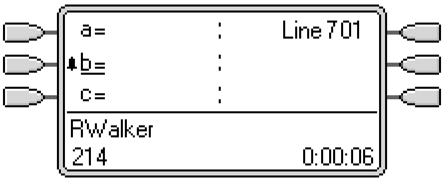
1. Phone Idle
   The phone is idle. The current selected button has been determined by Idle Line Preference as the first available idle call appearance button. This is shown by the _ underscore next to that button.



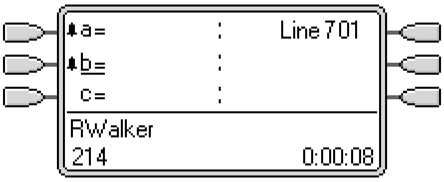
2. First Call Alerting
   A call for the user arrives. It alerts on the first available call appearance button. Ringing Line Preference uses this as the currently selected button as it is the only alerting call.



3. Second Call Alerting
   Another call for the user arrives. It alerts on the next available call appearance button. As the first call has been alerting longer, under Ringing Line Preference it retains the currently select button status.



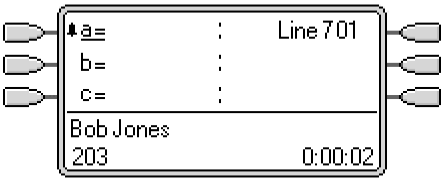
4. The First Call Abandons
   The first caller disconnects. Ringing Line Preference changes the currently selected button status to the second call appearance button.



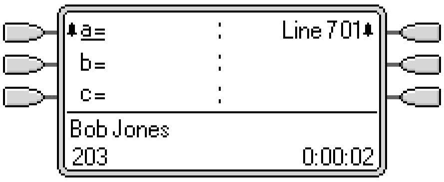
5. Another Call Arrives
   Another call arrives. It alerts as the first free call appearance button. However the call at the second call appearance has been alerting longer and so under Ringing Line Preference retain the currently selected button status.

Ringing Line Preference Example 2

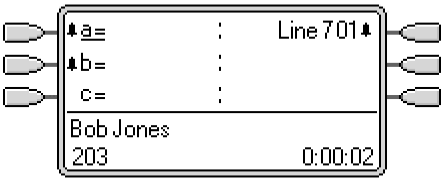
In this example, the user has both Ring Line Preference and Idle Line Preference programmed. The user has three call appearance buttons and one line appearance button programmed. They also have Ringing Line Preference on and Auto Hold is on. Answer Pre-Select is off.



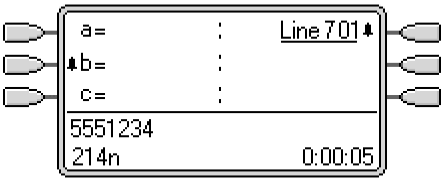
1. First Call to User
   A call for the user arrives. It alerts on the first available call appearance button. Ringing Line Preference uses this as the currently selected button as it is the only alerting call.



2. Call on Line 601
   The user's Line Appearance is alerting due to an incoming call on the associated line. Details of the call and its destination are shown. Ringing Line Preference keeps the currently selected button status on the call appearance button as this has been alerting longest.



3. Second Call to User
   A second call to the user arrives and alerts on the second call appearance button. Ringing Line Preference keeps the currently selected button status on the call appearance button as this has been alerting longest.



4. The First Caller Abandons
   The first call to the user disconnects. Ringing Line Preference passes the currently selected button status to the Line Appearance button as this has been alerting longest.

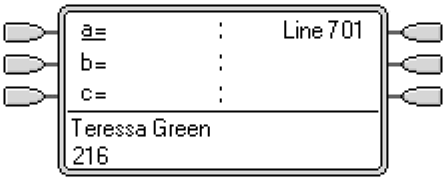
8.6.4 Answer Pre-Select

IP Office 4.0+: On some phones, only the details of the call alerting or connected on the current selected button are shown. The details of calls alerting on other buttons are not shown or only shown briefly when they are first presented and are then replaced again by the details of the call on the current selected button.

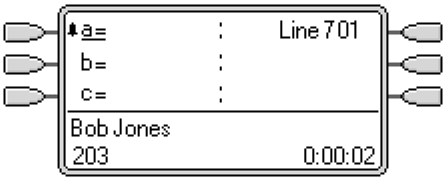
By default, pressing any of the other alerting buttons will answer the call on that button. Answer pre-select allows a user to press alerting buttons other than the current selected button without actually answering them. Instead the button pressed becomes the current selected button and its call details are displayed.

Note that using answer pre-select with a currently connected call will still either hold or end that call in accordance with the system's Auto Hold 704 setting.

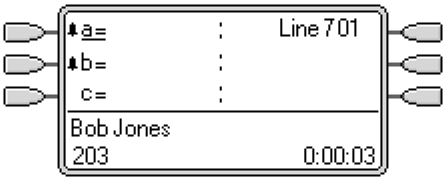
Answer Pre-Select Example 1



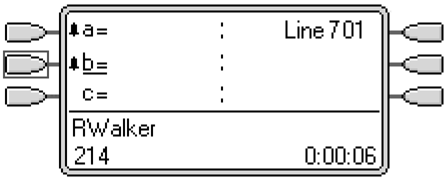
1. Phone Idle
   The phone is idle. The current selected button has been determined by Idle Line Preference as the first available idle call appearance button. This is shown by the _ underscore next to that button.



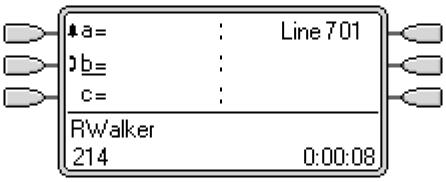
2. First Call Alerting
   A call for the user arrives. It alerts on the first available call appearance button. Ringing Line Preference uses this as the currently selected button as it is the only alerting call.



3. Second Call Alerting
   Another call for the user arrives. It alerts on the next available call appearance button. As the first call has been alerting longer, under Ringing Line Preference it retains the currently select button status.



4. The User Presses the Second Call Appearance
   Pressing the second call appearance overrides ringing line preference and assigns current selected button status to the button without actually answering the call. The details of the caller are displayed.



5. The User Answers the Call
   The user can press the button again to answer the call or just go off-hook to answer as it is now the currently selected button.

8.6.5 Auto Hold

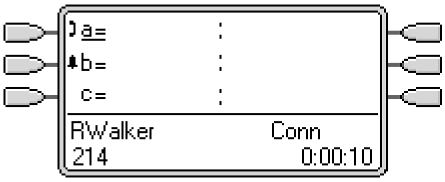
Auto Hold is a system wide feature that affects all appearance button users. This feature determines what happens when a user, who is already on a call, presses another appearance button. The options are:

- If Auto Hold is *off*, the current call is disconnected.
- If Auto Hold is *on*, the current call is placed on hold.

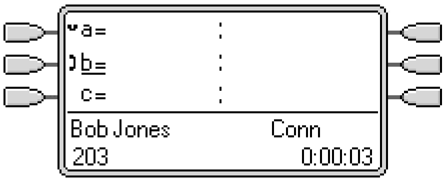
On IP Office 4.0 and higher systems Auto Hold is *on* by default. On previous levels of system software the default for US was *off*.

Auto Hold Example 1

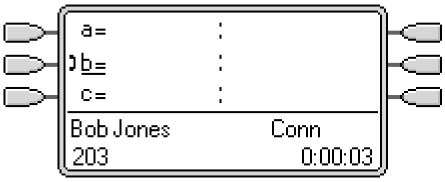
In this example, the user has two calls currently shown on call appearance buttons. Answer Pre-Select is off.



1. This user has three call appearance buttons. They have answer one call and are still connected to it, shown by the icon. A second call is now alerting on their second call appearance button, shown by the icon.
2. What happens when the user presses the second call appearance key is determined by the system's Auto Hold setting:



- Auto Hold On
  When the second call appearance key is pressed, that call is answered and the first call is put on hold, shown by the icon. The user can switch between calls using the call appearance buttons and make/receive other calls if they have additional call appearance buttons



- Auto Hold Off
  When the second call appearance key is pressed, that call is answered and the first call is disconnected.

8.6.6 Ring Delay

IP Office 3.2+: Ring delay can be applied to appearance buttons. This option can be used with all types of appearance buttons and can be selected separately for each appearance button a user has. Using ring delay does not affect the buttons visual alerting through the display and display icons or button lamps.

Ring delay is typically used with line appearance buttons for lines which a user wants to monitor but does not normally answer. However ring delay can be applied to any type of appearance button.

The selectable ring delay options for an appearance button are listed below. The option is selected as part of the normal button programming process.

- Immediate
  Provide audible alerting as per normal system operation.
- Delayed Ring
  Only provide audible alerting after the system ring delay or, if set, the individual user's ring delay.
- No Ring
  Do not provide any audible alerting.

There are two possible sources for the delay used when delayed ringing is selected for a button.

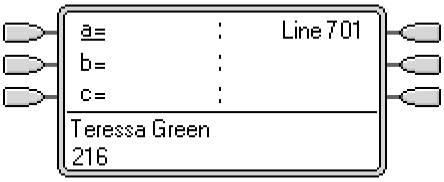
- System | Telephony | Telephony | Ring Delay: 160 *Default = 5 seconds, Range 1 to 98 seconds.*
  This is the setting used for all users unless a specific value is set for an individual user.
- User | Telephony | Multi-line Options | Ring Delay: 278 *Default = Blank (Use system setting), Range 1 to 98 seconds.*
  This setting can be used to override the system setting. It allows a different ring delay to be set for each user.

Notes

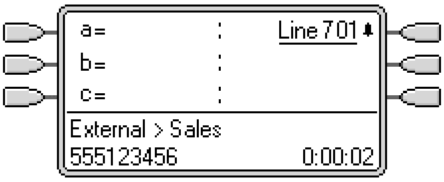
- Calls That Ignore Ring Delay
  Ring delay is not applied to hold recall calls, park recall calls, transfer return calls, voicemail ringback calls and automatic callback calls. For phones using Internal Twinning, ring delay settings are not applied to calls alerting at a secondary twinned extension (except appearance buttons set to *No Ring* which are not twinned).
- Auto Connect Calls
  IP Office 3.2/4.0: Auto-connected calls such as Internal Auto-Answer override Ring delay. IP Office 4.1+: Ring delay is applied to these calls before auto-connection. This does not apply to page calls.
- Multiple Alerting Buttons
  Where a call is presented on more than one button on a user's phone, see Multiple Alerting Buttons 712, the shortest delay will be applied for all the alerting buttons. For example, if one of the alerting buttons is set to Immediate, that will override any alerting button set to Delayed Ring. Similarly if one of the alerting buttons is set to No Ring, it will be overridden if the other alerting button is set to Immediate or Delayed Ring.
- Line Appearance Buttons
  Calls routed to a user that could potentially be presented on both a call appearance button and a line appearance button are only presented on the line appearance button. In this scenario, the ring delay settings used is that of the first free call appearance button.
- Delay on Analog Lines
  Analog lines set to Loop Start ICLID already delay ringing whilst the system waits for the full ICLID in order to resolve incoming call routing. In this scenario the ring delay operates in parallel to the routing delay.
- Ring Delay and Ringing Line Preference
  Appearance buttons can be set to *Delayed Ring* or *No Ring*. However ringing line preference is still applied to alerting buttons even if set to *Delayed Ring* or *No Ring*.
  - IP Office 4.0+: The user's Delayed Ring Preference setting is used to determine whether ringing line preference is used with or ignores buttons that are alerting but have *Delayed Ring* or *No Ring* set.

Ring Delay Example 1

In this example, the user has a line appearance button set but configured to no ring.



1. **Phone Idle**
   The phone is idle. The current selected button has been determined by Idle Line Preference as the first available call appearance button. This is shown by the _ underscore next to that button.



2. **Incoming Call Alerting on the Line**
   An incoming call arrives on the line and begin to alert somewhere on the system. The user's line appearance button shows this visually but doesn't ring audibly. Ringing line preference would makes the line appearance the user's currently selected button and therefore they would answer the line if they went off-hook.

8.6.7 Delayed Ring Preference

When a call is alerting at an idle phone, by default Ringing Line Preference 700 sets the call as the currently selected button and if the user then goes off-hook they will answer that call.

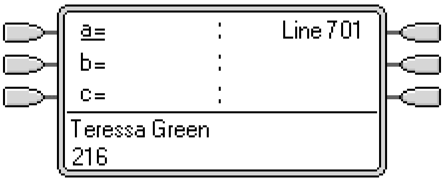
In most situations this is acceptable as the user hears ringing which informs them that there is a call waiting to be answered. If the user wants to make a call instead, they can press another call appearance button to go off-hook on that other button.

When ring delay is being used there can potentially be a problem if the user lifts the handset to make a call without looking at the display. If they do this while the a call is alerting silently on a button with ring delay, the user will actually answer the waiting call rather than get dial tone to make a call.

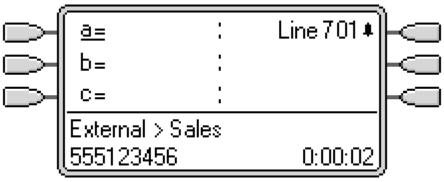
Once the call alerting on a button has currently selected call status, it retains that status even if a prior call on a button with ring delay applied comes out of its ring delay period.

Delayed Ring Preference Example 1

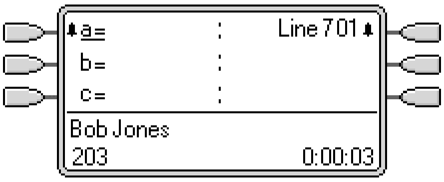
In this example the user has a line appearance button for a line they monitor. This line appearance button has been set to no ring as the user occasionally need to use that line but does not normally answer calls on that line.



1. Phone Idle
   The phone is idle. The current selected button has been determined by Idle Line Preference as the first available call appearance button. This is shown by the _ underscore next to that button.



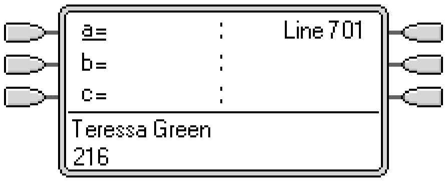
2. Incoming Call Alerting on the Line
   An incoming call arrives on the line and begin to alert somewhere on the system. The user's line appearance button shows this visually but doesn't ring audibly.
   - Normally ringing line preference would make the line appearance the user's currently selected button and therefore they would answer the line if they went off-hook expecting to make a call.
   - However, because Delayed Ring Preference is on for the user, ringing line preference is not applied and idle line preference makes their current selected button the first call appearance. If the user were to go off-hook they would be making a call on that call appearance.



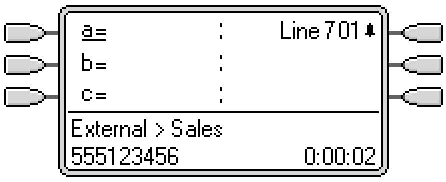
3. Call Alerting for the User
   A call for the user arrives. It alerts on the first available call appearance button. Ringing line preference is applied and makes that the users currently selected button. If the user goes off-hook now that will answer the call on the call appearance and not the line appearance.

Delayed Ring Preference Example 2

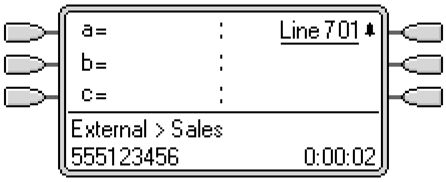
This is similar to the previous example except that the user and the line has been configured for a 15 second ring delay. This informs the users that the line has not been answered for some reason and allows them to answer it by just going off-hook.



1. Phone Idle
   The phone is idle. The current selected button has been determined by Idle Line Preference as the first available call appearance button. This is shown by the _ underscore next to that button.



2. Incoming Call Alerting on the Line
   An incoming call arrives on the line and begin to alert somewhere on the system. The user's line appearance button shows this visually but doesn't ring audibly. Because Delayed Ring Preference is on for the user, ringing line preference is not applied and idle line preference makes their current selected button the first call appearance. If the user were to go off-hook they would be making a call on that call appearance.



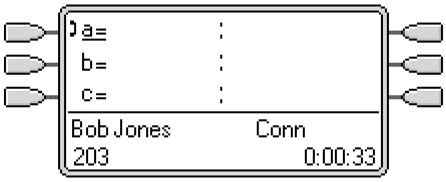
3. Call Continues Alerting
   When the ring delay for the line appearance expires, if no other call has taken ringing line preference it becomes the current selected call and will be answered if the user goes off-hook.

8.6.8 Collapsing Appearances

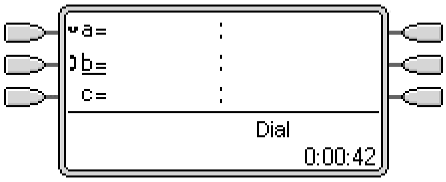
This topic covers what happens when a user with several calls on different appearance buttons, creates a conference between those calls. In this scenario, the call indication will collapse to a single appearance button and other appearance buttons will return to idle except for any line appearance buttons involved which will show 'in use elsewhere'.

Collapsing Appearances Example 1

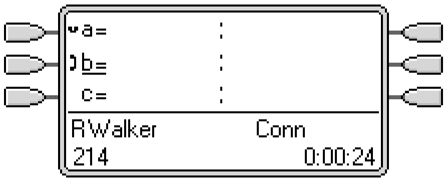
In this example, the user will setup a simple conference. Ringing Line Preference and Idle Line Preference are set for the user. Auto Hold for the system is on. Answer Pre-Select is off.



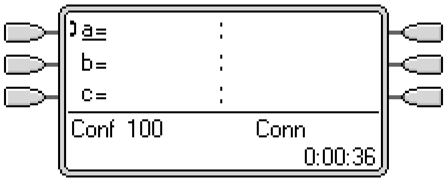
1. Initial Call
   The user has a call in progress, shown on their first call appearance button. It is decided to conference another user into the call.



2. Make Conference Enquiry
   Pressing the CONFERENCE button on the users phone automatically places the current call on hold and takes the phone off hook on the next available call appearance.



3. Enquiry in Progress
   The other extension has answered and is invited to join a conference call. The user presses the CONFERENCE button on their phone again.



4. Conference Starts/Call Appearances Collapse
   The conference call has started. The call appearances have collapsed to a single appearance.

8.6.9 Joining Other Calls (Bridging)

Appearance buttons can be used to "join" existing calls and create a conference call. A user can join calls that are shown on their phone as 'in use elsewhere'. These can be on bridged and/or line appearances.

- This feature is often referred to as 'bridging into a call'. However this causes confusion with Bridged appearance buttons and so the term should be avoided.

The ability to join calls is controlled by the following feature which can be set for each user:

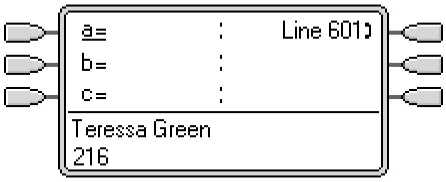
- Cannot be Intruded: *Default = On*
  If this option is set on for the user who has been in the call the longest, no other user can join the call. If that user leaves the call, the status is taken from the next internal user who has been in the call the longest. The exceptions are:
  - Voicemail and Conferencing Center calls are treated as Cannot be Intruded at all times.
  - When an external call is routed off switch by a user who then leaves the call, the Cannot be Intruded status used is that of the user who forwarded the call off switch.
  - Any call that does not involve an internal user at any stage is treated as Cannot be Intruded on. For example:
    - When an external call is routed off switch automatically using a short code in the incoming call route.
    - Small Community Network calls from other systems that are routed off-switch.
    - VoIP calls from a device not registered on the system.
  - The Can Intrude setting is not used for joining calls using appearance buttons.

The following also apply:

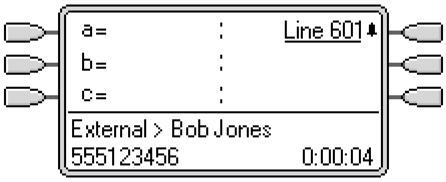
- Inaccessible
  In addition to the use of the Cannot be Intruded setting above, a call is inaccessible if:
  - The call is still being dialed, ringing or routed.
  - It is a ringback call, for example a call timing out from hold or park.
  - If all the internal parties, if two or more, involved in the call have placed it on hold.
- Conferencing Resources
  The ability to bridge depends on the available conferencing resource of the system. Those resources are limited and will vary with the number of existing parties in bridged calls and conferences. The possible amount of conferencing resource depends on the system type and whether Conferencing Center is also installed.
- Conference Tone
  When a call is joined, all parties in the call hear the system conferencing tones. By default this is a single tone when a party joins the call and a double-tone when a party leaves the call. This is a system setting.
- Holding a Bridged Call
  If a user puts a call they joined on hold, it is their connection to the joined call (conference) that is put on hold. The other parties within the call remain connected and can continue talking. This will be reflected by the button status indicators. The user who pressed hold will show 'on hold here' on the button they used to join the call. All other appearance users will still show 'in use here'.
- Maximum Two Analog Trunks
  Only a maximum of two analog trunks can be included in a conference call.
- Parked Calls
  A Line Appearance button may indicate that a call is in progress on that line. IP Office 4.0+: Such calls to be unparked using a line appearance.

Joining Example 1: Joining with a Line Appearance

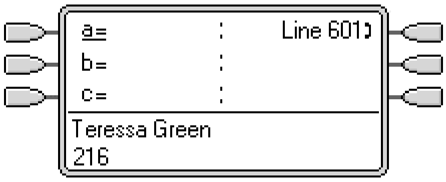
In this example, the user joins a call by pressing a line appearance button. Answer Pre-Select is off.



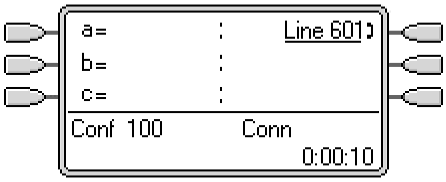
1. Line Goes Active
   A call is active on the line with line ID number 601.
   - If this is an incoming call, it will show active but will not alert until its call routing has been determined. On ICLID analog lines, alerting is delayed until the ICLID that might be used to do that routing has been received.



2. Line Appearance Alerting
   The call routing is completed and the call is now ringing against its target. The line appearance also begins alerting and Ringing Line Preference has made it the current selected button.



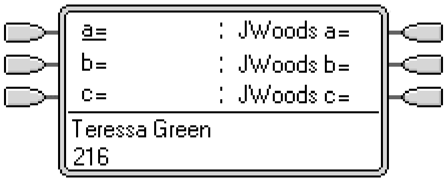
3. Call Answered
   Alerting on the line appearance has stopped but the line is still active. This indicates that the call has probably been answered. As our user's phone is idle, Idle Line Preference has returned the current select button to the first available call appearance button.



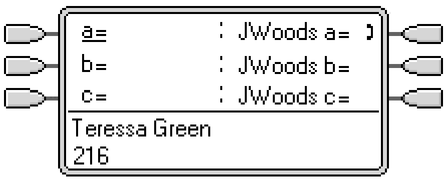
4. User Joins the Call
   Our extension user has been asked by their colleague to join the call just answered on line 601. By pressing the line appearance button they have joined the call on that line and created a conference call.

Joining Example 2: Joining with a Bridged Appearance

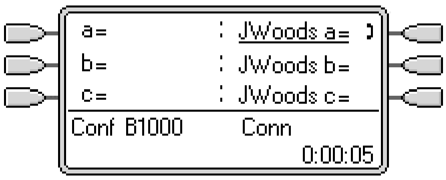
In this example, the user joins a call using a bridged appearance button. Answer Pre-Select is off.



1. User with Bridged Appearance Buttons
   The user has bridged appearance buttons that match their colleagues call appearance buttons.



2. Call on Bridged Appearance
   The colleague has a call in progress on their first call appearance. This is matched on the first bridged appearance button.



3. User Joins the Call
   Pressing the bridged appearance button will take our user off hook and join them into their colleagues call, creating a conference call.

8.6.10 Multiple Alerting Buttons

In some scenarios, it is may be potentially possible for the same call to alert on several appearance buttons. In this case the following apply:

- Line appearance buttons override call and bridged appearance buttons
  In cases where a call on a line goes directly to the user as the incoming call route's destination, the call will only alert on the line appearance. In this scenario the ring delay settings used is that of the first free call appearance button.
- A call can alert both call appearance, line appearance and bridged appearance buttons
  The most common example of this will be hunt group calls where the hunt group members also have bridged call appearances to each other. In this case the button used to answer the call will remain active whilst the other button will return to idle.
- Calls on a line or bridged appearance buttons can also alert on call coverage button
  In this case alerting on the call coverage button may be delayed until the covered user's Individual Coverage Time has expired.
- Ringing Line Preference Order
  When a call alerts on several of the user's appearance buttons and *Ringing Line Preference* is set for the user, the order used for current selected button assignment is:
  1. Call appearance.
  2. Bridged appearance.
  3. Call coverage.
  4. Line appearance.

- Example
  A user has a call to a covered user alerting initially on a line appearance button. *Ringing Line Preference* will assign current selected button status to the line appearance. When the same call also begins to alert on the call coverage appearance button, current selected button status switches to the call coverage appearance button.

Ring Delay
Where ring delays are being used, the shortest delay will be applied for all the alerting buttons. For example, if one of the alerting buttons is set to Immediate, that will override any alerting button set to Delayed Ring. Similarly if one of the alerting buttons is set to No Ring, it will be overridden if the other alerting button is set to Immediate or Delayed Ring.

8.6.11 Twinning

Twinning is a mechanism that allows an user to have their calls alert at two phones. The user's normal phone is referred to as the primary, the twinned phone as the secondary.

By default only calls alerting on the primary phone's call appearance buttons are twinned. For internal twinning, the system supports options to allow calls alerting on other types of appearance buttons to also alert at the secondary phone. These options are set through the User | Twinning section of the system configuration and are Twin Bridge Appearances, Twin Coverage Appearances and Twin Line Appearances. In all cases they are subject to the secondary having the ability to indicate additional alerting calls.

Call alerting at the secondary phone ignoring any Ring Delay settings of the appearance button being used at the primary phone. The only exception is buttons set to No Ring, in which case calls are not twinned.

8.6.12 Busy on Held

For a user who has Busy on Held selected, when they have a call on hold, the system treats them as busy to any further calls. This feature is intended primarily for analog phone extension users. Within Manager, selecting Busy on Held for a user who also has line appearance keys will cause a prompt offering to remove the Busy on Held selection.

8.6.13 Reserving a Call Appearance Button

Functions such as transferring calls using a Transfer key require the user to have at least one available call appearance button in order to complete the outgoing call part of the process. However by default all call appearance button are available to receive incoming calls at all times.

It is possible to reserve the users last call appearance button for the making of outgoing calls only. The method for doing this depends on the system software level.

- Pre-4.0 IP Office: On the User | Source Numbers tab, enter the line RESERVE_LAST_CA= .
- IP Office 4.0+: On the User | Telephony | Multi-line Options tab, select the option Reserve Last CA.

8.6.14 Logging Off and Hot Desking

Users can be setup to log in and log out at different phones, this is called 'hot desking'. All the users settings, including their extension number, are transferred to the phone at which the user is logged in. This includes their key and lamp settings and appearance buttons.

This type of activity has the following effect on appearance buttons:

- If logged out, or logged in at a phone that doesn't support appearance button functions:
  - Bridged appearances set to the user will be inactive.
  - Call coverage set to the user will still operate.

- If logged in at a phone with fewer buttons than programmed for the user:
  - Those buttons which are inaccessible on the logged in phone will be inactive.
  - Any bridged appearances to those button from other users will be inactive.

Remote Hot Desking

IP Office 4.0 supports, through the addition of license keys, users hot desking between systems within a Small Community Network (SCN). However the use of appearance buttons (call coverage, bridged appearance and line appearance) within a Small Community Network is not supported. Therefore when a user logs in to a remote system, any such button that they have will no longer operate. Similarly any button that other users have with the remote user as the target will not operate.

8.6.15 Applications

A number of system applications can be used to make, answer and monitor calls. These applications treat calls handled using key and lamp operation follows:

- Phone Manager / SoftConsole
  These applications are able to display multiple calls to or from a user and allow those calls to be handled through their graphical interface.
  - All calls alerting on call appearance buttons are displayed.
  - Calls on line, call coverage and bridged appearance buttons are not displayed until connected using the appropriate appearance button
  - Connected and calls held here on all appearance button types are displayed.
  - The status of alerting call appearance calls, connected and held calls is shown in the Phone Manager's call status panel. Clicking on a call here can be used to answer or unhold a call. This action will place any current connected call on hold regardless of the system's Auto Hold setting.

8.7 Programming Appearance Buttons

This section covers the programming of appearance buttons for users into existing system configurations.

- Appearance Functions
  The functions *Call Appearance*, *Bridged Appearance*, *Coverage* and *Line Appearance* are collectively known as "appearance functions". For full details of their operation and usage refer to the Key and Lamp Operation 674 section. The following restrictions must be observed for the correct operation of phones.

Software	
2.1 or 3.0DT	On systems running IP Office 2.1 or 3.0DT software, the first programmable button must be set as a call appearance button. This requirement is necessary for the correct operation of phone functions such as call log.
3.0+	For systems running 3.0 or higher (excluding 3.0DT), the first 3 programmable buttons must be set as call appearance buttons. This requirement is necessary for the correct operation of functions such as transfer, conference and call log. • On phones with only two physical programmable buttons both buttons must be used as call appearance buttons.
4.2+	T3 phones support Line Appearance buttons only and are exempt from the above rules.
5.0+	• Users can have line appearances programmed before call appearances. This option is only supported by users using 1400, 1600 and 9600 Series phones. • On 1400, 1600 and 9600 Series phones, all the appearance buttons must be programmed as a continuous block with no gaps between the appearance buttons.

- Appearance functions programmed to buttons without suitable status lamps or icons are treated as disabled. These buttons are enabled when the user logs in on a phone with suitable buttons in those positions.
- Line appearance buttons require line ID numbers to have been assigned, see Programming Line Appearance Numbers 719. The use of line appearances to lines where incoming calls are routed using DID (DDI) is not recommended.
- How many buttons are allowed?
  The recommended limits are as follows:
  - A maximum of 10 bridged appearances to the same call appearance.
  - A maximum of 10 line appearances to the same line.
  - A maximum of 10 call coverages of the same covered user.

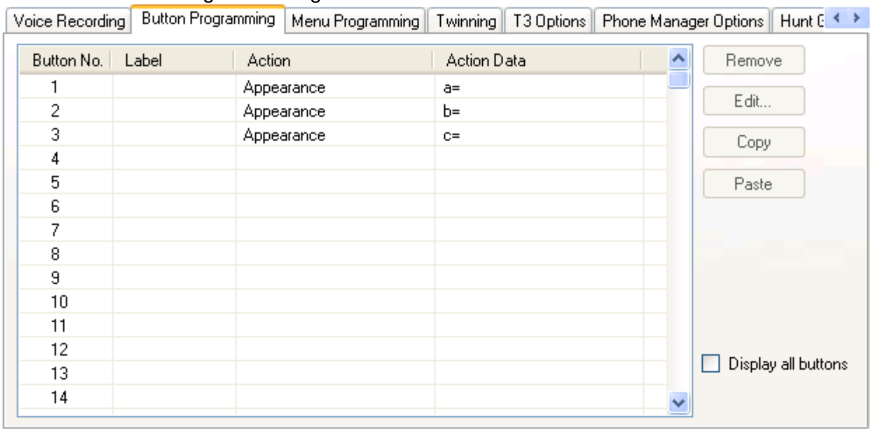
Programming Appearance Buttons Using Manager

If only button programming changes are required, the configuration changes can be merged back to the system without requiring a reboot.

1. Start Manager and load the current configuration from the system.

2. Locate and select the user for whom appearance buttons are required.

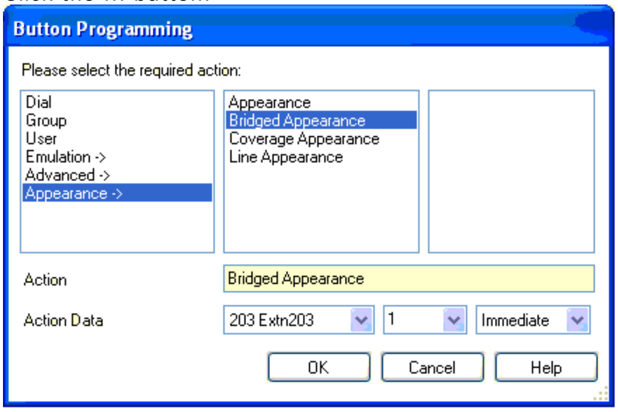
3. Select **Button Programming**.



- The number of buttons displayed is based on the phone associated with the user when the configuration was loaded from the system. This can be overridden by selecting **Display all buttons**.

4. For the required button, click the button number and then click **Edit**.

5. Click the ... button.



6. From the list of options that appears, click **Appearance**.

7. Select the type of appearance button required.

8. Use the **Action Data** drop-down fields to select the required settings. Click OK.

9. Repeat for any additional call appearance buttons required. Click OK.

10. Repeat for any other users requiring appearance buttons.

8.7.1 System Settings

System settings are applied to all users and calls. The system settings that affect appearance operation are found on the System | Telephony 160 tabs and are:

- Auto Hold: *Default = On. Software level = 3.0+.*
  Used for users with multiple appearance buttons. When on, if a user presses another appearance button during a call, their current call is placed on hold. When off, if a users presses another appearance button during a call, their current call is disconnected.

- Conferencing Tone: *Default = Entry & Exit Tones.*
  This settings controls how conference tones are used. It can be set to either *Entry & Exit Tones* or *Repeating Tone*. With *Entry & Exit Tones* a single tone is heard when a new party joins a conference and double-tone is heard when a party leaves the conference. With *Repeating Tone* a conference tone is heard every 10 seconds by all conference parties.

- Ring Delay: *Default = 5 seconds, Range = 0 to 98 seconds. Software level = 3.2+.*
  This setting is used when any of the user's programmed appearance buttons is set to Delayed ringing. Calls received on that button will initially only alert visually. Audible alerting will only occur after the ring delay has expired. This setting can be overridden by a ring delay set for an individual user (User | Telephony | Multi-line Options | Ring Delay 278).

- Visually Differentiate External Call: *Default = Off. Software level = 5.0+.*
  This setting is applied to the lamp flashing rate used for bridged appearance and call coverage appearance buttons on 1400, 1600 and 9600 Series phones and on SMB24, BM32 and DBM32 button modules. When selected, external calls alerting on those buttons will use a slow flash (200ms on/50ms off). If not selected or if the call is internal, normal flashing (500ms on/500ms off) is used.

8.7.2 User Settings

User settings are applied separately to each individual user. In addition to button programming, the following user settings are applicable to appearance button operation:

- Cannot be Intruded: *Default = On.*
  This feature controls whether other users can use their appearance buttons to join the users call. It applies when the user is the longest present internal party already within the call.
- Individual Coverage Time (secs): *Default = 10 seconds, Range 1 to 99999 seconds. Software level 3.0+.*
  This function sets how long the phone will ring at your extension before also alerting at any call coverage users. This time setting should not be equal to or greater than the No Answer Time applicable for the user.
- Ring Delay: *Default = Blank (Use system setting), Range = 0 (use system setting* 160*) to 98 seconds. Software level = 3.2+.*
  This setting is used when any of the user's programmed appearance buttons is set to Delayed ringing. Calls received on that button will initially only alert visually. Audible alerting will only occur after the ring delay has expired.
- Coverage Ring: *Default = Ring. Software level = 5.0+.*
  This field selects the type of ringing that should be used for calls alerting on any the user's call coverage and bridged appearance buttons. *Ring* selects normal ringing. *Abbreviated Ring* selects a single non-repeated ring. *No Ring* disables audible ringing. Note that each button's own ring settings (*Immediate*, *Delayed Ring* or *No Ring*) are still applied.
  - The ring used for a call alerting on a call coverage or bridged appearance button will vary according to whether the user is currently connected to a call or not.
    - If not currently on a call, the Coverage Ring setting is used.
    - If currently on a call, the quieter of the Coverage Ring and Attention Ring settings is used.

Attention Ring Setting	Coverage Ring Setting		
	Ring	Abbreviated	Off
Ring	Ring	Abbreviated	Off
Abbreviated	Abbreviated	Abbreviated	Off

- Attention Ring: *Default = Abbreviated Ring. Software level = 4.1+.*
  This field selects the type of ringing that should be used for calls alerting on appearance buttons when the user already has a connected call on one of their appearance buttons. *Ring* selects normal ringing. *Abbreviated Ring* selects a single ring. Note that each button's own ring settings (*Immediate*, *Delayed Ring* or *No Ring*) are still applied.
- Ringing Line Preference: *Default = On. Software level = 3.0+.*
  For users with multiple appearance buttons. When the user is free and has several calls alerting, ringing line preference assigns currently selected button status to the appearance button of the longest waiting call. Ringing line preference overrides idle line preference.
- Idle Line Preference: *Default = On. Software level = 3.0+.*
  For users with multiple appearance buttons. When the user is free and has no alerting calls, idle line preference assigns the currently selected button status to the first available appearance button.
- Delayed Ring Preference: *Default = Off. Software level = 4.0+.*
  This setting is used in conjunction with appearance buttons set to delayed or no ring. It sets whether ringing line preference should use or ignore the delayed ring settings applied to the user's appearance buttons.
  - When on, ringing line preference is only applied to alerting buttons on which the ring delay has expired.
  - When off, ringing line preference can be applied to an alerting button even if it has delayed ring applied. This is the same as pre-4.0 ringing line preference operation.
- Answer Pre-Select: *Default = Off. Software level = 4.0+.*
  Normally when a user has multiple alerting calls, only the details of the call on current selected button are shown. Pressing any of the alerting buttons will answer the call on that button, going off-hook will answer the current selected button. Enabling Answer Pre-Select allows the user to press any alerting button to make it the current selected button and displaying its call details without answering that call until the user either presses that button again or goes off-hook. Note that when both Answer Pre-Select and Ringing Line Preference are enabled, once current selected status is assigned to a button through ringing line preference it is not automatically moved to any other button.
- Reserve Last CA: *Default = Off. Software level = 4.0+.*
  Used for users with multiple call appearance buttons. When selected this option stops the user's last call appearance button from being used to receive incoming calls. This ensures that the user always has a call appearance button available to make an outgoing call and to initiate actions such as transfers and conferences. For pre-4.0 IP Office, this option is set by adding the RESERVE_LAST_CA= option on the User | Source Numbers 272 tab.
- Abbreviated Ring:
  This option has been replaced by the Attention Ring setting above.

8.7.3 Line Appearance ID Numbers

Line appearances are supported for analog, E1 PRI, T1, T1 PRI, and BRI PSTN trunks. They are not supported for E1R2, QSIG and IP trunks.

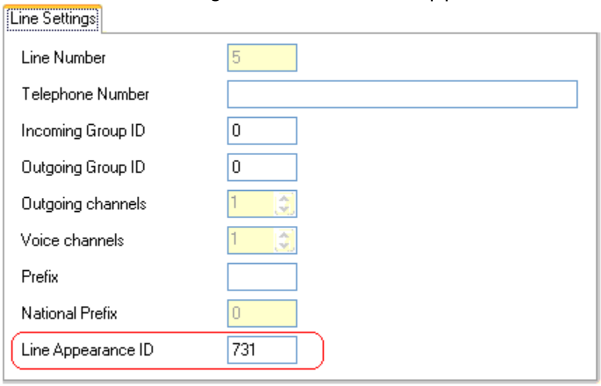
Note that setting and changing line settings including line appearance ID numbers requires the system to be rebooted.

Automatic Renumbering

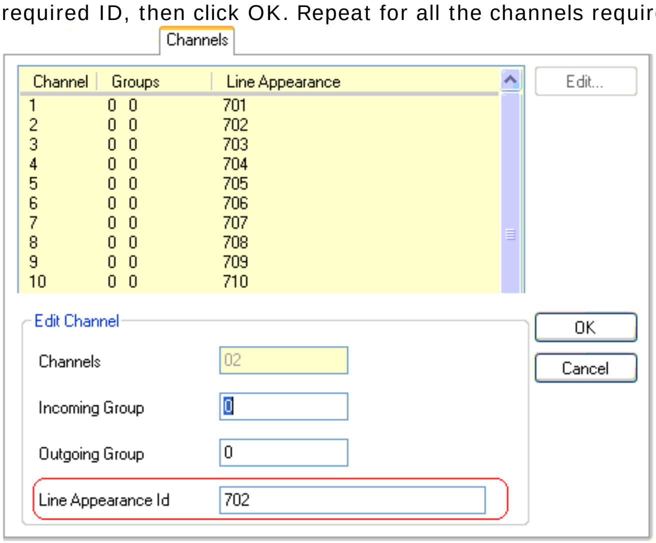
1. Select Tools | Line Renumber.
2. Select the starting number required for line numbering and click OK.
3. All lines that support Line Appearance ID will be numbered in sequence.

Manual Renumbering

1. Start Manager and load the current configuration from the system.
2. Select  Line.
3. Select the line required. The tab through which line appearance ID numbers are set will vary depending on the type of line. A couple of examples are shown below.
   - Analog Line
     On the Line Settings tab select Line Appearance ID and enter the ID required.



   - Basic/Primary Rate Trunks
     On the Channels tab select the individual channel and click Edit. Select Line Appearance ID and enter the required ID, then click OK. Repeat for all the channels required.



4. Click OK and repeat for any other lines.

8.7.4 Outgoing Line Programming

Assigning line ID numbers to lines and associating line appearance buttons to those lines is sufficient for answering incoming calls on those lines. However, to use line appearance buttons for outgoing calls may require further programming.

Short Codes and Outgoing Line Appearance Calls

Once a line has been seized using a line appearance button, short code matching is still applied to the number dialed. That can include user, system and ARS or LCR short codes.

- The short codes matching must resolve to an off-switch number suitable to be passed direct to the line.
- The final short code applied must specify a 'dial' feature. This allows call barring of specific matching numbers to be applied using short codes set to features such as 'Busy'.

8.8 Upgrading from Pre-3.0 Systems

Most of the aspects of appearance functions have no effect on systems being upgraded from versions earlier than 3.0. However the following existing configuration settings are affected:

- Call Appearance Buttons
  Call appearance buttons should start with button 1 and form a single block. On systems being upgraded where the call appearance buttons have been programmed differently, those buttons will be lost following the upgrade to 3.0.
- Call Coverage
  Call coverage support was originally added in IP Office 1.3. Call coverage operation in IP Office 3.0 is a total replacement and uses different methods of setup and operation. If upgrading from a version earlier than 3.0, all existing call coverage settings will be lost. Therefore it is necessary to make manual notes of the covered and covering users before upgrading to IP Office 3.0 or higher.
- Outgoing Call Routing
  If it is a requirement to use line appearance button for outgoing calls, then it will be necessary to add incoming line prefixes and to setup secondary dial tone short codes based on the same prefixes. See Outgoing Line Programming 720.

8.8.1 Call Handling Changes

People who have used phones on systems prior to IP Office 3.0 will find a number of changes to the way the phones operate.

- Selected Button
  By default one appearance button is indicated as the current selected button. This is done either by the adjacent red LED being on or the button display label being underlined.
  - The current selected button represents either the appearance button of the current connected call or the appearance button that will be used if the user goes off hook (ie. lifts the handset or presses the Speaker key).
- Auto Hold
  The default operation of this feature depends on the phone system's locale:
  - In the US, the default setting for Auto Hold is *off*. With a current call already connected on one appearance button, pressing another appearance key will disconnect the current call.
  - Outside the US, the default setting for Auto Hold is on. With a current call already connected on one appearance button, pressing another appearance key will hold the current call.
  - Note that for IP Office 4.0+, on all systems regardless of locale, the default for Auto Hold is *off*.
- Call Waiting
  Call waiting features are not applied to non-hunt group calls. Hunt group call waiting is applied, subject to the normal conditions for hunt group call waiting, but with the call waiting tone replaced by a call appearance button alert.
- Busy Status
  For a user with call appearance buttons, you are busy when:
  - For calls direct to your extension number, you are busy only when you have no further call appearance buttons available on which to present the call.
  - For calls to any hunt group of which you are a member, you are busy to further group calls once you have a call connected unless hunt group call waiting is being used. However you may still receive additional calls dialed or forwarded direct to your extension number if you have any free call appearance buttons.
  - In either case above, even when busy, you may be alerted on other types of appearance button, for example line appearance and call coverage.
  - Busy on held can be used to return busy to further calls when you have an existing call on hold, however this is not recommended in conjunction with appearance buttons.
- Retrieving Held Calls
  Pressing Hold will put the current connected call on hold. The appearance button will indicate the held call with, depending on the phone type, a flashing green lamp or a icon. However pressing Hold again will not retrieve the held call.
  - To retrieve a held call, press the appearance button against which the call was held.
  - An intermittent green flashing lamp or an icon indicate a call held by another user. These can also be retrieved by pressing that appearance button.
- Transferring Calls
  To transfer calls, the Transfer function of the phone or Phone Manager application should be used.
- Joining Another Call
  Appearance buttons can indicate a call taking place elsewhere. You may be able to join that call by pressing the appearance button.

- Permission to join a call is controlled by the intrusion settings of the party who has been in the call the longest, not by your intrusion settings. See Joining Other Calls 710 for full details.
- The joined call is a conference call and is subject to the system's conference controls. If you press **Hold**, it is your connection to the conference that is held, it does not affect the other parties in the call.

- **Unparking Calls**
Parked calls can be indicated by buttons programmed to the Park function and a particular park slot number. These can be calls parked by the phone user or parked by other users. If the phone is idle, the parked call can be unparked by pressing the park button. If another call is connected on an appearance button, that call must be put on hold or ended before the parked call can be unparked.

- **Unprogrammed/Inaccessible Keys**
Buttons to which no function has been set or which the function is currently not useable, will produce a short burst of tone or ringing when pressed. If the button has an associated status lamp, it will also flash briefly.

- **Call Count**
Avaya phones with twin LED buttons display a call count. This appears as a digit count, for example 01 on the display. It increments for all calls that alert at a user's extension including missed and unanswered calls.
  - Sequential calls from the same number do not increment the count more than once.
  - Calls alerting on bridged, line and call coverage buttons are not included in the call count and history.
  - The count is reset whenever the system is restarted.
  - On Avaya 4400, 4600 and 6400 Series phones with a MENU key, details of the call count can be accessed and entries called or deleted. Press MENU | MENU and select **Hist**.

- **Call Log**
On 2410, 2420, 4610, 4620, 5410, 5420, 5610 and 5620, the call count is replaced by a **Log** function instead of the call count and history detailed above.

- **Self-Administration of Appearance Keys**
Previously users with a programmable button set to **Self-Administer** (**Admin**) could program their own call appearance buttons. In IP Office 3.0 this is no longer possible. In addition users cannot overwrite keys already programmed to an appearance function.

8.8.2 Twin LED Button Users

These notes are for users of Avaya phones which have twin LED lamps next to the programmable buttons. That is most 4400 Series, 6400 Series and older style 4600 Series phones.

- Red LED On
  This is called the currently selected button. It indicates either the button that will be used if you go off-hook (pickup the handset or press the Spkr button) or the appearance button of the call to which you are currently connected. When your phone is idle your first call appearance button will show the red LED on.
- Green LED
  The green LED is used as follows on appearance buttons:
  - On / Off: The button is in use/idle. This can include being used by another user, for example a line appearance button of a shared line.
  - Flashing: A call is alerting you.
  - Fast Flash: The call is on hold.
  - Intermittent Flash: The call has been put on hold by another user (on hold elsewhere).
- Call Handling
  Having appearance keys means:
  - You can make and answer calls by pressing the appearance buttons.
  - Auto Hold: When connected to a call on one appearance button, pressing another appearance button will place the first call on hold and make/answer a call on the button pressed. For US systems this changes to disconnecting the first call.
  - Transferring Calls: To transfer a call, press Trnsfr, call the transfer destination and then press Trnsfr again.
  - Retrieving Held Calls: You cannot retrieve a held call by pressing Hold again. To retrieve a call from hold, press the appearance key associated with that call. Held calls are indicated by a fast flashing green lamp.
- Busy Status
  Call appearance buttons allow you to make, receive and switch between multiple calls. This changes when the system returns busy to additional calls
  - Hunt Group Calls: You are seen as busy to further hunt group calls once you have any appearance button in use.
  - Personal Calls: You are seen as busy to further calls directed to your extension number once all your call appearance buttons are in use.
  - Forward on busy, if set, is only used when all your call appearance buttons are in use. Even when busy, additional calls can alert you about other calls on call coverage, bridged appearance and line appearance button if programmed for you..
  - Busy on Held can be used but we strongly recommend it is avoided for appearance key users.

- Joining a Call
  When an appearance button indicates a call in progress elsewhere. You may be able to use that button to join the call. Whether you are allowed to join the call depends on the Cannot Be Intruded setting of the user who has been in the call the longest.
- Call waiting settings are ignored for direct call to appearance button users. Hunt group call waiting can be used if setup.
- Unprogrammed / Inaccessible Keys
  These keys will return a short tone and flash.
- Self-Administration: Users with access to this function can no longer program their own appearance keys or replace the programming of existing appearance buttons.
- Hot Desking / Logging On: If you move to a phone that doesn't have suitable programmable keys to support appearance buttons, you are treated as a normal PBX mode user while on that phone.

Twin LED Button Users Lamp States

The table below summarizes the meaning of the twin LED lamp states.

- Selected refers to the button indicating either the appearance button of the currently connected call or, if idle, the appearance button that would be used if the user goes off hook.

Call Appearance Buttons	IP Office 2.1/3.0DT	IP Office 3.0+
- Idle.	All off.	All off.
- Idle: Selected.	–	Red on.
- Alerting.	Green flash.	Green flash.
- Alerting: Selected.	–	Red on, Green flash.
- In Use: Here.	Green on.	Red on, Green on.
- In Use: Elsewhere.	–	Green on.
- Held: Here.	Red on.	Green fast flash.
- Held: Elsewhere.	–	Green intermittent flash.
Other Buttons	**IP Office 2.1/3.0DT**	**IP Office 3.0+**
- Parked: Here	Green flash.	Green flash.
- Parked: Elsewhere	Red flash.	Red flash.
- User: Ringing	Red flash.	Green flash.
- User: In use.	Red on.	Green on.
- Group: Ringing.	Green Flash.	Green flash.
- Group: Queued.	Red flash.	Red flash.
- Other toggling features: On.	Green on.	Green on.
- Other toggling features: Off.	All off.	All off.
- Non-toggling features	All off.	All off.

The flash rate on button lamps is sometimes used to indicate different status. For example a fast flash is used for calls you put on hold and a intermittent flash for calls put on hold by someone else.

- Steady Flash: **********---------- *(0.5s on/0.5 s off/...)*
  Used mainly to indicate alerting calls.
- Fast Flash: *-*-*-*-*-*-*-*-*-*- *(50ms on/50ms off/...)*
  Used mainly to indicate calls put on hold by you. This flash mode is also called 'Flutter'.
- Intermittent Flash: *----*----*----*---- (50ms on/200ms off/...)
  Used mainly to indicate calls on hold elsewhere, ie. put on hold by another user. This flash mode is also called 'Inverse Wink'.

Chapter 9.
Telephone Features

9. Telephone Features

This section provides descriptions of the following telephony functions provided by the system.

9.1 Date and Time

The control unit contains a battery backed clock which is used to maintain system time during normal operation and when mains power is removed.

For files stored on memory cards the system uses the UTC time. For other activities such as call logs, SMDR records, time display on phones; the local time (UTC + any offsets) is used.

The time can be set in a number of ways list below. The method used is set through the **Time Server IP Address** or **Time Setting Config Source** settings on the System | System 143 form.

- **Simple Network Time Protocol (SNTP - RFC4330)** *(IP Office 6.1+)*
  SNTP can be used to make time requests to a list of NTP servers. The response is just a UTC time value, therefore the system has to be configured with the required offset for the local time value and also optional daylight savings values. The time request is sent when the system is started and every hour afterwards.

- **Voicemail Pro/Manager (TIME RFC868)**
  Both the Voicemail Pro service and the Manager application can act as RF868 time servers, obtaining the time for the PC on which they are running. Use of other RFC868 server sources is not supported. In response to a request from a system, they will provide both the UTC time and local PC time. The time request is sent when the system is started and every 8 hours afterwards.

  - If you are running Manager when the Voicemail Server starts, then Voicemail does not start as a time server. It is therefore recommended that you have no copy of Manager running when you start or restart the Voicemail Server.

  - By default a broadcast address is used. A specific address for the time server that should be used can be set if required.

  - When using a time server located in a different time zone from the system, there are two mechanisms for applying an offset to the time. If Manager is acting as the time server, the time offset can be specified through the **Time Offset** of the BOOTP 140 entry for the system. Alternatively, the offset can specified in the system configuration using **Time Offset** 143 (System | System) 143.

- **Manual Time Control**
  The use of time requests using either of the above methods can be disabled. In that case the time and date used by the system is set manually using a system phone. See Manually Setting the System Time below.

  - **Pre-IP Office 6.1:** The use of automatic updates can be disabled by setting the **System | System | Time Server IP Address** 143 to *0.0.0.1*. If the system is still configured for automatic time updates, any manually set time and date will be overridden on the next automatic update.

  - **IP Office 6.1+:** Automatic time updates can be disabled by setting the **System | System | Time Setting Config Source** 143 to *None*. If automatic time updates are being used, the manual controls below are overridden and only allow display of the time and date information received from the server.

Manually Setting the System Time

For systems without access to a time server, a number of methods to manually set the system time exist. These are available in the IP Office 4.1+. In both cases the user's login code, if set, is used to restrict access to the time and date settings.

1400, 1600 and 9600 Phones

Phones in these series (excluding the 1403 and 1603) can set the system time and date when the extension user is configured with System Phone Rights 765. The user is able to access menu options to set the time, date and time offsets by selecting **Features | Phone User | System Administration**.

Other Phones

The following method is only supported on these phones: 2410, 2420, 4412, 4424, 4612, 4624, 4610, 4620, 4621, 5410, 5420, 5610, 5620, 5621, 6412, 6424. The phone needs to be configured with a **Self Admin 2** button.

Enabling a User Button for Time and Date

1. Select the user who can adjust the system time and date.
   - If necessary create a new user who can be used to log in and set the time and date before logging out.
2. On the **User | Telephony** tab, ensure that the user is configured as a System Phone 765 user.
3. On the **User | Button Programming** tab select and double-click on an available button.
4. Select **Emulation > Self-Administer** as the action.
5. In the Action Data field enter *2*.
6. Enter an appropriate label for the button.
7. Adding another button set to the **Emulation > Time of Day** action is recommended as this will allow the user to view and confirm the systems current time.
8. Send the updated configuration to the system.

Using the Programmable Button

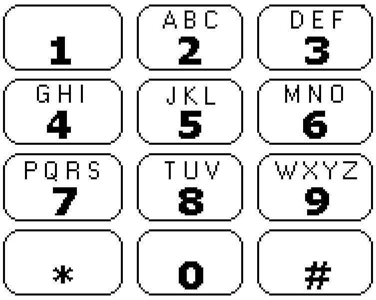
1. Press the programmed button. If the user has a **Login Code** set it is requested.
2. The options **Date** and **Time** are displayed.
3. To set the date press **Date**.
   1. The current system date is displayed.
   2. Enter the new date, using two digits for day and month. Use the * or # keys to insert the / separators. The format for date enter matches the locale of the system.
   3. During entry the key labeled <<< can be used to backspace.
   4. When the full date has been entered as required, press **Next**.
   5. Press **Done**.
   6. The phone will return to idle.
4. To set the time press **Time**.
   1. The current system time is displayed.
   2. Enter the new time using 24-hour clock format. Use the * or # keys to insert the : separator.
   3. When the full time has been entered as required, press **Done**.
   4. The phone will return to idle.

9.2 User Directory Access

On Avaya phones with suitable displays, the various system directories can be accessed. The user can then select which directory they want to search and then search that directory by dialing part of the required name and selecting from the displayed matches.

- On phones with a Contacts button, that button can be used to access the system directories plus the users own centralized personal directory. The user is able to select from directories of *All*, *Personal*, *External*, *User* or *Groups*.
- On other phones with a programmable button set to the Directory 615 (Dir) function. On some the user is then able to select from directories INDeX (internal extensions), Group (Hunt Groups) or Extrn (numbers in the system directory). On phones with a Contacts button it will access the same directories as the Contacts button detailed above.
- On phones with a Menu button, that button can be used to access the same directories as a Directory programmable button. Press Menu and select Dir. Alternatively, press Menu twice, then press ▶ and then select Dir.
- For many programmable button functions that require entry of a number when pressed, a Dir option may be displayed on the phone display to allow number selection using the directories. In this case the directories available may be limited to those supported by the button function. See Interactive Button Menus 515.
- When used to make internal calls, the name matches are based on the User Names and Full Names programmed into the system. If a user has a Full Name programmed then that takes precedence over their User Name. When used to make external calls, the name matches are based on entries in the system directory.

Name dialing functions on the system assume that the phone is using the standard ITU keypad as follows:



- Dialing Spaces
  For names that include spaces, the method of indicating a space has changed in IP Office 5.
  - Pre-IP Office 5
    To enter a name with a space, nothing is dialed for the space. For example "John S..." is dialed as 56467.
  - IP Office 5+
    To enter a name with a space, the 0 key is used for the space. For example "John S..." is dialed as 564607.

Selecting Dial By Name Mode

1. Double-click System.
2. Select the Telephony tab.
3. The Dial By Name checkbox operates as follows.
   - Dial By Name Selected
     Using the letter keys, start dialing the name that you want. For example, for names starting with John, dial 5646. The display will show the first match to the letters entered so far. Either enter further letters or use the ◀ and ▶ keys to scroll through the other matches found.
   - Dial By Name Not Selected *(This mode is no longer support for IP Office 5+)*
     Press the dial pad button that matches the first letter of the name you want. For example, to select L, press the 5 key three times.

9.3 Mobile Call Control

Mobile call control is only supported on IP500/IP500v2 system digital trunks including SIP trunks. It allows a user receiving a call on their twinned device to access system dial tone and then perform dialing action including making calls and activating short codes.

After answering a twinned call, the Mobile Call Control user can dial ** (within 1 second of each other) to place that call on hold and instead get dial tone from the system. Any dialing is now interpreted as if the user is logged into an analog extension on the system using their user settings. That also include user BLF status indication.

- Licenses are required in the system configuration for all the users that will be configured for any of the mobility features including Mobile Call Control and one-X Mobile Client usage. The same licenses also allows the user to use other mobility features such as mobile twinning, mobile call control and one-X Mobile if required.
  - Pre-IP Office 6.0: **Mobility Features** licenses are used.
  - IP Office 6.0: Users licensed for a Profile 265 of *Mobile Worker* or *Power User* are able to use mobility features.
  - If a Mobile Call Control user remote hot desks to another system within an SCN, they take their licensed status with them rather than consuming or requiring a license on the remote system.
- **Trunk Restrictions**
  This feature is only supported on the IP500/IP500v2 systems on the following trunk types:
  - IP500 Universal PRI (Not on T1 robbed-bit and E1R2 channels set to Loop Start emulation).
  - IP500 BRI.
  - IP500 SIP (RFC2833).
- Routing via trunks that do not support clearing supervision (disconnect detection) should not be used.
- DTMF detection is applied to twinned calls to a user configured for this feature. This will have the following effects:
  - DTMF dialing is muted though short chirps may be heard at the start of any DTMF dialing.
  - DTMF dialed by the user will not be passed through to other connected equipment such as IVR or Voicemail.

- **WARNING**
  This feature allows external callers to use features on your phone system and to make calls from the phone system for which you may be charged. The only security available to the system is to check whether the incoming caller ID matches a configured users' **Twinned Mobile Number** setting. The system cannot prevent use of these features by caller's who present a false caller ID that matching that of a user configured for access to this feature.

Mobile Call Control Features and FNE Codes

Mobile call control uses a short code set to invoke an FNE code. See one-X Mobile Client 734 for the full list of the codes, the code relevant to mobile call control are summarized below.

- **FNE 31 = Mobile Call Control** *(IP Office 4.2 and higher)*
  This code allows a user called or calling the system to invoke mobile call control and to then handle and make calls as if they were at their system extension.
- **FNE 32 = Mobile Direct Access** *(IP Office 4.2 Q3 2009 maintenance release and higher)*
  Mobile direct access 732 FNE32 immediately redials on switch the DDI digits received with the call rather than returning dial tone and waiting for DTMF digits as with FNE31 .
- **FNE 33 = Mobile Callback** *(IP Office 6.0 and higher)*
  Mobile callback 733 allows the user to call the system and then hang up. The system will then make a call to the user's CLI and when answered, provide them with dial tone from the system to make calls.

Using Mobile Call Control

In addition to using ** to access mobile call control, the user has access to the following additional controls:

- **Clearing a Call:** ***52**
  It may be necessary to clear a connected call, for example after attempting a transfer and hearing voicemail or ringing instead. To do this dial ** for dial tone and then ***52** (this is a default system short code 437 and can be changed if required).
- **Return to Dial Tone: ##**
  Return to dial tone after getting busy, number unobtainable or short code confirmation tones from the system.

Enabling Outgoing Mobile Call Control

1. License Mobile Features
   Enter the licenses for Mobile Twinning and confirm that they are valid by merging and retrieving the configuration.
2. Configure the user for Mobile Twinning and Mobile Call Control
   On the User | Mobility 293 tab do the following:
   - Enable Mobility Features for the user.
   - Set the Twinned Mobile Number for the user's twinned calls destination.
     - Digits are matched from right to left.
     - The match must be at least 6 digits. If either the CLI or the Mobile Twinned Number is less than 6 digits no match will occur.
     - Matching is done for up to 10 digits. Further digits are ignored. If either the CLI or Mobile Twinned Number is less than 10 digits, matching stops at that shorter length.
     - If multiple matches occur the first user in the configuration is used. Manager will warn against configuration where such a conflict may exist.
   - Select Can do Mobile Call Control.
3. On systems with some unsupported trunk types, further changes such as Outgoing Group ID, system shorts codes and ARS may be necessary to ensure that calls to the mobile twinned numbers are only routed via the

Incoming Mobile Call Control

The system can be configured to allow Mobile Call Control users to use this function when making an incoming call to the system. This requires the user to make the incoming call from the same CLI as their Mobile Twinning Number (even if they do not actually use Mobile Twinning).

The call will be rejected:

- If the caller ID is blank or withheld.
- If the caller ID does not match a Twinned Mobile Number of a user with Can do Mobile Call Control enabled.
- If the call is received on a trunk type that does not support Mobile Call Control.

Enabling Incoming Mobile Call Control

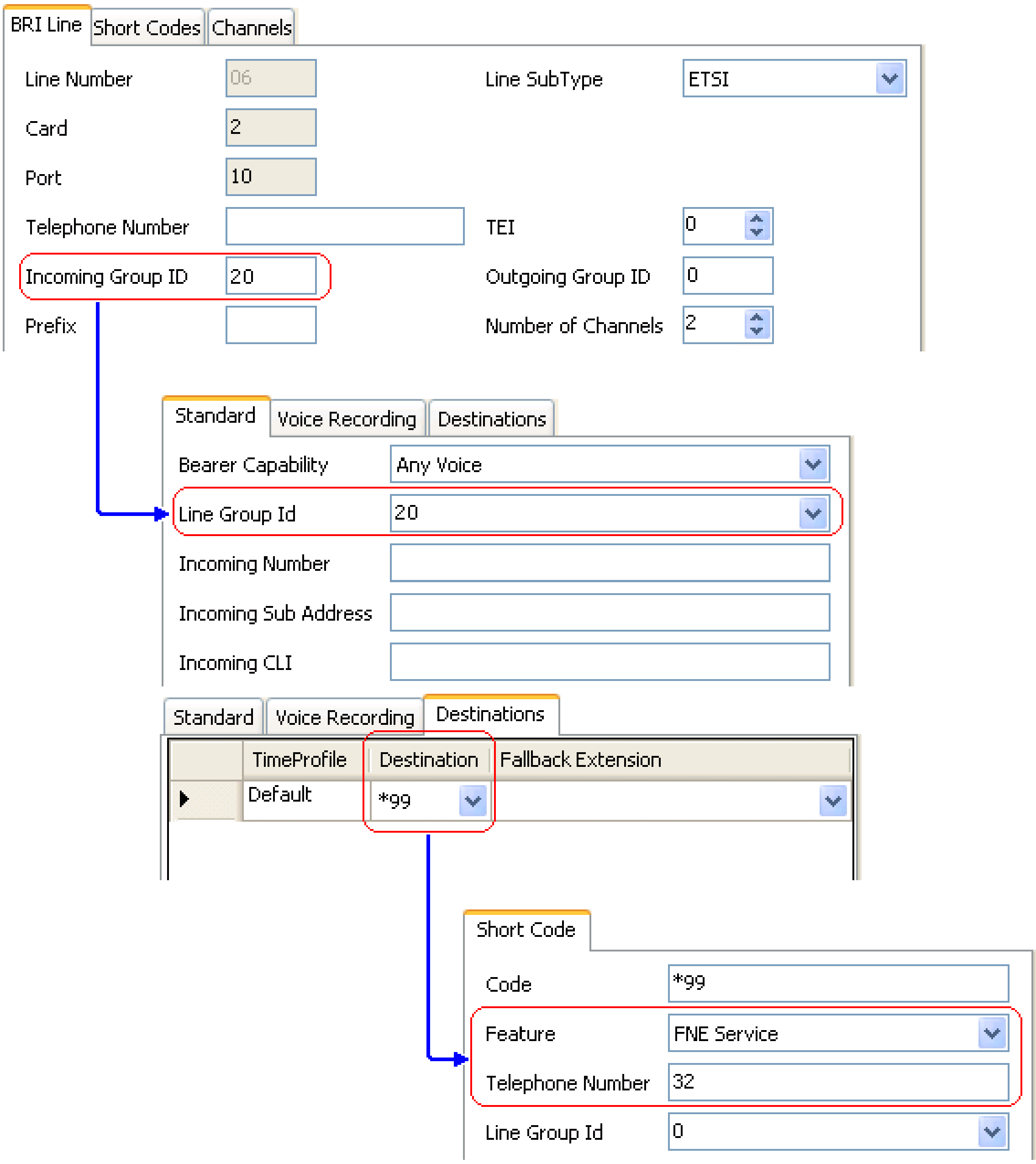
1. License Mobile Features
   Enter the licenses for Mobile Twinning and confirm that they are valid by merging and retrieving the configuration.
2. Configure the user for incoming Mobile Call Control
   On the User | Mobility 293 tab do the following:
   - Enable Mobility Features for the user.
   - Set the Twinned Mobile Number to match the CLI of the device from which the user will be making calls.
   - Select Can do Mobile Call Control.
3. Add a FNE Short Code
   In the system short codes section of the configuration add a short code similar to the following. Key points are the use of the *FNE Service* feature and the Telephone Number value *31*.
   - Short Code: *99
   - Feature: FNE Service
   - Telephone Number: 31
4. Add an Incoming Call Route for the user
   Create an incoming call route that matches the user's CLI and with the FNE short code created above as its destination.
5. On systems with some unsupported trunk types, further changes such as Incoming Group ID changes may be necessary to ensure that only calls received on trunks that support Mobile Call Control are routed to this short code.

9.3.1 Mobile Direct Access (MDA)

IP Office 4.2 Q3 2009 maintenance release and higher. For a Mobile Call Control or one-X Mobile client user, FNE32 immediately redials on switch the DDI digits received with the call rather than returning dial tone and waiting for DTMF digits as with FNE31. This is called Mobile Direct Access (MDA).

MDA requires the user's external telephony provider to provide a direct trunk with DDI to the system (ie. an ISDN or SIP trunk). By assigning a specific incoming line group ID to the trunk, an incoming call route can be created for the same line group ID with blanks incoming number and incoming CLI fields. The destination is a short code set to FNE32.

User validation is performed using the CLI in the same way as for normal Mobile Call Control. In addition the call will be rejected no DDI digits are provided. Once connected the user can use the other Mobile Call Control features such as **.



9.3.2 Mobile Callback

Mobile callback 733 allows the user to call the system and then hang up. The system will then make a call to the user's CLI and when answered, provide them with dial tone from the system to make calls.

Mobile callback is subject to all the normal trunk type and user licensing restrictions of mobile call control 730. In addition the user must have the **Mobile Callback (User | Mobility)** setting enabled in the system configuration.

- When the user makes a call using a DDI that is routed to an FNE33 short code, the system will not connect (answer) the call but will provide ringing while it waits for the user to hang up (after 30 seconds the system will disconnect the call).
  - The system will reject the call if the CLI does not match a user configured for Mobile Callback or does not meet any of the other requirements for mobile call control.
  - The system will reject calls using FNE33 if the user already has a mobile twinning or mobile call control call connected or in the process of being connected. This includes a mobile callback call in the process of being made from the system to the user.
- If the CLI matches a user configured for mobile callback and they hang up within the 30 seconds, the system will within 5 seconds initiate a callback to that user's CLI.
  - If the call is answered after the user's **Mobile Answer Guard** time and within the user's **No Answer Time**, the user will hear dial tone from the system and can begin dialling as if at their system extension.
  - If the call is not answered within the conditions above it is cleared and is not reattempted.

9.4 one-X Mobile Client

IP500 and IP500v2 systems running IP Office 4.2+: The system can support the Avaya one-X Mobile Client on Windows and Symbian mobile cell phones. This uses DID numbers (one for each feature required) to activate and use various one-X Mobile features supported by the system.

The system is configured to route incoming calls on those DID numbers to short codes for each feature. The caller's incoming CLI is then checked for a match against the mobile twinning and forwarding numbers of those users configured for one-X Mobile operation. If a match is found access to the one-X feature is allowed using the identity of that user as if they were now dialing from an extension on the system.

- WARNING
  This feature allows external callers to use features on your phone system and to make calls from the phone system for which you may be charged. The only security available to the system is to check whether the incoming caller ID matches a configured users' Twinned Mobile Number setting. The system cannot prevent use of these features by caller's who present a false caller ID that matching that of a user configured for this feature.

Example

Stage	Events
Setup the User's Mobile Cell Phone for one-X Mobile	1. The one-X Mobile client application has been loaded onto a user's mobile cell phone. 2. A configuration file for the system one-X functions is also loaded. Within this the DDI numbers allocated for one-X mobile features are split into two parts; a shared DDI prefix for system and then a DDI suffix for each require one-X feature. Whenever a one-X feature is selected by the user, the prefix and the appropriate suffix for the feature are combined into a number dialled by the users mobile cell phone.
User selects a one-X Mobile feature on their phone	1. Within the one-X Mobile client application on their phone, the user selects the 'dial tone' feature. 2. The DDI prefix and the DDI suffix in the phone's one-X Mobile configuration file are combined to dial the appropriate DDI number. For this example 555 776 9900.
The DDI call is received by the system	1. The call is received by the system along with the number of DDI digits that the line provider passes through. For this example the four digits 9900. 2. On the system an Incoming Call Route has been added with the Incoming Number field set to *99XX*. This will match the 9900 digits just received. 3. The Incoming Call Route is set with the Destination of *777"#"*. The *"#"* will be replaced by the *XX* part of the DDI number matched above. For this example, 00 will set the destination as 77700.
Check the CLI received with the call for a one-X Mobile User match	1. The system has a system short code *777XX*. Its settings are *777XX / N / FNE*. 2. The short code uses the FNE feature. This causes the system to check that it has received an incoming CLI number with the call. The last 6 to 10 digits of that number are checked for a match against the Twinned Mobile Number or Forward Number of those users on the system who have been configured as Can have one-X Mobile Client. 3. Once a match occurs the call proceeds using the identity of the user matched.
The appropriate one-X Mobile feature is provided to the user.	1. The short code takes *777XX* and then passes the value *N* to the FNE feature. Therefore in our example 77700 will pass the value 00. 2. The value 00 matches the FNE feature Dial Tone (also called Idle Appearance Select). This feature causes the system to return system dial tone to the user so that they can now dial the number they require as a call from the system.

Restrictions

- Supported Phones and Phone Client Software
  Avaya one-X Mobile Client on mobile cell phones that run Windows Mobile V5/V6 and Symbian Single-Mode (V4.0). For the regularly updated list refer to the Avaya one-X Mobile section of the Avaya support website (support.avaya.com). This same site is also the location of the client software files that need to be installed on the phone and how to transfer those files to the phone.
  - Each phone requires a settings file. That file is used to configure the features the user can access and the DID number that the phone should dial in order to access the feature.
- Trunk Restrictions
  This feature is only supported on the IP500/IP500v2 systems on the following trunk types:
    - IP500 Universal PRI (Not on T1 robbed-bit and E1R2 channels set to Loop Start emulation).
    - IP500 BRI.
    - IP500 SIP (RFC2833).
  - Routing via trunks that do not support clearing supervision (disconnect detection) should not be used.
- Users
  - one-X Mobile is a system licensed feature using a Mobility Features license for each user.
  - It is only useable by users who have Can use one-X Mobile Client and Mobility Features enabled in the system configuration.
  - The CLI provided on calls from their mobile cell phone to the system must match the Twinned Mobile Number or Forwarding number set on the system. At least 6 digits are required.
      - For user's setup for one-X Mobile Client, changes to their Mobile Twinning status made through the system configuration or using a Twinning button are not reflected in the status of the Extension to Cellular icon on their mobile client. However, changes to the Extension to Cellular status made from the mobile client are reflected by the Mobile Twinning field in the system configuration. Therefore, for one-X Mobile Client users, it is recommended that they control their Mobile Twinning status through the one-X Mobile Client rather than through a Twinning button.

Additional Documentation

The following guides include details of how to install and configure the software on the phones. These documents are available from the one-X Mobile client section of the Avaya support website (support.avaya.com).

Document	Doc Code
Avaya one-X Mobile Edition Device Compatibility List	327824
Avaya one-X Mobile Edition for S60 User Documentation	16-601288
Avaya one-X Mobile Edition for Windows Mobile 5 Pocket PC and Treo Edition Install, Admin and User Guide R4.0	16-601613
Avaya one-X Mobile Edition for Windows Mobile 5 Smartphone Edition User Guide	16-602323

Feature Name Extension (FNE) Features

The one-X Mobile client accesses system features using DID numbers routed to system short codes set to the FNE feature. The number set in the short code indicates the particular FNE feature required.

The table below lists the FNE features supported by system. The settings file tag is the string used in the phone software settings file to associate a feature with a DID number.

Value	Function (Setting File Tag)
00	Dial Tone (**IDLE_APPEARANCE_SELECT**) - Provide system dial tone to the user. Similar to Mobile Call Control 730 .
01	Active Appearance Call Select (**ACTIVE_APPEARANCE_SELECT**) - Attempt a call acquire (steal) on the user's primary extension. If there is no active or ringing call to acquire, give system dial tone.
02	Automatic Callback (**AUTO_CALL_BACK_TOGGLE**) - Set a ringback when free when calling a system extension number that does not answer or returns busy.
04	Call Forwarding All Calls (**CALL_FORWARDING_ALL_ACTIVATION**) - Switch forward unconditional on.
05	Call Forwarding Busy (**CALL_FORWARDING_BUSY_NO_ANSWER_ACTIVATION**) - Switch forward on busy and forward on no answer on.
06	Call Forwarding Deactivation (**CALL_FORWARDING_DISABLE**) - Switch all forwarding off.
07	Call Park (**CALL_PARK**) - Park call on the system using the user's extension ID.
08	Call Unpark (**CALL_UNPARK**) - Unpark a call parked using the user's extension ID.
09	Call Pickup Group (**CALL_PICKUP_GROUP**) - Call Pickup Any 444
10	Call Pickup Directed (**CALL_PICKUP_DIRECTED**) - Directed Call Pickup using the entered user or hunt group number.
12	Calling Party Number Block (**CALLING_PARTY_NUMBER_BLOCK**) - Make a call from the system with CLI withheld.
13	Calling Party Number Unblock (**CALLING_PARTY_NUMBER_UNBLOCK**) - Make a call from the system with CLI allowed.
14	Conference on Answer (**CONFERENCE_ON_ANSWER**) - Make an additional call and add them to a conference.
15	Drop (**DROP_LAST_ADDED_PARTY**) - Drop last party added to a conference.
16	Exclusion (**EXCLUSION**) - Make the existing call private 757. Repeat to remove privacy from existing call.
17	Held Appearance Select (**HELD_APPEARANCE_SELECT**) - Pickup a held call on the user's primary extension.
18	Dial Tone (**IDLE_APPEARANCE_SELECT**) - Same as 00 - Dial Tone.
19	Off PBX Call Enable (**OFF_PBX_ENABLE**) - Switch the user's Mobile Twinning on.
20	Off PBX Call Disable (**OFF_PBX_DISABLE**) - Switch the user's Mobile Twinning off.
24	Send All Calls Enable (**SEND_ALL_CALLS_ENABLE**) - Do Not Disturb On.
25	Send All Calls Disable (**SEND_ALL_CALLS_DISABLE**) - Do Not Disturb Off.
26	Transfer On Hang Up (**TRANSFER_ON_HANGUP**) - Unsupervised transfer.
27	Transfer to Voicemail (**TRANSFER_TO_COVERAGE**) - Transfer the held system call to the user's voicemail mailbox.
31	Used for Mobile Call Control 730.
32	Used for Mobile Direct Access 732.
33	Used for Mobility Callback with system mobile call control.

Example Configuration Overview

In this example, two DID numbers are going to be used to allowed one-X Mobile Users to access the FNE functions 00, 01 and 26. This will allows those users respectively to obtain system dial tone, to acquire a call alerting on their normal extension and to transfer a call.

The DID numbers requested were 555 76769900, 555 76769901 and 555 76769926 with 4 digits being passed by the provided to the system, ie. 9900, 9901 and 9926.

- This example is made much simpler by using both a continuous DID range and one where the last two digits of the DID numbers match the system FNE short code features required. This allows implementation using an single short code and a single incoming call route.

System Configuration

1. **IP Office one-X Mobile Client Licensing**
   Mobility Features licenses are required in the system configuration for all the users that will be configured for any of the system mobility features including one-X Mobile Client usage. The same single license also allows the user to use mobile twinning and mobile call control if required. Check that the configuration contain valid **Mobility Features** licenses.

2. **User Configuration**
   For each user, the following must be set on the User | Mobility 293 tab.

   - Select **Mobility Features**.
   - Select **Can have one-X Mobile Client**.
   - Set their **Twinned Mobile Number** to match the CLI of calls from the device on which they will be using the one-X Mobile Client software. This match is used to determine the user for the FNE feature.
     - Digits are matched from right to left.
     - The match must be at least 6 digits. If either the CLI or the Mobile Twinned Number is less than 6 digits no match will occur.
     - Matching is done for up to 10 digits. Further digits are ignored. If either the CLI or Mobile Twinned Number is less than 10 digits, matching stops at that shorter length.
     - If multiple matches occur the first user in the configuration is used. Manager will warn against configuration where such a conflict may exist.

3. **FNE System Short Code**
   This short code is used as the destination for calls received on the DID numbers requested for one-X Mobile usage.

   - **Short Code:** 777XX
   - **Number:** N
   - **Feature:** FNE 469

4. **Incoming Call Route**
   An incoming call route is required to send the DID calls to the FNE short code. Since a continuous range of DID numbers has been requested, with end digits that match the feature required, this can be done with a single incoming call route entry in the system configuration.

   - **Incoming Number:** 99XX
   - **Destination:** 777#
     The # matches any digits received for the XX part of the incoming number.
   - Other fields can be configured as required. For example using line group ID's to restrict one-X Mobile client access to those trunks known to support this feature.

one-X Mobile Client Configuration File

During installation of the one-X Mobile software, a settings files is installed on the phone. For Windows phones, the file is called *settings.ini* file, for Symbian single-mode phones the file is called *setting.1xme* file. For some phones, once the software has been installed, the settings file values can be changed through the phone interface.

Example Settings Files	
Windows Mobile 'settings.ini'	**Symbian 'settings.1xme'**
Dialing Prefixes and Codes	
PRE_IMS=Dial; DID_PREFIX = 1555776; INTERNATIONAL_DIRECT_DIAL_PREFIX = 011; NATIONAL_DIRECT_DIAL_PREFIX = 1; HOME_COUNTRY_DIAL_CODE = +1; ARS_CODE = ; EXTENSION_LENGTH = 3; NATIONAL_NUMBER_LENGTH = 10; USERS_EMERGENCY_NUMBERS = 112,999,911; SETTINGS_PIN = 1234;	DID_PREFIX = 1555776; INTERNATIONAL_DIRECT_DIAL_PREFIX = 011; NATIONAL_DIRECT_DIAL_PREFIX = 1; HOME_COUNTRY_DIAL_CODE = +1; ARS_CODE = ; EXTENSION_LENGTH = 3; NATIONAL_NUMBER_LENGTH = 10; USERS_EMERGENCY_NUMBERS = 112,999,911; SETTINGS_PIN = 1234;
DDI Suffixes for Features These are used with the DDI_Prefix above to complete the DDI number for a particular FNE feature (see the table of possible features above).	
IDLE_APPEARANCE_SELECT = 9900; ACTIVE_APPEARANCE_SELECT = 9901; AUTO_CALL_BACK_TOGGLE = 9902; DISABLE_AUTO_CALL_BACK_TOGGLE = 9903; CALL_FORWARDING_ALL_ACTIVATION = 9904; CALL_FORWARDING_BUSY_NO_ANSWER_ACTIVATION = 9905; CALL_FORWARDING_DISABLE = 9906; CALL_PARK = 9907; CALL_UNPARK = 9908; CALL_PICKUP_GROUP = 9909; CALL_PICKUP_DIRECTED = 9910; CALL_PICKUP_GROUP_EXTENDED = ; CALLING_PARTY_NUMBER_BLOCK = 9912; CALLING_PARTY_NUMBER_UNBLOCK = 9913; CONFERENCE_ON_ANSWER = 9914; DROP_LAST_ADDED_PARTY = 9915; EXCLUSION = 9916; HELD_APPEARANCE_SELECT = 9917; OFF_PBX_ENABLE = 9919; OFF_PBX_DISABLE = 9920; SEND_ALL_CALLS_ENABLE = 9924; SEND_ALL_CALLS_DISABLE = 9925; TRANSFER_ON_HANGUP = 9926; TRANSFER_TO_COVERAGE = 9927;	IDLE_APPEARANCE_SELECT = 9900; ACTIVE_APPEARANCE_SELECT = 9901; AUTO_CALL_BACK_TOGGLE = 9902; DISABLE_AUTO_CALL_BACK_TOGGLE = 9903; CALL_FORWARDING_ALL_ACTIVATION = 9904; CALL_FORWARDING_BUSY_NO_ANSWER_ACTIVATION = 9905; CALL_FORWARDING_DISABLE = 9906; CALL_PARK = 9907; CALL_UNPARK = 9908; CALL_PICKUP_GROUP = 9909; CALL_PICKUP_DIRECTED = 9910; CALL_PICKUP_GROUP_EXTENDED = ; CALLING_PARTY_NUMBER_BLOCK = 9912; CALLING_PARTY_NUMBER_UNBLOCK = 9913; CONFERENCE_ON_ANSWER = 9914; DROP_LAST_ADDED_PARTY = 9915; EXCLUSION = 9916; HELD_APPEARANCE_SELECT = 9917; OFF_PBX_ENABLE = 9919; OFF_PBX_DISABLE = 9920; SEND_ALL_CALLS_ENABLE = 9924; SEND_ALL_CALLS_DISABLE = 9925; TRANSFER_ON_HANGUP = 9926; TRANSFER_TO_COVERAGE = 9927;
Enterprise Settings	
SUB_MENU_NAME = My Company; <Voicemail> = *17;	SUB_MENU_NAME = My Company; <Voice Mail> = *17;

- Do NOT modify the tag names shown before the equals signs.
- Do NOT remove any tag rows. If a features is not being used or is unavailable leave the value between the '=' and ';' blank.
- Do NOT remove the terminating semi-colon.
- All values MUST go between the ‘=’ and ‘;’.
- For those tags that match FNE features, the value should be the DID digits that will be passed to the system for routing to the appropriate short code. This value is used in conjunction with the DID prefix to form the full telephone number.
- The following tags MUST have values set (not necessarily those shown as examples below):

```
DDI_PREFIX = +1555776
INTERNATIONAL_DIRECT_DIAL_PREFIX = 011;
NATIONAL_DIRECT_DIAL_PREFIX = 1;
HOME_COUNTRY_DIAL_CODE = 1;
ARS_CODE = ;
EXTENSION_LENGTH = 3;
NATIONAL_NUMBER_LENGTH = 10;
USERS_EMERGENCY_NUMBERS = 112,999,911;
```

Configuration File Fields

- LOCATION_NAME = ; *for example LOCATION_NAME = Boston;*
  This is a required tag for Windows Mobile .ini settings file only. It is not present or required for a Symbian configuration file. The string defines the location name of the one-X Mobile server and it will be displayed in the Avaya one-X Mobile menu on the mobile phone. If you have multiple locations, each location must have a unique LOCATION_NAME value.

- PRE_IMS = ; *This tag is not used.*

- DID_PREFIX = ; *for example DID_PREFIX = 173255512;*
  This required tag defines the DID/DDI prefix that is used to specify the FNE features in the system. For example, if you are using 73255512XX as DID/DDI numbers to activate your FNEs, the DID_PREFIX as shown above would be set intentionally short 2 digits since the FNE tag will be populated with the last 2 digits (e.g. IDLE_APPEARANCE_SELECT = 85;). The DID_PREFIX and the FNE digits when put together must equal the full DID/DDI number like in this example.

- INTERNATIONAL_DIRECT_DIAL_PREFIX = ; *for example INTERNATIONAL_DIRECT_DIAL_PREFIX = 011;*
  This required tag specifies the international call prefix for your country.

- NATIONAL_DIRECT_DIAL_PREFIX = ; *for example NATIONAL_DIRECT_DIAL_PREFIX = 1;*
  This optional tag specifies the prefix for dialing national numbers.

- HOME_COUNTRY_DIAL_CODE = ; *for example HOME_COUNTRY_DIAL_CODE = 1;*
  This required tag is the home country code.

- ARS_CODE = ; *for example ARS_CODE = 9;*
  This optional tag set the ARS access code.

- EXTENSION_LENGTH = ; *for example EXTENSION_LENGTH = 5;*
  This required tag sets the dial plan length of the system.

- NATIONAL_NUMBER_LENGTH = ; *for example NATIONAL_NUMBER_LENGTH = 9, 10;*
  This required tag specifies the number of digits for a national number. This field accepts multiple numbers separated by commas for countries that have different number lengths.

- USERS_EMERGENCY_NUMBERS = ; *for example USERS_EMERGENCY_NUMBERS = 911;*
  This is a required tag and it specifies the numbers dialed for emergency.

- SETTINGS_PIN = ; *for example SETTINGS_PIN = 1234;*
  This is an optional tag. When set, the client will prompt for this pin when installing or modifying the configuration.

9.5 Advice of Charge

IP Office 4.0+: The system supports advice of charge (AOC) on outgoing calls to ISDN exchanges that provide AOC information. It supports AOC during a call (AOC-D) and at the end of a call (AOC-E). This information is included in the SMDR output.

AOC is only supported on outgoing ISDN exchange calls. It is not supported on incoming calls, reverse charge calls, QSIG and non-ISDN calls. Provision of AOC signalling will need to be requested from the ISDN service provider and a charge may be made for this service.

For users, display of AOC information is only supported on T3 phones, T3 IP phones and Phone Manager.

- The user who makes an outgoing call is assigned its charges whilst they are connected to the call, have the call on hold or have the call parked.
- If AOC-D is not available, then all indicated by AOC-E are assigned to the user who dialed the call.
- If AOC-D is available:
  - If the call is transferred (using transfer, unpark or any other method) to another user, any call charges from the time of transfer are assigned to the new user.
  - If the call is manually transferred off-switch, the call charges remain assigned to the user who transferred the call.
  - If the call is automatically forwarded off switch, subsequent call charges are assigned to the forwarding user.
  - AOC-D information will only be shown whilst the call is connected. It will not be shown when a call is parked or held.
  - Call charges are updated every 5 seconds.
- For conference calls all call charges for any outgoing calls that are included in the conference are assigned to the user who setup the conference, even if that user has subsequently left the conference.

Enabling AOC Operation

1. **Set the System Currency**
   The Default Currency 160 (System | Telephony | Telephony 160) setting is by default set to match the system locale. Note that changing the currency clears all call costs stored by the system except those already logged through SMDR.

2. **Set the Call Cost per Charge Unit for the Line**
   AOC can be indicated by the ISDN exchange in charge units rather than actual cost. The cost per unit is determined by the system using the **Call Cost per Charge Unit** setting which needs to be set for each line. The values are 1/10,000th of a currency unit. For example if the call cost per unit is £1.07, a value of 10700 should be set on the line.

3. **Applying a Call Cost Markup**
   It may be a requirement that the cost applied to a user's calls has a mark-up (multiplier) applied to it. This can be done using the Call Cost Markup 274 (User | Telephony | Call Settings 274) setting. The field is in units of 1/100th, for example an entry of 100 is a markup factor of 1.

4. **Enable User AOC Display**
   By default users do not see call charges. The **Display Charges** (User | T3 Options | T3 Telephony) setting is used to switch this option on or off. Note that the display of AOC information is only supported on T3 phones.

AOC Short Codes
A number of short code features exist that can be used with AOC. These features can only be used with T3 phones.

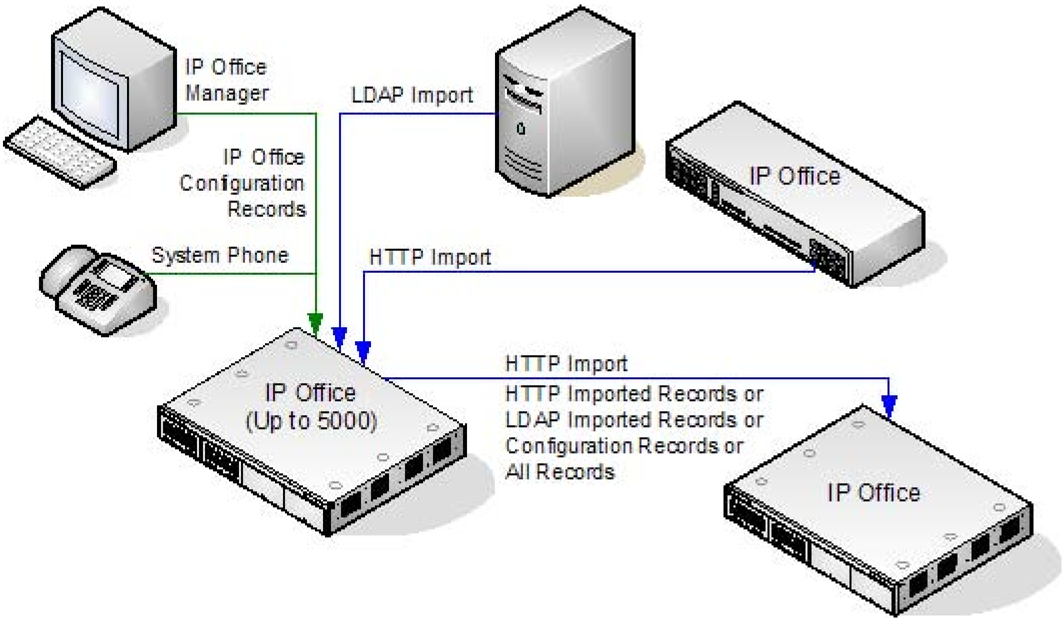
- **AOC Previous Call**
  Displays the call costs of the user's previous call if AOC information was provided with that call.
- **AOC Total**
  Display the cumulative total cost of the user's calls for which AOC information is available.
- **AOC Reset Total**
  Set the cumulative total (units and cost) for the user's calls back to zero.

9.6 Centralized System Directory

Directory services can be used to import directory entries (names and numbers) from external sources. These sets of entries are regularly re-imported.

For systems, the directory records can come from the following sources:

- LDAP Import 169
  The system can import up to 5000 LDAP records for use within directories shown by user phones and applications. LDAP import is configured through the System | Directory Services | LDAP 169 form. For pre-IP Office Release 5.0+ system, LDAP was restricted to 500 records and only displayed within user applications.
- HTTP Import 171 (IP Office Release 5.0+)
  Systems are able to import the directory entries from another system using HTTP. HTTP import is configured through the System | Directory Services | HTTP 171 form by specifying an IP address (or SCN connection). The records imported can be any or all of the following record types held by the system from which the records are being imported: LDAP imported entries, HTTP imported entries, configuration entries.
- System Directory Records 358 *(Configuration entries)*
  Up to 2500 records can be entered directly into the system configuration through the Directory 358 menu. System directory records override matching LDAP/HTTP imported records. For pre-IP Office Release 5.0+ system, the number of system directory records was 1000.
  - IP Office Release 5+: 1400, 1600 and 9600 Series phones with a CONTACTS button and System Phone 765 privileges, can add, delete and edit the system directory records of the system at which they are logged in. They cannot edit LDAP or HTTP imported entries.



System	Number of Directory Records			Total Number of Directory Records
	Configuration	LDAP Import	HTTP Import	
IP500/ IP500v2	2500	5000	5000	5000
IP412	2500	2500	2500	2500
IP406 V2	2500	2500	2500	2500

Directory entries are used for two types of function:

- **Directory Dialing**
  Directory numbers are displayed by user applications such as Phone Manager and SoftConsole. Directory numbers are viewable through the Dir 615 function on many Avaya phones (**Contacts** or **History**). They allow the user to select the number to dial by name. The directory will also contain the names and numbers of users and hunt groups on the system.
  - The Dir function groups directory entries shown to the phone user into the following categories. Depending on the phone, the user may be able to select the category currently displayed. In some scenarios, the categories displayed may be limited to those supported for the function being performed by the user:
    - **External**
      Directory entries from the system configuration. IP Office 5.0+: This includes HTTP and LDAP imported entries.
    - **Groups**
      Groups on the system. If the system is in a Small Community Network it will also include groups on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - **Users or Index**
      Users on the system. If the system is in a Small Community Network it will also include users on other systems in the network (For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses).
    - **Personal**
      Available for T3 phones, T3 IP phones, 1400, 1600 and 9600 Series phones. These are the user's personal directory entries stored within the system configuration.
- **Name Matching**
  Directory entries are also used to associate a name with the dialled number on outgoing calls or the received CLI on incoming calls. When name matching is being done, a match in the users personal directory overrides any match in the system directory. Note that some user applications also have their own user directory.
  - The IP Office Phone Manager and SoftConsole applications have their own user directories which are also used by the applications name matching. Matches in the application directory may lead to the application displaying a different name from that shown on the phone.
  - Name matching is not performed when a name is supplied with the incoming call, for example QSIG trunks.
  - Directory name matching is not supported for DECT handsets.

The following characters *(IP Office Release 5.0+)* are supported in directory entries. They are supported in both system configuration entries and in HTTP/LDAP imported entries.

- **? = Any Digit**
  Directory entries containing a ? are only used for name matching against the dialed or received digits on outgoing or incoming. They are not included in the directory of numbers to dial available to users through their phones or applications. The wildcard can be used in any position but typically would be used at the end of the number.
  - In the following example, any calls where the dialed or received number is 10 digits long and starts 732555 will have the display name Homdel associated with them.
    - **Name:** Holmdel
    - **Number:** 9732555????
- **(and) brackets = Optional Digits**
  These brackets are frequently used to enclose an optional portion of a number, typically the area code. Only one pair of brackets are supported in a number. Entries containing digits inside ( ) brackets are used for both name matching or user dialling. When used for name matching, the dialed or received digits are compared to the directory number with and without the ( ) enclosed digits. When used for dialling from a phone or application directory, the full string is dialed with the ( ) brackets removed.
  - The following example is a local number. When dialed by users they are likely to dial just the local number. However on incoming calls, for the CLI the telephony provider includes the full area code. Using the ( ) to enclose the area code digits, it is possible for the single directory entry to be used for both incoming and outgoing calls.
    - **Name:** Raj Garden
    - **Number:** 9(01707)373386
- **Space and - Characters**
  Directory entries can also contain spaces and - characters. These will be ignored during name matching and dialing from the directory.

Imported Records

- Imported directory records are temporary until the next import refresh. They are not added to the system's configuration.
- They cannot be viewed or edited using Manager or edited by a System Phone user.
- The temporary records are lost if the system is restarted. However the system will request a new set of imported directory records after a system restart.
- The temporary records are lost if a configuration containing Directory changes is merged. The system will then import a new set of temporary records without waiting for the *Resync Interval*.
- If an configuration record is edited by a System Phone user to match the name or number of a temporary record, the matching temporary record is discarded.

Importation Rules

When a set of directory records is imported by HTTP or LDAP, the following rules are applied to the new records:

- Imported records with a blank name or number are discarded.
- Imported records that match the name or number of any existing record are discarded.
- When the total number of directory records has reached the system limit, any further imported records are discarded.

System	Number of Directory Records			Total Number of Directory Records
	Configuration	LDAP Import	HTTP Import	
IP500 / IP500v2	2500	5000	5000	5000
IP412	2500	2500	2500	2500
IP406 V2	2500	2500	2500	2500

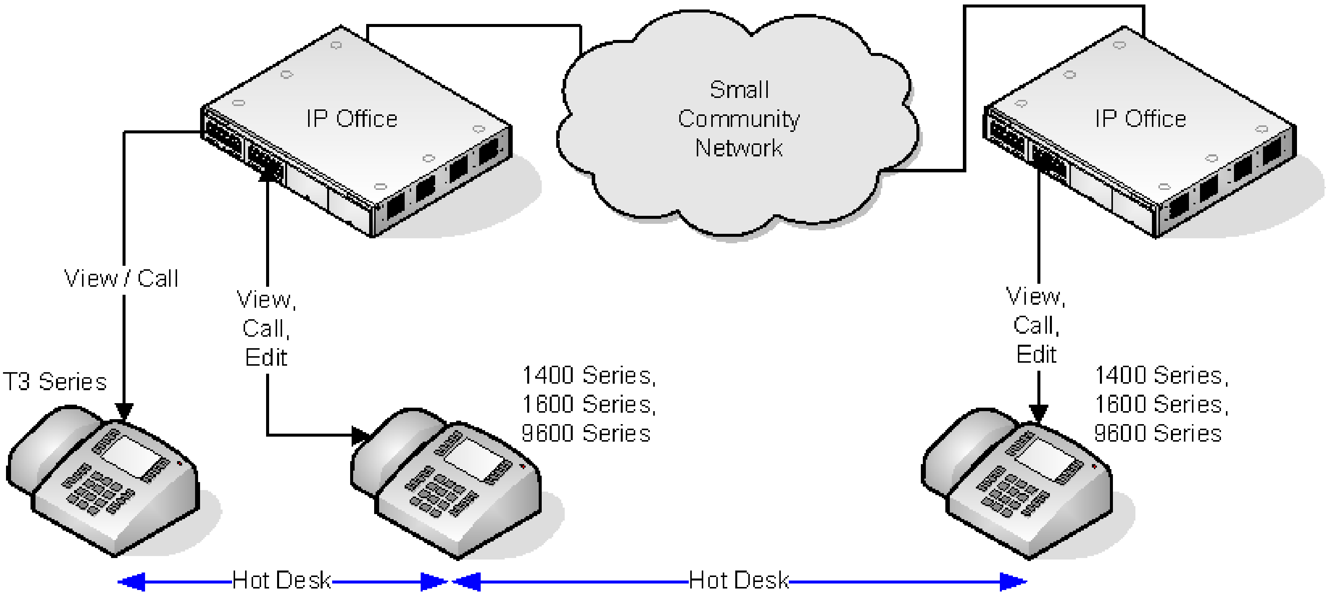
9.7 Centralized Personal Directory

Each system user is able to have up to 100 personal directory records stored by the system unless the system limit has been reached.

Directory Records	Per User	System Total Maximum
IP500/IP500v2	100	10800
IP412	100	3600
IP406 V2	100	1900

Pre-IP Office 5.0: A user's personal directory is only usable from T3 Series phones.

IP Office 5.0+: A user's personal directory is also useable with 1400, 1600 and 9600 Series phones with a CONTACTS button. The user can view these records and use them to make calls. 1400, 1600 and 9600 Series phone users can edit their personal directory records through the phone. The user personal directory entries can be edited using the Manager User | Personal Directory 301 menu.



When the user hot desks to another phone that supports the centralized personal directory, their personal directory entries become accessible through that phone. For IP Office Release 5.0+ that also includes hot desking to another IP Office in the Small Community Network.

Users can also use and edit their personal directory records using one-X Portal for IP Office. Note that using one-X Portal for IP Office, users can have more that 100 personal directory records, with excess records stored by the one-X Portal server.

9.8 Centralized Call Log

IP Office 5.0+: The system can store a centralized call log 745 for users. Each users' centralized call log can contain up to 30 call records for user calls (10 on IP412 and IP406 V2 systems). When this limit is reached, new call records replace the oldest record.

On 1400, 1600 and 9600 Series phones with a Call Log or History button, that button can be used to display the user's centralized call log. The user can use the call log to make calls or to store as a personal speed dial. They can also edit the call log to remove records. The same call log is also if the user logs into one-X Portal for IP Office application.

The centralized call log moves with the user if they log on and off from different phones. This includes if they hot desk within a Small Community Network.

Call Log Information

The following information is included in each centralized call log record:

Information	Description
Name	The name, of the caller or the party called, if available. Up to 31 characters. This text is similar to that shown on the phone display of phones when they receive the call. For example, on forwarded call details of the original target and the caller name are included, eg. Bob > Sue.
Number	The number associated with the call. Up to 31 digits.
Tag	A text tag can be associated with calls by several different methods. See Call Tagging 756. Up to 31 characters. The tag is not shown within the call log display on phones.
Time and Date	The time and date of the call using the system time.
Duration	The call duration. For outgoing and answered calls this is the call connection time. For missed calls this is the call ringing time.

Record Type	Call log records can be Incoming, Outgoing or Missed. Note that these are calls to or from the user, not the phone, so it can include calls handled through a twinned device, mobile call control 730, one-X Mobile Client 734 or Phone Manager Telecommuter mode.
	
	• Incoming
	Calls to the user that the user then answered. This includes calls that the user answers on a twinned device. This also includes outgoing calls that are transferred to and answered by the user.
	
	• Outgoing
	Calls made by the user.
	
	• Missed
	Calls to the user that they did not answer. This includes calls while the user is logged off or in Do Not Disturb state.
	  ◦ Missed call records include an indication of what happened to the missed call. Options are Answered by Another, Answered by Voicemail or Lost (not answered on the system).
	  ◦ Missed call records are also marked as either acknowledged or unacknowledged. If the user's call log contains any unacknowledged call log records, the Call Log lamp is lit when using a 1608 or 1616 phone. From the phone, viewing an unacknowledged record changes it to acknowledged.
	  ◦ If the user has also be configured to included missed hunt group calls in their call log, those are also marked as acknowledged or unacknowledged.

Count	The number of times a matching call has been logged. A matching call is one with the same name, number and type. Only one record is kept for matching calls, with the count increased by 1 and using the time and date of the most recent matching call.

If missed hunt group calls are also being logged, the system stores up to 10 call records for each hunt group. When this limit is reached, new call records replace the oldest record.

Controlling Centralized Call Logging

The following controls exist for which users have their calls included in the centralized call log and which calls are included.

User Setting

The user centralized call log settings can be set through the user configuration (User | Telephony | Call Log 280) or through their associated user rights (User Rights | Telephony | Call Log 394).

- Centralized Call Log: *Default = System Default (On)*
  This setting allows the use of centralized call logging to be enabled or disabled on a per user basis. The default is to match the system setting Default Centralized Call Log On 165 (System | Telephony | Call Log 165). The other options are *On* or *Off* for the individual user. If off is selected, the call log shown on the users phone is the local call log stored by the phone.

System Settings (System | Telephony | Call Log 165)

- Default Centralized Call Log On: *Default = On.*
  When selected, each user is defaulted to have the system store a call log of their calls. This call log will be displayed on the phone when the user is using a 1400, 1600 or 9600 Series phone. The use of centralized call logging can be enabled/disabled on a per user basis using the Centralized Call Log 280 user setting (User | Telephony | Call Log 280).
- Logged Missed Calls Answered at Coverage: *Default = Off.*
  This setting controls how calls to a user, that are answered by a covering user should be logged in the centralized call log. This option applies for calls answered elsewhere (covered) by pickup, call coverage (call coverage 686 buttons or coverage group 748), bridged appearance button, user BLF, voicemail, etc.

Setting	Targeted User	Covering User
Off (Default)	Nothing	Answered Call
On	Missed Call	Answered Call

- Log Missed Hunt Group Calls: *Default = Off.*
  By default, hunt group calls are not included in any user's centralized call log unless answered by the user. If this option is selected, a separate call log is kept for each hunt group of calls that are not answered by anyone. It includes hunt group calls that go to voicemail.
  - If missed hunt group calls are also being logged, the system stores up to 10 call records for each hunt group. When this limit is reached, new call records replace the oldest record.
  - Within the user call log setting (User | Telephony | Call Log 280) the list of hunt groups allows selection of which hunt groups' missed call records should be displayed as part of the centralized call log when a user is using a 1608 or 1616 phone.

Call Scenarios

This is not a comprehensive list. However it summarizes how the user call log is used in some common call scenarios.

Scenarios	User Call Log Notes
Authorization / Account Codes	Account and authorization codes used as part of a call are not included in user call logs.
Automatic Callback	If answered, they will show as an outgoing call to the target.
Application Calls	Calls made and answered using applications (including CTI interfaces) are logged as if the user made or answered the call using an extension.
Conference Calls	Conference calls are not included in the user call log.
Hold	When a user holds and then un-holds a call, the call duration includes the time the call was on hold.
Follow-Me	Calls to the user still appear in their user call log. The follow me calls do not appear in the user call log of the user who was the follow me destination.
Forward on Busy	If the forwarded call is answered, the forwarding user will have a Missed - Answered by Other call log record.
Forward on No Answer	If the forwarded call times out to voicemail, the user will have a Missed - Answered by Voicemail call log record.
Forward Unconditional	When forwarding to another number, there will be no record of forwarded calls in the forwarding users call log. When using the To Voicemail option, the forwarded call will by logged as a Missed - Answered by Voicemail call record.
Page Calls	Page calls are not included in any user call logs unless the page is answered (by pressing Conference). When answered the page is logged as a normal call between the two users involved.
Park	Retrieving a call from Park (even if the user is the one who parked the call) is logged as a incoming call.
Short Codes	Calls are only logged if they result in a call being made or a call being answered. Calls made using Break Out are not included.
Suppressed Digits	Calls made with digit suppression enabled (AD Suppress button 578) are not included in the users call log.
Transfers	If the user answers and accepts a supervised transfer, they will have a incoming calls records. One for the transfer enquiry call and one for the transferred call. If the user is the target of an unsupervised transfer, they will have an Incoming or Missed call log. Note that even if the call being transferred was originally an outgoing call, for the user answering the transfer it is logged as a incoming call.
Twinning and Mobility	When a user has a twinned device (either internal twinning or mobile twinning), the user's call log operates regardless of which device the user uses to make or answer calls. Calls between the twinned devices, ie. the user transferring a call between devices, are not included in their call log. This includes calls made using mobile call control or a one-X Mobile client.

Small Community Network	The user's call log records are stored by the system that is their home system, ie. the one on which they are configured. When the user is logged in on another system, new call log records are sent to the user's home system, but using the time and date on the system where the user is logged in. Hunt group call log records are stored on the system on which the hunt group is configured.

9.9 Coverage Groups

IP Office 5.0+: For users with a **Coverage Group** selected, coverage group operation is applied to all external calls that are targeted to the user.

For external calls:

- In scenarios where an external call would normally have gone to voicemail, it instead continues ringing and also starts alerting the members of the coverage group.
- The follow me settings of Coverage Group members are used, the forwarding settings are not.
- If the user is not available, for example if they have logged off or set to do not disturb, coverage group operation is applied immediately.
- If the user is configured for call forward on busy, coverage operation is applied to the user's calls forwarded to the forward on busy destination.

Coverage group operation is not applied to the following types of call:

- Hunt group calls.
- Recall calls such as transfer return, hold recall, park recall, automatic callback.

The Coverage Group is set through the user's User | Telephony | Supervisor Settings 276 or through their associated User Rights | Telephony | Supervisor Settings 393. The only group settings used are:

- The list of group members. They are treated as a collective group regardless of the group's configuration.
- If the group has **Night Server Fallback Group** and or **Out of Service Fallback Group** set, the members of those groups are used if the coverage group is set to night service mode or out of service mode respectively.

9.10 Fax Relay

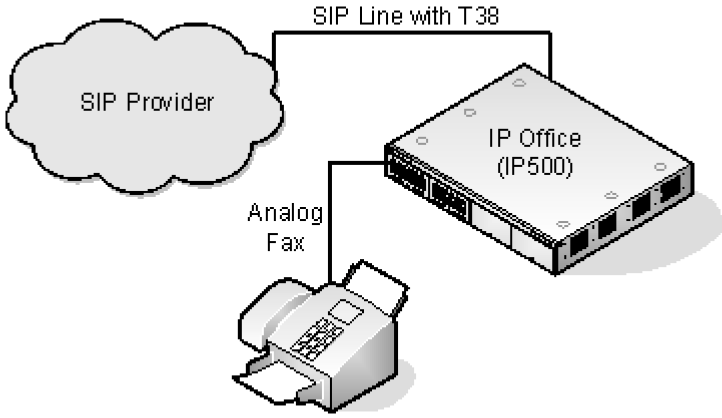
Group 3 Fax is the common standard for fax transmission over analog and digital (TDM) phone lines. However, the sending of fax calls over IP lines is not normally possible or proves unreliable due to the distortion caused during encoding of the audio signal. However some SIP line providers support fax using specific codecs.

For a IP500/IP500v2 system with an IP500 VCM card, T38 *(IP Office 6.0+)* can be selected for fax over SIP lines. T38 Fax Relay is a public set of protocols that allow fax calls to be reliable transported over IP connections that have a T38 Fax gateway at each end.

Fax support is enabled using the SIP line's Fax Transport Support setting. Within a Small Community Network 810 (SCN) of systems, Fax Transport Support can also be enabled on the H323 IP lines between the systems. This allows fax calls at one system to be sent to another system.

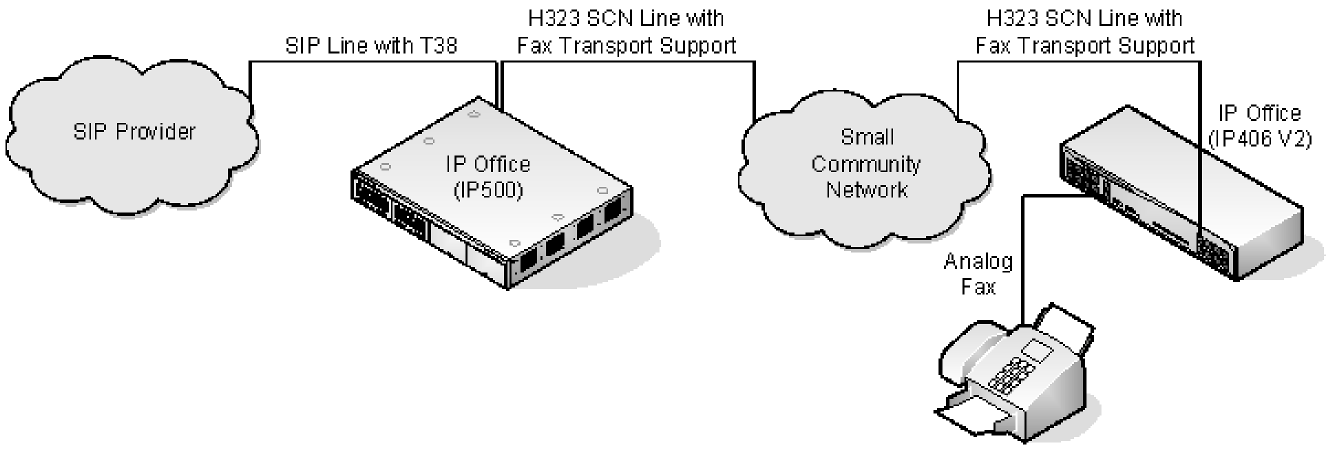
Both system SIP extensions and system SIP lines can be configured for T38. They can then be used as the point at which fax calls are sent and or received. Each fax call uses a VCM channel. The SIP line or extension must support Re-Invite. In addition, the existing system Fax Transport Support can still be used to transport the faxes over an SCN. That includes faxes being made or received via a SIP line or extension. The conversion from T38 or G711 to SCN or vice versa requires two VCM channels.

Scenario 1:
In this scenario, the SIP provider supports T38. By configuring the system's SIP line for T38 operation, the analog fax machine attached to the system can make and receive fax calls via the SIP provider.

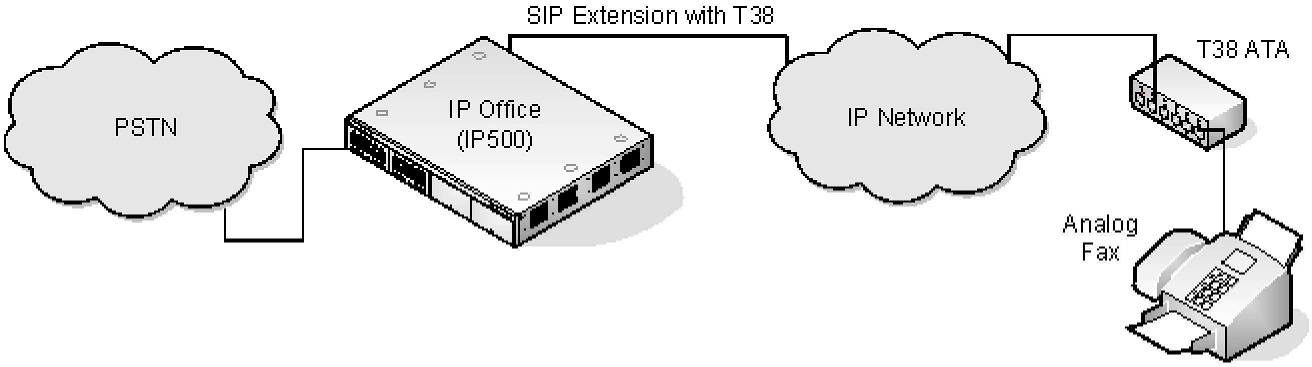


Scenario 2:
This scenario is similar to the previous. However, fax calls via the SIP line are also transported across the Small Community Network to a fax machine attached to another system in the network. In this scenario, on the IP500, 2 VCM channels are used for the T38 fax call.

This method is also significant in that it allows non-IP500 systems running IP Office 5 to make use of SIP lines for fax.



Scenario 3:
In this scenario, an ATA (Analog Telephone Adaptor) that supports T38 is used to connect the fax machine as a SIP extension on the system.



System Requirements for T38

- IP500/IP500v2 with an IP500 VCM.
- IP Office 5 software.
- The SIP line or extension must support Re-Invite.

Outgoing Fax Calls

When sending a fax via T38, the call must be correctly indicated as being (or potentially being) a fax call. This can be done in 2 ways.

- **Analog Fax Extension Setting**
  For an analog fax device, the extension's Equipment Classification 254 setting (Extension | Analog 254) can be set to *FAX Machine*.
- **Dial Fax Short Code**
  Calls can be routed to a Dial Fax 459 short code which has the SIP line as its destination.

9.11 Dial Tone Transfer

IP Office 5+: A user who is not able to make external calls to any or some external numbers, can be transferred to dial tone by a user who is able to make external calls.

1. The restricted user wanting to make the external call, dials the unrestricted user and requests dial tone.
2. The unrestricted user initiates a transfer and dials the prefix for an ARS form configured to provide secondary dial tone.
    - The prefix is a short code set up to access the required ARS form. While this can be a system short code, using a user or user rights short code will allow control over who can provide dial tone transfer for restricted users.
3. When they hear the secondary dial tone, the unrestricted user completes the transfer.
4. The restricted user now hears the secondary dial tone and is now able to make an external call.
    - The restricted user is now able to make calls as permitted by the short codes in the ARS form.
    - The restricted user is not able to transfer the dial tone to another user.

The ARS form being used can still contain short codes that restrict the dialing that can be attempted after the restricted user hears secondary dial tone. Other ARS features can also be used such as alternate routing or time profiles to provide out of hours routing. The ARS form timers are run from when the unrestricted caller dials the ARS form. They are not reset when the restricted user is transferred to the ARS form.

Multiple prefixes and ARS forms can be used if required to create more complex scenarios. For example, one where the unrestricted user can transfer the restricted users to an ARS forms that allows international calls or to an ARS form that only allows national dialing.

Example Configuration

The example below is a simple configuration that allows the unrestricted user to use 8 as a transfer destination that provides secondary dial tone.

1. Create an ARS Form for Secondary Dial Tone
   The ARS form needs to be created before short codes can be added to route callers to it.
    - Enter a Route Name to identify the ARS form, for example *Dial Tone Trans*.
    - Select Secondary Dial Tone.
    - Select either *System Tone* (this matches locale specific 844 normal dial tone) or *Network Tone* (this matches locale specific 787 secondary dial tone). For some locales both tones are the same.
    - Enter short codes that will take any digits dialed by the restricted user and process them for external dialing to an outgoing line group. For this example we will allow any digits dialed to be presented to the first trunk seized in outgoing line group 0.

Code	N
Telephone Number	N
Feature	Dial
Line Group ID	0

    - Other short codes can be used to allow or bar the dialing of specific numbers or types of numbers.
    - Configure the rest of the ARS form as required. For full details on ARS form configuration see ARS 407.
2. Create a Short Code for Dial Tone Transfer
   For this example we will allow the prefix 8 to be used to access an ARS form created above.
    - In the user short codes of the unrestricted user, create a short code that invokes the ARS form created above. For example:

Code	8
Telephone Number	
Feature	Dial
Line Group ID	51 Dial Tone Trans

    - It is important that the short code does not pass any digits to the ARS form. Once the ARS form receives any digits, it starts short code matching and ends secondary dial tone.
    - The short code could also be setup as a system or user rights short code.
3. The unrestricted user is now able to provide secondary dial tone to other users by on request by pressing Transfer, dialing 8 and then pressing Transfer again.

Account and Authorization Codes

If the restricted user enters an account or authorization code while calling the unrestricted user to request dial tone, that value is not carried forward with their external call once they have been provided with secondary dial tone.

If the unrestricted user enters an account or authorization code while dialing the ARS form, that value remains associated with the call made by the restricted user.

If the ARS form short code used to route the restricted users call requires an account or authorization code, the value already entered is used, otherwise the restricted user is prompted to enter a value.

Call Logging

The restricted user's outgoing call log will include the call to the unrestricted user and the outgoing external call they subsequently make. The outgoing external call record will include the prefix dialed by the unrestricted user to access the ARS form.

The unrestricted users call log will include just an incoming call from the restricted user.

Within the SMDR output, the calls by the restricted user are included. The call by the unrestricted user is not included.

9.12 Malicious Call Tracing (MCID)

IP Office 4.0+: MCID (Malicious Caller ID) is an ISDN feature. It is supported on BRI and PRI trunks to ISDN service provider who provide MCID.

When used, it instructs the ISDN exchange to perform a call trace on the user's current call and to keep a record of the call trace at the exchange for the legal authorities. Trace information is not provided to or displayed by the system or system phones.

The use of MCID is subject to local and national legal requirements that will vary. The feature may also not be enabled until specifically requested from the service provider. You should consult with your ISDN service provider and with appropriate legal authorities before attempting to use MCID.

Successful use of this feature is indicated by a tone on the phone and a "User Registered" message on T3 phones.

Activating MCID

1. Liaise with the ISDN Service Provider
   MCID should not be used with first confirming its usage with the ISDN service provider.

2. Enabling MCID Call Tracing on a Line
   BRI and PRI lines include a Support Call Tracing Option. which by default is off.

3. Enabling MCID Call Tracing for a User
   First the user must be allowed to use call tracing. Each user has a Can Trace Calls 276 (User | Telephony | Supervisor Settings 274) option. This option is off by default.

4. Providing an Active MCID Control
   The user needs to be provided with a mechanism to trigger the MCID call trace at the exchange. This can be done using either a short code or a programmable button.

   - MCID Activate Button
     The action MCID Activate 640 (Advanced | Miscellaneous | MCID Activate 640) can be assigned to a programmable buttons. It allows a malicious call trace to be triggered during a call.

   - MCID Activate Short Codes
     The feature MCID Activate can be used to create a short code to triggering a malicious call trace.

9.13 Call Barring

Call barring can be applied in a range of ways.

- **Barring a User From Making Any External Calls**
  For each user, Outgoing Call Bar 276 (User | Telephony | Supervisor Settings 276) can be selected to stop that user from making any outgoing calls.

- **Barring Particular Numbers/Number Types**
  System short codes are used to match user dialing and then perform a specified action. Typically the action would be to dial the number to an external line. However, short codes that match the dialing of particular numbers or types of numbers can be added and set to another function such as Busy. Those short codes can be added to a particular user, to a User Rights associated with several users or to the system short codes used by all users.

  - The system allows short codes to be set at user, user rights, system and least cost route. These have a hierarchy of operation which can be used to achieve various results. For example a system short code for a particular number can be set to busy to bar dialing of that number. For a specific user, a user short code match to the same number but set to Dial will allow that user to override the system short code barring.

- **Using Account Codes**
  The system configuration can include a list of account codes. These can be used to restrict external dialing only to users who have entered a valid account code.

  - **Forcing Account Code Entry for a User**
    A user can be required to enter an account code before the system will return dialing tone. The account code that they enter must match a valid account code stored in the system configuration. The setting for this is Force Account Code 276 (User | Telephony | Supervisor Settings 276).

  - **Forcing Account Code Entry for Particular Numbers**
    Each system short code has a Force Account Code option. Again the account code entered must match a valid account code stored in the system configuration. for the call to continue.

- **Barring External Transfers and Forwards**
  A user cannot forward or transfer calls to a number which they cannot normally dial. In addition there are controls which restrict the forwarding or transferring of external calls back off-switch. See Off-Switch Transfer Restrictions 782.

9.14 Caller Display

Caller display displays details about the caller and the number that they called. On internal calls, the system provides this information. On external calls it uses the Incoming Caller Line Identification (ICLID) received with the call. The number is also passed to system applications and can be used for features such as call logging, missed calls and to make return calls.

Analog extension can be configured for caller display via the system configuration (Extension | Extn | Caller Display Type).

- Adding the Dialing Prefix
  Some systems are configured to require a dialing prefix in front of external numbers when making outgoing calls. When this is the case, the same prefix must be added to the ICLID received to ensure that it can be used for return calls. The prefix to add is specified through the Prefix field of each line.
- Directory Name Matching
  The system configuration contains a directory of names and numbers. If the ICLID of an incoming call matches a number in the directory, the directory name is associated with that call and displayed on suitable receiving phones.
  - The SoftConsole and Phone Manager applications also have directories that can be used for name matching. If a match occurs, it overrides the system directory name match for the name shown by that application.

Extended Length Name Display

In some locales, it may be desirable to change the way names are displayed on phones in order to maximize the space available for the called or calling name. There are two hidden controls which can be used to alter the way the Manager displays calling and called information.

These controls are activated by entering special strings on the Source Numbers tab of the NoUser user. These strings are:

- LONGER_NAMES
  This setting has the following effects:
  - On DS phones, the call status display is moved to allow the called/calling name to occupy the complete upper line and if necessary wrap-over to the second line.
  - For all phone types:
    - On incoming calls, only the calling name is displayed. This applies even to calls forwarded from another user.
    - On outgoing calls, only the called name is displayed.
- HIDE_CALL_STATE
  This settings hides the display of the call state, for example CONN when a call is connected. This option is typically used in conjunction with LONGER_NAMES above to provide additional space for name display.

9.15 Call Intrusion

Call intrusion allows a user to join another users existing conversation. Once the intrusion has occurred, all parties can hear and talk to each other. Note that intruding uses Manager conference resources.

The ability to intrude is controlled by two Manager configuration settings, the Can Intrude 276 (User | Telephony | Supervisor Settings 276) and the Cannot Be Intruded 276 (User | Telephony | Supervisor Settings 276). By default no users can intrude and all users cannot be intruded.

- Bridging
  Users with call appearance buttons may be able to bridge into other calls. This is similar to intrusion but subject to different operation.

- Privacy *(IP Office 4.0+)*
  The system supports privacy features that allow a user to indicate that a call cannot be intruded on. See Private Calls 757.

Below is an example of a short code, which can be used to attempt call intrusion. Using it the intruder would dial *90*N#, replacing the N with the extension number of the user into whose call they need to intrude.

- Short Code: *90*N#
- Telephone No: N
- Feature: CallIntrude

The Dial Inclusion short code feature can be used instead of Call Intrude. It allows the intruder and the intrusion target to talk without the third party hearing them. During this type of intrusion, all parties hear a repeated intrusion tone. When the intruder hangs-up the original call parties are reconnected.

9.16 Call Tagging

Call tagging associates a text string with a call. That string remains with the call during transfers and forwards. That includes calls across a Small Community Network (SCN).

On Avaya display phones, the text is shown whilst a call is alerting and is then replace by the calling name and number when the call is connected. On analog phones with a caller ID display, the tag text replace the normal caller information.

Applications such as Phone Manager and SoftConsole display any call tag associated with a call. If the call is parked, the tag is shown on the call park slot button used.

A call tag can be added when making a call using Phone Manager, SoftConsole or one-X Portal for IP Office. A tag can be added to a call by an Incoming Call Route or by an Voicemail Pro Assisted Transfer action.

9.17 Private Calls

IP Office 4.0+: This feature allows users to mark a call as being private.

When on, any subsequent calls cannot be intruded on, bridged into or silently monitored until the user's private call status is switched off.

Note that use of private calls is separate from the user's intrusion settings. If a user is set to Cannot be Intruded, switching private calls off does not affect that status. To allow private calls to be used to fully control the user status, Cannot be Intruded should be disabled for that user.

Use of private calls can be changed during a call. Enabling privacy during a call will stop any current recording, intrusion or monitoring.

Privacy only applies to the speech part of the call. Call details are still recorded in the SMDR output and other system call status displays.

- Button Programming
  The button programming action Advanced | Call | Private Call can be used to switch privacy on/off. Unlike the short code features it can be used during a call to apply or remove privacy from current calls rather than just subsequent calls. On suitable phones the button indicates the current status of the setting.

- Short Codes
  A number of short code features are available for privacy.

  - Private Call
    Short codes using this feature toggle private status on/off for the user's subsequent calls.

  - Private Call On
    Short codes using this feature enable privacy for all the user's subsequent calls until privacy is turn off.

  - Private Call Off
    Short codes using this feature switch off the user's privacy if on.

9.18 Call Pickup

Call pickup allows a user to answer a call ringing at another phone.

The following default short codes can be used:

- *30 - Call Pickup Any
  Answers the longest ringing call on the system. On large systems it is recommended that this short code is removed as it becomes difficult for users to predict which call they are answering.

- *31 - Call Pickup Group
  Pickup the longest ringing call to the hunt groups of which the user is a member.

- *32*N# - Call Pickup Extn
  Pick up the call ringing at a specific extension. When dial, N is replaced by the extension number.

- *53*N# - Call Pickup Members
  Pick up any call ringing on another extension that is a member of the Hunt group specified. The call picked up does not have to be a hunt group call. When dial, replace N with the hunt group extension number.

9.19 Call Waiting

Call waiting allows a user who is already on a call to be made aware of a second call waiting to be answered.

User Call Waiting

Call waiting is primarily a feature for analog extension users. The user hears a call waiting tone and depending on the phone type, information about the new caller may be displayed. The call waiting tone varies according to locale 846.

For Avaya feature phones with multiple call appearance buttons, call waiting settings are ignored as additional calls are indicated on a call appearance button if available.

To answer a call waiting, either end the current call or put the current call on hold, and then answer the new call. Hold can then be used to move between the calls.

Call waiting for a user can be enabled through the system configuration (User | Telephony | Call Settings | Call Waiting On 274) and through programmable phone buttons.

Call waiting can also be controlled using short codes. The following default short codes are available when using Call Waiting.

- ***15 - Call Waiting On**
  Enables call waiting for the user.
- ***16 - Call Waiting Off**
  Disables call waiting for the user.
- ***26 - Clear Call and Answer Call Waiting**
  Clear the current call and pick up the waiting call.

Hunt Group Call Waiting

Call waiting can also be provided for hunt group calls. The hunt group **Ring Mode** must be ***Collective Call Waiting***.

On pre-IP Office 4.0 systems, all the group members must have their own call waiting setting switched on, this no longer applies for Manager 4.0+.

On phones with call appearance buttons, the call waiting indication takes the form of an alert on the next available call appearance button. On other phones, call waiting indication is given by a tone in the speech path (the tone is locale specific).

- Pre-IP Office 4.0: All the users in the group must also have their own **Call Waiting** setting set to **On**.
- IP Office 4.0+: The user's own **Call Waiting** setting is overridden when they are using a phone with call appearances. Otherwise the user's own **Call Waiting** setting is used in conjunction with the hunt group setting.

9.20 Parking Calls

Parking a call is an alternative to holding a call. A call parked on the system can be retrieved by any other user if they know the system park slot number used to park the call. When the call is retrieved, the action is known as Unpark Call or Ride Call. While parked, the caller hears music on hold if available.

Each parked call requires a park slot number. Attempting to park a call into a park slot that is already occupied causes an intercept tone to be played. Most park functions can be used either with or without a specified park slot number. When parking a call without specifying the park slot number, the system automatically assigns a number based on the extension number of the person parking the call plus an extra digit 0 to 9. For example if 220 parks a call, it is assigned the park slot number 2200, if they park another call while the first is still parked, the next parked call is given the park slot number 2201 and so on.

- Park slot IDs can be up to 9 digits in length. Names can also be used for application park slots.

The **Park Timeout** setting in the system configuration (System | Telephony | Telephony | Park Timeout 160) controls how long a call can be left parked before it recalls to the user that parked it. The default time out is 5 minutes. Note that the recall only occurs if the user is idle has no other connected call.

There are several different methods by which calls can be parked and unparked. These are:

Using Short Codes
The short code features, Park Call 484 and Unpark Call 500, can be used to create short codes to park and unpark calls respectively. The default short codes that use these features are:

- *37*N# - Parks a call in park slot number N.
- *38*N# - Unparks the call in park slot number N.

Using the Phone Manager and SoftConsole Applications
The Phone Manager and SoftConsole applications all support park buttons. Phone Manager provides 4 park slot buttons, numbered 1 to 4. SoftConsole provides 16 park slot buttons numbered 1 to 16 by default. In both cases the park slot number for each button can be changed if required. Clicking on the buttons allows the user to park or unpark calls in the park slot associated with each button.

In addition, when a call is parked in one of those slots by another user, the application user can see details of the call and can unpark it at their extension.

Using Programmable Buttons
The Call Park feature can be used to park and unpark calls. If configured with a specified park slot number, the button can be used to park a call in that slot, unpark a call from that slot and will indicate when another user has parked a call in that slot. If configured without a number, it can be used to park up to 10 calls and to unpark any of those calls.

Phone Defaults
Some telephones support facilities to park and unpark calls through their display menu options (refer to the appropriate telephone user guide). In this case parked calls are automatically put into park slots matching the extension number.

9.21 Ring Back When Free

Also called 'callback' and 'automatic callback'. If the user called is busy, setting a ringback allows you to hang up and wait until they are free. When set, once the busy extension becomes free, the system will ring your telephone when also idle and, when answered, call the original target extension.

Note

- If the extension called has multiple call appearance buttons, you will not receive busy until all its call appearance buttons are in use.
- You must be idle (have no call connected or held) to receive the ringback.

This feature can also be set via the followings methods. These options also allow a ringback to be set when the target just rings, often known as a ringback when next used.

- Using a button programmed to the Ring Back When Free function
  If the button has status indication, it will show that a ring back has been set. The button can also be pressed again to cancel the ringback.
- Use a short code set to the Ring Back When Free function
  A short code set to this function can be used to set a ringback on any extension without having to actually make a call.
- Phone Manager
  This application provides the user with a button to set a ringback.
- Analog Phones
  Analog phone users can set a ringback when free by pressing any DTMF key before hanging up when they hear busy tone.

9.22 Message Waiting Indication

Message waiting indication (MWI) or a message lamp is supported for a wide variety of phones. It is used to provide the user with indication of when their voicemail mailbox contains new messages. It can also be configured to provide them with indication when selected hunt group mailboxes contain new messages.

Avaya digital and IP phones all have in-built message waiting lamps. Also for all phone users, the one-X Portal for IP Office and Phone Manager application provides message waiting indication.

Analog Phone Message Waiting Indication

For analog phones, the system supports a variety of analog message waiting indication (MWI) methods. Those methods are 51V Stepped, 81V, 101V and Line Reversal. The 101V method is only supported when using a Phone V2 expansion module.

81V is typically used in European countries. 51V Stepped is used in most other countries. However the actual method used for a particular model of analog phone should be confirmed with the phone manufacturer's documentation.

The method used for an individual analog extension is set for the Extn | Extn | Message Waiting Lamp Indication Type 252 field. This field also provides options for *None* (no MWI operation) and *On*. *On* selects a default message waiting indication method based on the system locale 844.

'On' Method	Locale
81V	Belgium, Denmark, Finland, France, Germany, Greece, Hungary, Iceland, Italy, Netherlands, Norway, Poland, Portugal, Russia, Saudi Arabia, Sweden, Switzerland, United Kingdom.
51V Stepped	Argentina, Australia, Brazil, Canada, Chile, China, Colombia, Japan, Korea, Mexico, New Zealand, Peru, South Africa, Spain, United States.

- For the United Kingdom system locale (eng), the default Caller Display Type (UK) allows updates of an analog phone's ICLID display whilst the phone is idle. The system uses this facilities to display the number of new messages and total number of messages in the users own mailbox. This feature is not supported with other Caller Display Types.

Hunt Group Message Waiting Indication

By default no message waiting indication is provided for hunt group voicemail mailboxes. Message waiting indication can be configured by adding an H entry followed by the hunt groups name to the Source Numbers tab of the user requiring message waiting indication for that hunt group. For example, for the hunt group Sales, add HSales. Hunt group message waiting indication does not require the user to be a member of the hunt group

9.23 Ring Tones

Ring tones can be defined in the following terms:

- Distinctive Ringing - Inside, Outside and Ringback
  A distinctive ring tone can be given for each of the different call types: an internal call, an external call and a ringback calls (voicemail calls, ringback when free calls, calls returning from park, hold or transfer).
  - The distinctive ringing patterns used for most non-analog phones are as follows:
    - Internal Call: Repeated single-ring.
    - External Call: Repeated double-ring.
    - Ringback Call: Repeated single-ring followed by two short rings.
  - For analog extensions, the ringing pattern used for each call type can be set through the system configuration in Manager. This is done using the settings on System | Telephony | Tones & Music 163 and or User | Telephony | Call Settings 274 tabs.
  - For non-analog extension the ringing pattern used for each call type by the system is not configurable.
- Personalized Ringing
  This term refers to control of the ringing sound through the individual phones. For non-analog phones, while the distinctive ringing patterns cannot be changed, the ringer sound and tone may be personalized depending on the phone's own options. Refer to the appropriate telephone user guide.

Analog Phone Ringing Patterns

For analog phone users, the distinctive ringing pattern used for each call type can be adjusted. From the System | Telephony | Tones & Music 163 tab, the default ring tone for each call type can be configured. The setting for an individual user associated with the analog extension can be altered from the system default through the User | Telephony | Call Settings 274 tab.

Note that changing the pattern for users associated with fax and modem device extensions may cause those devices to not recognize and answer calls.

The selectable ringing patterns are:

- RingNormal
  This pattern varies to match the Locale set in the System | System tab. This is the default for external calls.
- RingType1: 1s ring, 2s off, etc. This is the default for internal calls.
- RingType2: 0.25s ring, 0.25s off, 0.25s ring, 0.25s off, 0.25s ring, 1.75s off, etc. This is the default for ringback calls.
- RingType3: 0.4s ring, 0.8s off, ...
- RingType4: 2s ring, 4s off, ...
- RingType5: 2s ring, 2s off, ...
- RingType6: 0.945s ring, 4.5s off, ...
- RingType7: 0.25s ring, 0.24 off, 0.25 ring, 2.25 off, ...
- RingType8: 1s ring, 3s off, ...
- RingType9: 1s ring, 4s off, ...
- RingType0: Same as RingNormal for the United Kingdom locale.
- Default Ring: Shown on the User | Telephony | Call Setting 274 tab. Indicates follow the settings on the System | Telephony | Tones & Music 163 tab.

9.24 Music On Hold (MOH)

The system can provide music on hold (MOH) from either internally stored files or from externally connected audio inputs. Up to 4 different hold music sources are supported.

Legal Requirements

You must ensure that any MOH source you use complies with copyright, performing rights and other local and national legal requirements.

System Source

The first source is called the System Source. The options for this source are *Tone* (see Default Tone below), *External* (use the input to the AUDIO port on the system) or *WAV File* (a file called *holdmusic.wav* downloaded by the system). This source is numbered source 1.

Alternate Sources

IP Office 4.2+: The names of up to 3 alternate sources can be specified on the System | Telephony | Tones & Music 163 tab. Each can be the name of a wav file or the base extension number of an analog extension port *(IP Office 6.1+)*. These alternate sources are numbered sources 2, 3 and 4.

- The wav files are specified as WAV: followed by the file name. For example WAV:music2.wav for the file music2.wav. Once an alternate wav file source is specified, in situations where the system would download holdmusic.wav it will also attempt to download the alternate wav file.
- An analog extension port is specified as XTN: followed by the base extension number. For example XTN:224 for base extension 224. The analog extension must be one that has had its Equipment Classification (Extension | Analog 254) set to *MOH Source*.

WAV Files

The system can use internal music on hold files that it stores in its non permanent memory. The file properties should be: PCM, 8kHz 16-bit, mono, maximum length 90 seconds on a IP500/IP500v2 system (30 seconds on other systems). If a file of another format is downloaded it will be discarded from memory after the download.

The files, when specified by the System Source and Alternate Source settings) are loaded as follows:

- Following a reboot, the system will try using TFTP to download the file or files.
- The initial source for TFTP download is the system's configured TFTP server (System | System | LAN Settings | TFTP Server IP Address 149). The default for this is a broadcast to the local subnet for any PC running a TFTP server.
- By default Manager acts as a TFTP server while it is running. If Manager is used as the TFTP server then the holdmusic.wav file should be placed in the Manager applications working directory 107.
- Memory Card Usage
  For IP Office 3.1+:
  - If no successful TFTP download occurs, the system will automatically look for the file on the control unit's memory card and will download it from there if found.
  - If an internal music on hold file is downloaded, the system will automatically write a copy of that file to its memory card if present. This will overwrite any existing file of the same name already stored on the card.
  - IP Office 6.1+: For files downloaded from a System SD card. The system will download the file again if the SD card is shutdown and restarted or if files are copied to the card or uploaded using File Manager.

External Inputs

A system can accept external audio inputs from the following sources:

- AUDIO Port
  If the System Source is set to *External*, the audio input to the 3.5mm port marked AUDIO on the back of the control unit is used following a reboot.
- Analog Extension Port *(IP Office 6.1+)*
  The audio input from an analog extension port can also be used as any of the 3 alternate sources. The analog extension must be one that has had its Equipment Classification (Extension | Analog 254) set to *MOH Source*.

Default Tone

If no internal music on hold file is available and *External* is not selected as the System Source, then the system will provide a default tone for music on hold. The tone used is double beep tone (425Hz repeated (0.2/0.2/0.2/3.4) seconds on/off cadence).

- On IP Office 3.0(50)+: This option is only supported for the Italian locale.
- On IP Office 4.0+: This option is supported in all locales.
- On IP Office 4.2+: *Tone* can be selected as the System Source, overriding both the use of the external source port and the downloading of *holdmusic.wav*.

Controlling the Music on Hold Source Used for Calls

Unless specified, the System Source is used for any calls put on hold by system users. For any call, the last source specified for the call is the one used. The following options allow the source to be changed.

- **Hunt Group**
  Each hunt group can specify a Hold Music Source (Hunt Group | Hunt Group 306). That source is then used for calls presented to the hunt group.
- **Incoming Call Route**
  Each incoming call route can specify a Hold Music Source (Incoming Call Route | Standard 344). That source is then used for incoming calls routed by that incoming call route.
- **Short Code** *(IP Office 6.1+)*
  The h character can be used in the Telephone Number field of short codes to specify the hold music to associate with calls routed by that short code. The format h(*X*) is used where *X* is the source number. This method can be used to specify a hold music source for outgoing calls.

Checking Music on Hold

The system short code feature Hold Music 479 can be used to listen to the hold music sources.

- Pre-IP Office 4.2: The default system short code is **34#* which allows you to listen to a system's current music on hold.
- IP Office 4.2+: Dial **34N#*, replacing *N* with the source number 1 (System Source) or 2 to 4 (Alternate Sources).

9.25 System Phone Features

The user option System Phone Rights (User | User 264) can be used to designate a user as being a System Phone user. System Phone users can access a number of additional function not available to other phone users. Note that if the user has a login code set, they will be prompted to enter that code in order to access these features. For pre-IP Office 6.0, the user option System Phone (User | Telephony | Call Settings | System Phone 274) was used.

- None
  The user cannot access any system phone options.
- Level 1
  The user can access all system phone options supported on the type of phone they are using except system management and memory card commands.
- Level 2
  The user can access all system phone options supported on the type of phone they are using including system management and memory card commands. Due to the nature of the additional commands a login code should be set for the user to restrict access.

All IP Offices

The following functions have been supported for System Phone:

- MENU to set date/time
  Restricted to 4412, 4424, 4612, 4624, 6408, 6416 and 6424 phones where supported by the system. Note 4612 and 4624 not support for 4.1+. On these phones, a System Phone user can manually set the system date and time by pressing *Menu* | *Menu* | *Func* | *Setup*.
- SoftConsole Send Message
  If the System Phone user is using SoftConsole, they can access the SoftConsole function Send Message to send a short text message (up to 16 characters) to a display phone. Refer to the SoftConsole documentation for details. Note that this is no longer required for 4.0+.

IP Office 4.1+

The following function is supported for System Phone users on IP Office 4.1+. Not supported on analog, T3, T3 IP and wireless phones.

- Date/Time Programmable Button: *Software level = 4.1+.*
  Allows System Phone users to manually set the system date and time through a programmable button (see Self Administer 651 and Date and Time 727).

IP Office 4.2+

The following function is supported for System Phone users on IP Office 4.2+ (and 4.1 Q4 maintenance release).

- Change Login Code of Other Users
  Using a short code with the Change Login Code 451 feature, system phone users can change the login code of other users on the system.
- Outgoing Call Bar Off
  Using a short code with the Outgoing Call Bar Off 483 feature, system phone users can switch off the outgoing call bar status of other users on the system.

IP Office 5.0+

The following function is supported for System Phone users on IP Office 5.

- Edit System Directory Records
  Using a 1400, 1600 or 9600 Series phone, a system phone user can edit system directory entries stored in the configuration of the system on which they are hosted. They cannot edit LDAP and/or HTTP imported entries.

IP Office 6.0+

The following functions are supported for System Phone users on IP Office 6.0 These commands are only supported using 1400, 1600 and 9600 Series phones.

Due to the nature of the commands a login code should be set for the user to restrict access. The commands are accessed through the Features | Phone User | System Administration menu. For full details refer to the appropriate phone user guide or the Manager Installation manual.

- **System Management** *(IP500/IP500v2 only)*
  Allows the user to invoke a system shutdown command.

- **Memory Card Management**
  Allows the user to shutdown, startup memory cards and to perform actions to move files on and between memory cards.

- **System Alarms** *(IP500v2 only)*
  For certain events the system can display an S on the user's phone to indicate that there is a system alarm. The user can then view the full alarm text in the phone's Status menu. The possible alarms in order of priority from the highest first are:

  1. Memory Card Failure.
  2. Expansion Failure.
  3. Voicemail Failure.
  4. Voicemail Full.

  1. Voicemail Almost Full.
  2. License Key Failure.
  3. System Boot Error.
  4. Corrupt Date/Time.

9.26 The 'No User' User

It is possible to have an extension which has no default associated user. This can occur for a number of reasons:

- The extension has no Base Extension setting associating it with a user who has the same setting as their Extension to indicate that they are the extension's default associated user.
- The extension's default associated user has logged in at another extension. Typically they will be automatically logged back in at their normal extension when they log out the other phone.
- The extension's default associated user cannot be automatically logged in as they are set to Forced Login.

Phones with no current user logged in are associated with the setting of the NoUser user in the system configuration. This user cannot be deleted and their Name and Extension setting cannot be edited. However their other settings can be edited to configure what functions are available at extensions with no currently associated user.

By default the NoUser user has Outgoing Call Bar enabled so that the extension cannot be used for external calls. IP Office 4.0+: The users first programmable button is set to the Login action.

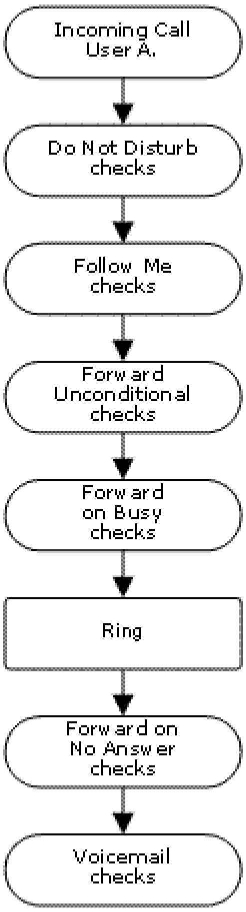
Avaya 1100 Series and 1200 Series, when logged out as No User, the phones are restricted to logging in and dial emergency calls only.

NoUser Source Numbers

The SourceNumbers tab of the NoUser user is used to configure a number of special options. These are then applied to all users on the system. For details refer to the User | Source Numbers 272 section.

9.27 Forwarding Calls

This section contains topics looking at how users can have their calls automatically redirected. As illustrated, there is an order of priority in which the redirect methods are used.



1. Do Not Disturb (DND) 769
   Redirect all calls to voicemail if available, otherwise return busy tone. DND overrides all the redirect method below unless the calling number is in the user's DND Exception Numbers List.
2. Follow Me 771
   Redirect all calls to another extension that the users is temporarily sharing. Follow Me overrides Forward Unconditional. The Follow Me destination is busy or does not answer, the user's Forward on Busy or Forward on No Answer options can be used if set.
3. Forward Unconditional 773
   Redirect the user's external calls to another number. That number can be any number the user can normally dial including external numbers. Forwarding of hunt group and internal calls is optional. Forward Unconditional overrides Forward on Busy and Forward on No Answer.
   - If the destination is an internal user on the same system, they are able to transfer calls back to the user, overriding the Forward Unconditional.
4. Forward on Busy 775
   Redirects the user's external calls when the system sees the user as being busy. Uses the same number as Forward Unconditional unless a separate Forward on Busy Number is set. Forwarding internal calls is optional. Forward on Busy overrides Forward on No Answer.
5. Forward on No Answer 777
   Redirects the user's external calls when they ring for longer than the user's No Answer Time. Uses the same number as Forward Unconditional unless a separate Forward on Busy Number is set. Forwarding internal calls is optional.

Notes

- Retrieving Externally Forwarded Calls
  Where a call is forwarded to an external destination and receives busy or is not answered within the forwarding user's No Answer Time, the system will attempt to retrieve the call. If forwarded on a trunk that does not indicate its state, for example an analog loop start trunks, the call is assumed to have been answered.
- Off-Switch Forwarding Restrictions
  User forwarding is subject to the same restrictions as transferring calls. To bar a user from forwarding calls to an external number, the Inhibit Off-Switch Forward/Transfers 276 (User | Telephony | Supervisor Settings 276) option. To bar all users from forwarding calls to external numbers the Inhibit Off-Switch Forward/Transfers 160 (System | Telephony | Telephony 160) option can be used. See Off-switch Transfer Restrictions 782.
- IP Office 4.0+: When transferring a call to another extension that has forwarding enabled, the type of call being transferred is used. For example, if transferring an external call, if the transfer target has forwarding of external calls enabled then the forward is used.

9.27.1 Do Not Disturb

Summary: Redirect all calls to busy tone or to voicemail if available except those in your DND exceptions list.

Do Not Disturb (DND) is intended for use when the user is present but for some reason does not want to be interrupted. Instead calls are sent to voicemail if available, otherwise they receive busy tone.

- Exceptions
  Specific numbers can be added to the user's Do Not Disturb Exception List. Calls from those numbers override DND. N and X wildcards can be used at the end of exception numbers to match a range of numbers. For external numbers, this uses the incoming caller line ID (ICLID) received with the call.

- Priority
  Enabling DND overrides any Follow Me or forwarding set for the user, except for calls in the user's Do Not Disturb Exception List.

- Phone
  When enabled, the phone can still be used to make calls. An N is displayed on many Avaya phones. When a user has do not disturb in use, their normal extension will give alternate dialtone 846 when off hook.

- Applied to

Call Types Blocked		Call Treatment
Internal	✓	Voicemail if available, otherwise busy tone.
External	✓	Voicemail if available, otherwise busy tone.
Hunt Group	✓	Call not presented (DND exceptions are not used).
Page	✓	Call not presented.
Follow Me	✗	Rings.
Forwarded	✓	Busy.
VM Ringback	✗	Rings
Automatic Callback	✗	Rings
Transfer Return	✗	Rings.
Hold Return	✗	Rings.
Park Return	✗	Rings.
Twinning	✓	Voicemail if available, otherwise busy tone.

Do Not Disturb and Twinning

- Mobile Twinning
  Selecting DND disables mobile twinning.

- Internal Twinning
  - Logging out or setting do not disturb at the primary stops twinned calls alerting at the secondary also.
  - Logging out or setting do not disturb at the secondary only affects the secondary.

- Do Not Disturb Exceptions List
  For both types of twinning, while DND is selected calls from numbers entered in the user's Do Not Disturb Exception List 270 are only presented to the primary phone.

<table>
<tr><th colspan="2">Do Not Disturb</th></tr>
<tr><td>Manager</td><td>A user's DND settings can be viewed and changed through the User | DND 270 tab within the system configuration settings.</td></tr>
<tr><td>Controls</td><td>The following short code features/button programming actions can be used:
<table>
<tr><th>Feature/Action</th><th>Short Code</th><th>Default</th><th>Button</th></tr>
<tr><td>Do Not Disturb On 466</td><td>✓</td><td>*08</td><td>✓ - Toggles.</td></tr>
<tr><td>Do Not Disturb Off 466</td><td>✓</td><td>*09</td><td>✓</td></tr>
<tr><td>Do Not Disturb Exception Add 465</td><td>✓</td><td>*10*N#</td><td>✓</td></tr>
<tr><td>Do Not Disturb Exception Delete 465</td><td>✓</td><td>*11*N#</td><td>✓</td></tr>
<tr><td>Cancel All Forwarding 450</td><td>✓</td><td>*00</td><td>✓</td></tr>
</table></td></tr>
<tr><td>Voicemail</td><td>If voicemail is available, it is used instead of busy tone for callers not in the users exceptions list.

For Voicemail Pro, the Play Configuration Menu action can be used to let callers switch DND on or off.</td></tr>
</table>

<table>
<tr><td>Phone Manager</td><td>Users can enable DND and set exception numbers by clicking and select the Do Not Disturb tab.

When a user has DND enabled, it is indicated by DND in the title bar. It is indicated by a blue cross symbol on Speed Dial icons set to the user.</td></tr>
<tr><td>SoftConsole</td><td>A SoftConsole user can view and edit a user's DND settings except exception numbers. Through the directory, select the required user. Their current status including DND is shown. Double-click on the details to adjust DND on or off.</td></tr>
</table>

9.27.2 Follow Me

Summary: Have your calls redirected to another user's extension, but use your coverage, forwarding and voicemail settings if the call receives busy tone or is not answered.

Follow Me is intended for use when a user is present to answer calls but for some reason is working at another extension such as temporarily sitting at a colleague's desk or in another office or meeting room. Typically you would use Follow Me if you don't have a Hot Desking log in code or if you don't want to interrupt your colleague from also receiving their own calls, ie. multiple users at one phone.

- Priority
  Follow Me is overridden by DND except for callers in the user's DND Exception Numbers List. Follow Me overrides Forward Unconditional but can be followed by the user's Forward on Busy or Forward on No Answer based on the status of the Follow Me destination.
- Destination
  The destination must be an internal user extension number. It cannot be a hunt group extension number or an external number.
- Duration
  The Follow Me user's no answer timeout is used. If this expires, the call either follows their Forward on No Answer setting if applicable, or goes to voicemail is available. Otherwise the call continues to ring at the destination.
- Phone
  When enabled, the phone can still be used to make calls. When a user has follow me in use, their normal extension will give alternate dialtone 846 when off hook.
- Exceptions
  - The Follow Me destination extension can make and transfer calls to the follow me source.
  - The call coverage settings of the user are applied to their Follow Me calls. The call coverage settings of the destination are not applied to Follow Me calls it receives.
- Calls Forwarded

Call Types Redirected		
Internal	✓	Redirected.
External	✓	Redirected.
Hunt Group	✓	Redirected*.
Page	✓	Redirected.
Follow Me	✗	Not redirected.
Forwarded	✓	Redirected.
VM Ringback	✗	Not redirected.
Automatic Callback	✗	Not redirected.
Transfer Return	✗	Not redirected.
Hold Return	✗	Not redirected.
Park Return	✗	Not redirected.

*Except calls for "Longest Waiting" type hunt groups.

Follow Me	
Manager	A user's Follow Me settings can be viewed and changed through the User | Forwarding 281 tab within the system configuration settings. Note that on this tab, entering a Follow Me Number also enables Follow Me.
Controls	The following short code features/button programming actions can be used:
Voicemail	For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination. For Voicemail Pro, the Play Configuration Menu action can be used to let callers alter or set their current Follow Me destination.

Feature/Action	Short Code	Default	Button
Follow Me Here 470	✓	*12*N#	✓
Follow Me Here Cancel 471	✓	*13*N#	✓
Follow Me To 471	✓	*14*N#	✓
Cancel All Forwarding 450	✓	*00	✓

Phone Manager	Users can set a Follow Me To Number and enable Follow Me through the Forwarding tab. Click ▤ and select the Forwarding tab. When a user has Follow Me enabled, it is indicated by FollowTo and the destination in the title bar. It is not indicated on Speed Dial icons set to the user. Other status icons remain linked to the user's normal telephone and not to the status of the destination.
SoftConsole	A SoftConsole user can view and edit a user's Follow Me settings. Through the directory, select the required user. Their current status including Follow Me is shown. Double-click on the details and select Forwarding to alter their forwarding settings including Follow Me.

9.27.3 Forward Unconditional

Summary: Have your calls redirected immediately to another number including any external number that you can dial.

- Priority
  This function is overridden by DND and or Follow Me if applied. Forward Unconditional overrides Forward on Busy and Forward on No Answer.

- Destination
  The destination can be any number that the user can dial. If external and Inhibit Off-Switch Transfers is applied, the caller is directed to voicemail if available, otherwise they receive busy tone.

  - If the destination is an internal user on the same system, they are able to transfer calls back to the user, overriding the Forward Unconditional.

- Duration
  The destination is rung using the forwarding user's No Answer Time. If this expires, the call goes to voicemail if available. Otherwise the call continues to ring at the destination. Calls to an external destination sent on trunks that do not signal their state, for example analog loop start trunks, are assumed to have been answered.

- Phone
  When enabled, the phone can still be used to make calls. An D is displayed on DS phones. When a user has forward unconditional in use, their normal extension will give alternate dialtone 846 when off hook.

- Calls Forwarded
  Once a call has been forwarded to an internal destination, it will ignore any further Forward No Answer or Forward on Busy settings but may follow additional Forward Unconditional settings.

Call Types Forwarded		
Internal	✓	Optional.
External	✓	Forwarded.
Hunt Group	✓	Optional.*
Page	✗	Not presented.
Follow Me	✗	Rings.
Forwarded	✓	Forwarded.
VM Ringback	✗	Rings.
Automatic Callback	✗	Rings.
Transfer Return	✗	Rings.
Hold Return	✗	Ring/hold cycle.
Park Return	✗	Rings.

*Optional only for calls targeting sequential and rotary type groups. Includes internal call to a hunt group regardless of the forward internal setting.

- To Voicemail: *Default = Off. Software level = 5.0+.*
  If selected and forward unconditional is enabled, calls are forwarded to the user's voicemail mailbox. The Forward Number and Forward Hunt Group Calls settings are not used. This option is not available if the system's Voicemail Type is set to *None*. 1400, 1600 and 9600 Series phone users can select this setting through the phone menu. Note that if the user disables forward unconditional, for example using Phone Manager or a short code, the To Voicemail setting is cleared.

<table>
<tr><th colspan="2">Forward Unconditional</th></tr>
<tr><td>Manager</td><td>A user's forwarding settings can be viewed and changed through the User | Forwarding 281 tab within the system configuration settings.</td></tr>
<tr><td>Controls</td><td>The following short code features/button programming actions can be used:
<table>
<tr><th>Feature/Action</th><th>Short Code</th><th>Default</th><th>Button</th></tr>
<tr><td>Forward Number 473</td><td>✓</td><td>*07*N#</td><td>✓</td></tr>
<tr><td>Forward Unconditional On 476</td><td>✓</td><td>*01</td><td>✓ - Toggles.</td></tr>
<tr><td>Forward Unconditional Off 476</td><td>✓</td><td>*02</td><td>✓</td></tr>
<tr><td>Forward Hunt Group Calls On 472</td><td>✓</td><td>✗</td><td>✓ - Toggles.</td></tr>
<tr><td>Forward Hunt Group Calls Off 472</td><td>✓</td><td>✗</td><td>✓</td></tr>
<tr><td>Disable Internal Forwards 463</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Enable Internal Forwards 467</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Disable Internal Forward Unconditional 463</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Enable Internal Forward Unconditional 467</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Set No Answer Time 496</td><td>✓</td><td>✗</td><td>✓</td></tr>
<tr><td>Cancel All Forwarding 450</td><td>✓</td><td>*00</td><td>✓</td></tr>
</table>
</td></tr>
<tr><td>Voicemail</td><td>For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination.

For Voicemail Pro, the Play Configuration Menu action can be used to let callers set their current forwarding destination and switch Forwarding Unconditional on/off.</td></tr>
</table>

Phone Manager	Users can set a forward destination number and enable Forward Unconditional through the Forwarding tab. Click and select the Forwarding tab. When a user has Forward Unconditional enabled, it is indicated by *Fwd unconditional* and the destination in the title bar. It is indicated by a green arrow symbol on Speed Dial icons set to that user.
SoftConsole	A SoftConsole user can view and edit a user's forwarding settings. Through the directory, select the required user. Their current forwarding status is shown. Double-click on the details and select Forwarding to alter their forwarding settings.

9.27.4 Forward on Busy

Summary: Have your calls redirected when you are busy to another number including any external number that you can dial.

The method by which the system determines if a user is 'busy' to calls depends on factors such as whether they have multiple calls appearance buttons or Call Waiting and or Busy on Held set. See Busy 779.

- Priority
  This function is overridden by DND and or Forward Unconditional if applied. It can be applied after a Follow Me attempt. It overrides Forward on No Answer.
- Destination
  The destination can be any number that the user can dial. The Forward Unconditional destination number is used unless a separate number Forward on Busy Number is set. If Inhibit Off-Switch Transfers is applied, the caller is directed to voicemail if available, otherwise they receive busy tone.
- Duration
  The destination is rung using the forwarding user's No Answer Time. If this expires, the call goes to voicemail is available. Calls to an external destination sent on trunks that do not signal their state, for example analog loop start trunks, are assumed to have been answered.
- Phone
  Forward on Busy is not indicated and normal dial tone is used.
- Calls Forwarded
  Once a call has been forwarded to an internal destination, it will ignore any further Forward No Answer or Forward on Busy settings but may follow additional Forward Unconditional settings.

Call Types Forwarded		
Internal	✓	Optional.
External	✓	Forwarded.
Hunt Group	✗	Not presented.
Page	✗	Not presented.
Follow Me	✗	Rings.
Forwarded	✓	Forwarded.
VM Ringback	✗	Rings.
Automatic Callback	✗	Rings.
Transfer Return	✗	Rings.
Hold Return	✗	Ring/hold cycle.
Park Return	✗	Rings.

Forward on Busy	
Software Level	A user's forwarding settings can be viewed and changed through the User | Forwarding 281 tab within the system configuration settings.
Controls	The following short code features/button programming actions can be used:

Feature/Action	Short Code	Default	Button
Forward Number 473	✓	*07*N#	✓
Forward on Busy Number 473	✓	*57*N#	✓
Forward on Busy On 474	✓	*03	✓ - Toggles.
Forward on Busy Off 474	✓	*04	✓
Disable Internal Forwards 463	✓	✗	✗
Enable Internal Forwards 467	✓	✗	✗
Disable Internal Forward Busy or No Answer 464	✓	✗	✗
Enable Internal Forward Busy or No Answer 467	✓	✗	✗
Set No Answer Time 496	✓	✗	✓
Cancel All Forwarding 450	✓	*00	✓

Voicemail	For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination. For Voicemail Pro, the Play Configuration Menu action can be used to let callers set the forward destination.

Phone Manager	Users can set a forward destination number and enable Forward on Busy through the Forwarding tab. Click and select the Forwarding tab. When a user has Forward on Busy enabled, it is indicated by Fwd on Busy and the destination in the title bar. It is not indicated by Speed Dial icons set to that user.
SoftConsole	A SoftConsole user can view and edit a user's forwarding settings. Through the directory, select the required user. Their current forwarding status is shown. Double-click on the details and select Forwarding to alter their forwarding settings.

9.27.5 Forward on No Answer

Summary: Have your calls redirected another number if it rings without being answered.

- Priority
  This function is overridden by DND and Forward on Busy if applied. It can be applied after a Follow Me attempt. Forward Unconditional overrides Forward on Busy and Forward on No Answer.

- Destination
  The destination can be any number that the user can dial. The Forward Unconditional destination number is used unless a separate number Forward on Busy Number is set. If Inhibit Off-Switch Transfers is applied, the caller is directed to voicemail if available, otherwise they receive busy tone.

- Duration
  The destination is rung using the forwarding user's No Answer Time. If this expires, the call goes to voicemail is available. Otherwise the call continues to ring at the destination. Calls to an external destination sent on trunks that do not signal their state, for example analog loop start trunks, are assumed to have been answered.

- Phone
  Forward on No Answer is not indicated and normal dial tone is used.

- Calls Forwarded
  Once a call has been forwarded to an internal destination, it will ignore any further Forward No Answer or Forward on Busy settings but may follow additional Forward Unconditional settings.

Call Types Forwarded		
Internal	✓	Optional.
External	✓	Forwarded.
Hunt Group	✗	Not applicable.
Page	✗	Not applicable.
Follow Me	✗	Rings.
Forwarded	✓	Forwarded.
VM Ringback	✗	Rings.
Automatic Callback	✗	Rings.
Transfer Return	✗	Rings.
Hold Return	✗	Ring/hold cycle.
Park Return	✗	Rings.

Forward on No Answer	
Manager	A user's forwarding settings can be viewed and changed through the User | Forwarding 281 tab within the system configuration settings.
Controls	The following short code features/button programming actions can be used:

Feature/Action	Short Code	Default	Button
Forward Number 473	✓	*07*N#	✓
Forward on Busy Number 473	✓	*57*N#	✓
Forward on No Answer On 475	✓	*05	✓ - Toggles.
Forward on No Answer Off 475	✓	*06	✓
Enable Internal Forwards 467	✓	✗	✗
Disable Internal Forwards 463	✓	✗	✗
Enable Internal Forward Busy or No Answer 467	✓	✗	✗
Disable Internal Forward Busy or No Answer 464	✓	✗	✗
Set No Answer Time 496	✓	✗	✓
Cancel All Forwarding 450	✓	*00	✓

Voicemail	For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination. For Voicemail Pro, the Play Configuration Menu action can be used to let callers set the forward destination. It cannot however be used to enable Forward on Busy or set a separate Forward on Busy number.

Phone Manager	Users can set a forward destination number and enable Forward on No Answer through the Forwarding tab. Click and select the Forwarding tab. When a user has Forward on No Answer enabled, it is indicated by Fwd on No Answer and the destination in the title bar. It is not indicated by Speed Dial icons set to that user.
SoftConsole	A SoftConsole user can view and edit a user's forwarding settings. Through the directory, select the required user. Their current forwarding status is shown. Double-click on the details and select Forwarding to alter their forwarding settings.

9.27.6 Determining a User's Busy Status

Various system features allow users to handle more than one call at a time. Therefore the term "busy" has different meanings. To other users it means whether the user is indicated as being busy. To the system it means whether the user is not able to receive any further calls. The latter is used to trigger 'busy treatment', either using a user's Forward on Busy settings or redirecting calls to voicemail or just returning busy tone.

- Busy Indication - In Use
  The user busy indication provided to programmable buttons and to user applications, is based on the monitored user's hook switch status. Whenever the user is off-hook, they will be indicated as being busy regardless of call waiting or call appearance settings.

- Busy to Further Calls
  Whether a user can receive further calls is based on a number of factors as described below.

  - Logged In and Present
    Is the user logged into an extension and is that extension physically connected to the system.

  - Busy on Held
    If a user enables their Busy on Held setting, whenever they have a call on hold, they are no longer available to any further incoming calls.

  - Appearance Buttons
    A user's call appearance button are used to receive incoming calls. Normally, whilst the user has any free call appearance buttons, they are available to receive further calls. Exceptions are:

    - Reserve Last Appearance
      Users with appearance buttons require a free call appearance button to initiate transfers or conferences. Therefore it is possible through the user's configuration settings to reserve their last call appearance button for outgoing calls only.

    - Other Appearance Buttons
      Calls may also be indicated on line, call coverage and bridged appearance buttons.

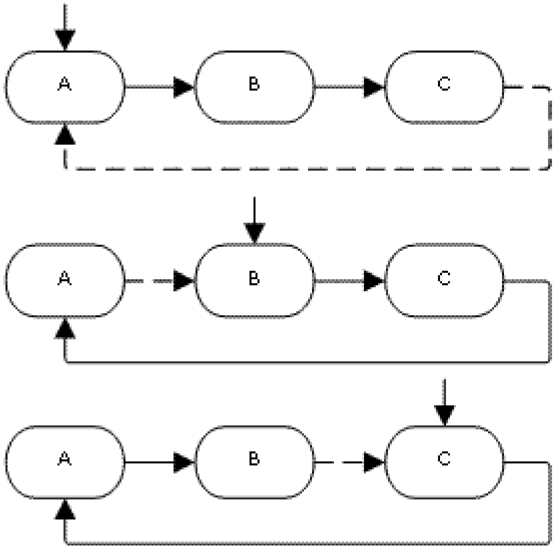
- Call Waiting
  Users of phones without appearance buttons can use call waiting. This adds an audio tone, based on the system locale, when an additional call is waiting to be answered. Only one waiting call is supported, any further calls receive busy treatment.

- Hunt Group Calls
  A user's availability to receive hunt group calls is subject to a range of other factors. See Member Availability325.

9.27.7 Chaining and Loops

Chaining is the process where a call forward to an internal user destination is further forwarded by that user's own forwarding settings.

- **Follow Me Calls**
  Follow Me calls are not chained. They ignore the forwarding, Follow Me and Do Not Disturb settings of the Follow Me destination.
- **Voicemail**
  If the call goes to voicemail, the mailbox of the initial call destination before forwarding is used.
- **Looping**
  When a loop would be created by a forwarding chain, the last forward is not applied. For example the following are scenarios where A forwards to B, B forwards to C and C forwards to A. In each case the final forward is not used as the destination is already in the forwarding chain.



- **Hunt Group Loop**
  If a user forwards a call to a hunt group of which they are a member, the group call is not presented to them but is presented to other members of the hunt group.
- **Maximum Number of Forwards**
  A maximum of 10 forwarding hops are supported for any call.
- **Calls Forwarded**
  Once a call has been forwarded to an internal destination, it will ignore any further **Forward No Answer** or **Forward on Busy** settings but may follow additional **Forward Unconditional** settings.

9.28 Transferring Calls

The following are some of the methods usable to transfer calls.

- **Supervised Transfer**
  This is a transfer where the user waits for the transfer destination to answer and talks to that party before completing the transfer, this is referred to as a consultation call. They then either complete the transfer or drop the call and return to the held for transfer call.
  - **Pre-IP Office 4.0:** The initial consultation stage is presented as an internal call with internal call details and ringing.
  - **IP Office 4.0+:** The call details, display, ringing and forwarding applied are appropriate to the type of call (internal or external) being transferred.
- **Unsupervised Transfer**
  This is a transfer completed whilst the destination is still ringing.
- **Automatic Transfer - Forwarding**
  The system allows users to automatically transfer calls using forwarding options such as Forward on Busy, Forward on No Answer and Forward Unconditional. For full details see DND, Follow Me and Forwarding 768.
- **Transfers to a Forwarded Extension**
  IP Office 4.0+: When transferring a call to another extension that has forwarding enabled, the type of call being transferred is used. For example, if transferring an external call, if the transfer target has forwarding of external calls enabled then the forward is used.
- **Transfer Return Time (secs):** *Default = Blank (Off), Range 1 to 99999 seconds.*
  Sets the delay after which any call transferred by the user, which remains unanswered, should return to the user. A return call will continue ringing and does not follow any forwards or go to voicemail.
  - **Pre-3.2 IP Office:** The transfer return only occurs if the user has no other connected call.
  - **IP Office 3.2+:** Transfer return will occur if the user has an available call appearance button.
  - Transfer return is not applied if the transfer is to a hunt group that has queuing enabled.

Tool	Unsupervised Transfer	Supervised Transfer	Reclaim
Analog Phone	1. Press R. Note that broken dial tone is heard while a call is on hold. 2. Dial the transfer destination number. 3. Hang-up.	1. Press R. 2. Dial the transfer destination number. 3. If the destination answers and accepts the call, hang-up. 4. If the called party does not answer or does not want to accept the call, press R again. 5. To return to the original caller press R.	*46
Avaya DS Phone	1. Press Transfer. 2. Dial the transfer destination number. 3. Press Transfer again to complete the transfer.	1. Press Transfer. 2. Dial the transfer destination number. 3. If the destination answers and accepts the call, press Transfer again to complete the transfer. 4. If the called party does not answer or does not want to accept the call, press Drop. 5. To return to the original caller press it's call appearance button.	*46
Phone Manager	1. Click . 2. Enter the transfer destination in the Number box. 3. Select Blind Transfer button or click .	1. Click . 2. Enter the transfer destination in the Number box. 3. Select Transfer. The original call will be put on Hold. 4. Once the call has been answered you can talk with the transfer target. 5. To transfer the call, click . 6. To cancel the transfer and reconnect the held call press End.	Function | Reclaim

9.28.1 Off-Switch Transfer Restrictions

Users cannot transfer calls to a destination that they cannot normally dial. This applies to manual transfers and also to automatic transfers (forwarding). In addition to call barring applied through short codes, the following system settings may restrict a users ability to transfer calls.

User Specific Controls

- Outgoing Call Bar: *Default = Off (User | Telephony | Supervisor Settings 276)*
  When enabled, this setting stops a user from making any external calls. It therefore stops them making any external transfers or forwards.

- Inhibit Off-Switch Forward/Transfer: *Default = Off (User | Telephony | Supervisor Settings 276). Software level = 3.2+.*
  When enabled, this setting stops the specific user from transferring or forwarding calls externally. This does not stop another user transferring the restricted users calls off-switch on their behalf.

  - When either system or user Inhibit Off-Switch Forward/Transfer is enabled, it affects the operation of the user's phone and applications. In Phone Manager, the destination fields on the Forwarding tab are changed to drop-down lists containing only internal destinations. User attempts to set an external forward destination via a short code will receive error tone. User attempt to set an external forward destination via a programmable button on their phone will not have a Next option allowing the number to be saved.

Line Specific Control

- Analog Trunk to Trunk Connection: *Default = Off (Line | Analog Options 189)*
  When not enabled, users cannot transfer or forward calls on one analog trunk back off-switch using another analog trunk.

System Wide Controls

- Inhibit Off-Switch Forward/Transfer: *Default = Off (System | Telephony | Telephony 160)*
  When enabled, this setting stops any user from transferring or forwarding calls externally.

- Allow Outgoing Transfer: *Default = Off (System | Telephony | Telephony 160). Software level = Pre-4.0.*
  When not enabled, users are only able to transfer or forward incoming external calls back off-switch. For IP Office 4.0 and higher this option is not present, instead outgoing transfers are allowed by default.

- Restrict Network Interconnect: *Default = Off (System | Telephony | Telephony 160). Software level = 4.2+.*
  When this option is enabled, each trunk is provided with a Network Type option that can be configured as either *Public* or *Private*. The system will not allow calls on a Public trunk to be connected to a Private trunk and vice versa, returning busy indication instead.

  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.

Conference Control

Users can use conference controls to effectively transfer calls. This includes transferring an external call to another external number.

- Pre-IP Office 5.0: The off-switch transfer restrictions above are applied.

- For IP Office 5.0+: The use of conferencing to effect off-switch transfers can be restricted using the Inhibit External Only Impromptu Conference 160 setting (System | Telephony | Telephony 160).

9.28.2 Centrex Transfer

IP Office 4.0+: Centrex Transfer is a feature provided by some line providers on external analog lines. It allows the recipient of a calls on such a line to transferred that call to another external number. The transfer is then performed by the line provider and the line is freed. Without Centrex Transfer, transferring an external call to another external number would occupy both a incoming and outgoing line for the duration of the call.

The following are the supported controls and usages for Centrex Transfer:

- Centrex Transfer Button Operation
  The action Flash Hook (Advanced | Miscellaneous | Flash Hook) can be assigned to programmable buttons on DS and IP phones. This button can be configured with or without a telephone number for an automatic or manual transfer respectively.
  - Manual Transfer
    If the programmable button is setup as a Flash Hook button without a target telephone number, pressing the button returns dial tone to the user. They can then dial the required transfer number and when they hear ringing or an answer, hang up to complete the Centrex Transfer.
  - Automatic Transfer
    If the programmable button is setup as a Flash Hook button with a target telephone number, pressing the button performs the Centrex Transfer to the numbers as a single action.
- Centrex Transfer Short Code Operation
  The short code feature Flash Hook can be used with system short codes. It can be setup with or without a telephone number in the same way as a Flash Hook programmable button detailed above. The line group must be the group of analog lines from the Centrex service line provider.
  - Centrex Transfer Operation for Analog Extensions
    Most analog phones have a button that performs the action of sending a hook flash signal. The marking of the button will vary and for example may be any of R, H, Recall or Hold. For system analog extensions, pressing this button sends a hook flash to the system to hold any current call and return dial tone.
    - To perform a Centrex Transfer, pressing the analog extension's hook flash button should be followed by the dialing of a Flash Hook short code.
    - For analog extension users with Call Waiting enabled, pressing the hook flash button during a call will hold the current call and connect any call waiting. Therefore it is recommend that analog extension users wanting to use Centrex Transfer should not also have Call Waiting enabled.
  - Transfer from Voicemail/Auto-Attendant
    This operation is only supported through Voicemail Pro using an Assisted Transfer action with the destination set to a Flash Hook short code.

Additional Notes

- Addition Prefix Dialing
  In some cases the Centrex service provider may require a prefix for the transfer number. If that is the case, that prefix must be inserted in the button programming or the short code used for the Centrex Transfer.
- Phone Manager and SoftConsole
  Centrex Transfer is not supported for calls being held and transferred through applications such as Phone Manager and SoftConsole.
- Conference Calls
  Centrex Transfer is not supported with conference calls.

9.28.3 Handsfree Announced Transfers

IP Office 4.2 Q4 2008 maintenance release+: This feature allows the enquiry call part of a supervised transfer to be answered handsfree. In addition the system can be optionally configured to allow both the enquiry call and completed transfer call to be auto-answered.

Example

1. User 201 answers a call that they then want to transfer to user 203.
2. They press TRANSFER to put the call on hold pending transfer.
3. They then press a Dial Direct button and dial 203.
4. The transfer enquiry call is auto answered by User 203's phone. User 201 is able to announce the pending transfer and hear if User 203 wants to accept the call.
    - The auto-answer only occurs if the target user's extension is idle. If the target is already connected to a call, the transfer enquiry will be presented as normal call.
5. If the transfer is accepted, User 201 can press TRANSFER again to complete the transfer process.
    - The transferred call will then ring at the target. However, if required the system can be configured to also auto-answer the completed transfer.

Configuration

Handsfree announced transfers are supported when using one of the following features after having pressed TRANSFER.

Button Features	Short Code Features
• Dial Direct 606 • Automatic Intercom 581 • Dial Intercom 608	• Dial Direct 457

- User Button Usage
  Following the use of any of the buttons above, if the button has not been programmed with a specific target, a User 666 button can be used to indicate the target for the enquiry call. This gives the advantage of being able to see the target user's status before attempting the transfer.
    - For Automatic Intercom and Dial Intercom buttons without a pre-specified target, the User button must be on a button module.
    - For Dial Direct buttons without a pre-specified target, the User button can be on the phone or button module. Due to this and the support for Dial Direct across an Small Community Network, we recommend that a Dial Direct button is used for handsfree announced transfers.

Phone Support

Handsfree announced transfer is supported for calls being transferred to the following phones:

Full Support	Partial Support	Not Supported
The following system phones support full announced transfer operation. • 1603, 1608, 1616, 2410, 2420, 5410, 5420, 4610, 4621, 4625, 5610, 5620, 5621. • All T3 phones. • Analog Off-Hook Stations *(See notes below)*.	The following phone can auto-answer announced transfers but require the user to use the handset to respond. • 2402, 4601, 4602, 5402, 5601, 5602.	Announced transfer is not supported for any phones not listed in the other column. On unsupported phones the transfer eqnuiry consultation call will be presented as a normal call.

Notes

- On supported phones, if the target user's phone is not idle when the enquiry call attempt is made, the enquiry call is turned into a normal transfer attempt, eg. alerting on an available call appearance.
- Enabling the extension specific setting Disable Speakerphone 252 will turn all auto-answer calls, including handsfree announced transfers to the extension, into normal calls.

- **Off-Hook Station Analog Phones**
  Analog phone extensions configured as Off-Hook Station 275 can auto-answer transfers when off-hook and idle.

- **Headset Users**
  The following applies to users on supported phones with a dedicated HEADSET button. These users, when in headset mode and idle will auto-answer the announced transfer enquiry call through the headset after hearing 3 beeps. The transfer completion will require them to press the appropriate call appearance unless they are set to Headset Force Feed 637.

- **Twinning**
  Handsfree announced transfer calls to users with twinning enabled will be turned into normal calls.

- **Small Community Network Support**
  Dial Direct is supported to targets across a Small Community Network, therefore allowing handsfree announced transfers to remote SCN users.

Full Handsfree Transfer Operation

If required the system can be configured to allow the full handsfree announced transfer process, ie. both the enquiry call and the transfer, to be auto-answered on supported phones. This is done by entering *FORCE_HANDSFREE_TRANSFER* into the Source Numbers 272 of the *NoUser* 767 user 767 and rebooting the system

9.28.4 One Touch Transferring

IP Office 5 Q4 2009 maintenance release+: This feature allows selected users to transfer calls to each other using a reduced number of key presses.

- With this option, a call can be transferred by simply selecting the transfer destination and then hanging up (or pressing Transfer if working handsfree).
- Without this option the normal sequence is to press Transfer, dial the destination and then hanging up (or pressing Transfer if working handsfree).

For one touch transfer the transfer destination number must be selected using a button programmed to one of the following features:

- User
- Dial
- Abbreviated Dial
- Automatic Intercom
- Dial Intercom
- Dial Direct

This feature is enabled on a per user basis by adding *Enable_OTT* to the user's Source Number settings. This feature is supported on all Avaya phones that support the programmable button features above except T3 phones.

9.29 Conferencing

IP Office systems support the following conference capabilities:

Control Unit	DSP		Conference Capability
Small Office Edition	✓	24	Supports up to 24 conference parties with a maximum of 6 parties in any particular conference.
IP403	✗	64	Support multiple conferences totaling up to 64 parties. For example: • 20 x 3-way conferences plus 1 x 4-way conference. • 1 x 10-way conference (10 parties) plus 11 x 3-way conferences (33 parties) and free capacity for 20 more conference parties to join new or existing conferences.
IP406 V1	✗	64	
IP406 V2	✓	64	
IP500/IP500v2	✓	64	
		128	For IP Office 5+, the IP500 capacity has been increased to 128. However, the maximum of 64 parties still applies to each individual conference. The system capacity is reduced by the number of embedded voicemail channels is use.
IP412	✗	128	The IP412 supports two 64 party conference banks. When a new conference is started, the bank with the most free capacity is used for that conference. However, once a conference is started on one conference bank, that conference cannot use any free capacity from the other conference bank.

Notes:

- **DSP**
  The DSP column in the table above indicates systems that automatically apply silence suppression to quiet parties in conferences involving 10 or more parties.

- **Other Uses of Conference Resources**
  System features such as call intrusion, call recording and silent monitoring all use conference resources for their operation. On IP500/IP500v2 systems, each embedded voicemail call in progress also reduces the conference capacity.

- **Automatically Ending Conferences**
  The behavior for the system automatically ending a conference varies as follows:

  - Pre-IP Office 4.0: If a conference has just two parties, and one party leaves, the conference call is ended. This may affect conferences that are just beginning but currently only contain the first two parties to join.

  - IP Office 4.0+: A conference remains active until the last extension or trunk with reliable disconnect leaves. Connections to voicemail or a trunk without reliable disconnect (for example an analog loop-start trunk) will not hold a conference open.

  - For IP Office 5.0+: The Drop External Only Impromptu Conference setting controls whether a conference is automatically ended when the last internal party exits the conference.

- **Analog Trunk Restriction**
  In conferences that include external calls, only a maximum of two analog trunk calls are supported. This limit is not enforced by the system software.

- **Recording Conferences**
  If call recording is supported, conference calls can be recorded just like normal calls. Note however that recording is automatically stopped when a new party joins the conference and must be restarted manually. This is to stop parties being added to a conference after any "advice of recording" message has been played.

- **Conferencing Center**
  If Conferencing Center is installed, 5 conference resources are reserved for use by the system. The maximum number of callers in any one conference and the total number of people on conference calls is reduced by 5.

- **IP Trunks and Extensions**
  Conferencing is performed by services on the system's non-IP interface. Therefore a voice compression channel is required for each IP trunk or extension involved in the conference.

- **Call Routing**
  A short code routing calls into a conference can be used as an Incoming Call Route destination.

- **Conference Tones**
  The system provides conference tones. These will be either played when a party enters/leaves the conference or as a regularly repeated tone. This is controlled by the Conferencing Tone 163 (System | Telephony | Tones & Music 163) option.

9.29.1 Default Conference Handling

The methods below use the system's default system short codes.

To start/add to a conference:

1. Place your first call or the existing conference on hold. Existing conference parties will still be able to talk to each other.
2. Call the new party.
   - If not answered, or diverted to voicemail, or answered but the party does not want to join the conference; put them on hold and dial *52 to clear the call.
3. If answered and the other party wants to join the conference, put them on hold and dial *47.
4. All held calls are now in conference.
   - Digital display extensions will see CONF followed by the conference number.

To exit a conference:

1. Parties wanting to leave a conference can simply hang-up.

9.29.2 Using Conference Meet Me

Each conference on the system is assigned a conference number. This number is displayed on suitable display phones.

Conference Meet Me allows users to join or start a specific numbered conference. This method of operation allows you to advertise a conference number and then let the individual parties join the conference themselves.

Through the Button Programming tab within Manager, the Conference Meet Me function can be assigned to a DSS key (select *Advanced | Call | Conference Meet Me*). This allows simple one key access by internal users to specific conferences.

For IP Office 4.1+ the following enhancements apply to use buttons programmed to Conference Meet Me:

- When the conference is active, any buttons associated with the conference ID indicate that active state.
- Calls can be transferred to the conference by pressing Transfer and then pressing the Conference Meet Me button.

Note:

- Conference Meet Me can create conferences that include only one or two parties. These are still conferences using slots from the system's conference capacity.
- For pre-IP Office 5 systems, this function is not supported on IP500 systems without an *IP500 Upgrade Standard to Professional* license.

Example 1: Meet Me to a user specified conference
The following example system short code allows any extension to dial *67* and then the number of the conference which they want to join followed by #. For example dialing *67*600# will put the user into conference 600.

- Short Code: *67*N#
- Telephone Number: N
- Feature: Conference Meet Me

Example 2: Meet Me to a system specified conference number
The following example system short code allows the dialing extension to join a specific conference, in this case 500.

- Short Code: *500
- Telephone Number: 500
- Feature: Conference Meet Me

If you are asked to add a party to a conference, having a conference meet me short code is very useful. With the conference in progress, call the new party. When they answer, hold the call, dial the conference meet me short code and then hang-up.

9.30 Hot Desking

Hot desking allows users to log in at another phone. Their incoming calls are rerouted to that phone and their user settings are applied to that phone. There are a number of setting and features which affect logging in and out of system phones.

- In order to hot desk, a user must be assigned a Login Code 276 (*User | Telephony | Supervisor Settings* 276) in the system configuration.
- By default, each system extension has an Base Extension setting. This associates the extension with the user who has the matching Extension settings as being that extension's default associated user.
  - By leaving the Base Extension setting for an extension blank, it is possible to have an extension with no default associated user. All extensions in this state use the settings of a special user named *NoUser*. On suitable phones the display may show *NoUser*.
  - You can create users whose Extension directory number is not associated with any physical extension. These users must have a log in code in order to log in at a phone when they need to make or receive calls. In this way the system can support more users than it has physical extensions.
- When another user logs in at an extension, they take control of that phone. Any existing user, including the default associated user, is logged out that phone.
  - Any user settings not applicable to the type of phone on which the user has logged in become inaccessible. For example some programmable button features will become inaccessible if the phone at which a user logs in does not have a sufficient number of programmable buttons.
  - Note that settings that are stored by the phone rather than by the system remain with the phone and do not move when a user hot desks.
- When a user logs off or is logged out by someone else logging in, they are automatically logged back in at the extension for which they are the default associated user if no one else is logged in at that extension. However this does not happen for users set to Forced Login (*User | Telephony | Supervisor Settings* 276).
- For each user, you can configure how long the extension at which they are logged in can remain idle before they are automatically logged out. This is done using the Login Idle Period option. This option should only be used in conjunction with Force Login.
- Logged in users who are members of a hunt group can be automatically logged out if they do not answer hunt group calls presented to them. This is done by selecting *Logged Off* as the user's Status on No Answer (*User | Telephony | Supervisor Settings* 276) setting.
- Calls to a logged out user are treated as if the user is busy until the user logs in.
- Logging in and out at a phone can be done either using system short codes or programmable buttons.
  - The default system short code for logging in, is **35*N#* where the user replaces N with their extension number and then log in code separated by a *. This uses the short code feature ExtnLogin 468. If the user dials just a log in code as N, it is checked against the user with the same extension number as the extension's base extension number.
  - The default system short code for logging out is **36*. This uses the short code feature ExtLogout 468. For IP Office 4.0 and higher this feature cannot be used by a user who does not have a log in code or by the default associated user of an extension unless they are set to forced log in.
  - The ExtnLogin 619 and ExtnLogout 619 features can be assigned to programmable buttons on suitable Avaya phones. The ExtnLogin button will then prompt the user to enter their details.

9.30.1 Remote Hot Desking

IP Office 4.0+ supports hot desking 789 between systems within a Small Community Network 810. In the descriptions below, the system on which the user is configured is termed their 'home' system, all other systems are 'remote' systems. For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses.

When a user logs in to a remote system:

- The user's incoming calls are rerouted across the SCN.
- The user's outgoing calls uses the settings of the remote system.
- The user's own settings are transferred. However some settings may become unusable or may operate differently.
  - User rights are not transferred to the remote system but the name of any user rights associated with the user are transferred. If user rights with the same name exist on the remote system, then they will be used. The same applies for user rights applied by time profiles, if time profiles with the same name exist on the remote system .
  - Appearance buttons configured for users on the home system will no longer operate.
  - Various other settings may either no longer work or may work differently depending on the configuration of the remote system at which the user has logged in. For example T3 phones the personal directory is not transferred with the user.
  - For IP Office Release 6.0+, the rights granted to the user by their **Profile** settings are retained by the user. There is no requirement for the remote system to have the appropriate licenses for the **Profile**.
- If the user's home system is disconnected from the SCN while the user is remotely hot desked, the user will remain remotely hot desked. They can remain in that state for up to 24 hours before being automatically removed from the remote system if their home system has not reconnected to the SCN. Note also that when the user's home system is reconnected to the SCN, the user may be automatically logged on that system.

- **Break Out Dialing**
  In some scenarios a hot desking user logged in at a remote system will want to dial a number using the system short codes of another system. This can be done using either short codes with the **Break Out** feature or a programmable button set to **Break Out**. This feature can be used by any user within the Small Community Network but is of most use to remote hot deskers.

9.30.2 Call Center Agents

On systems with a call center application such as Compact Contact Center (CCC) or Compact Business Center (CBC), logging in and logging out is a key part of tracking and reporting on call center agents. It also controls call distribution as, until the agent logs in, their hunt group membership is seen as disabled.

For CCC, CBC and Delta Server, an agent is defined as being a user with a Login Code and set to Forced Login. Those users consume a CCC agent license.

9.30.3 Hot Desking Examples

The following are example of different ways that the hot desking settings can be used.

Scenario 1: Occasional Hot Desking

In this scenario, a particular user, for this example extension 204, needs to occasionally work at other locations within the building.

1. A Login Code is added to the user's configuration settings, for this example 1234.
2. The user can now log in when needed at any other phone by dialing *35*204*1234#. The phone's default associated user is logged out by this and their calls get busy treatment. User 204 is also logged out their normal phone and their calls now rerouted to the phone at which they have logged in.
3. When finished, the user can dial *36 to log out.
4. This logs the phone's normal default user back on. Its also logs the hot desking user back on at their normal extension.

Scenario 2: Regular Hot Desking

This scenario is very similar to the one above. However the user doesn't want to be automatically logged back in on their normal phone until they return to its location.

1. A Login Code is added to the user's configuration settings, for this example 1234.
2. The Forced Login option is selected.
3. When the user logs out of the phone that they are currently using, they are no longer automatically logged in on their normal extension. When they return to it they must dial *35*204*1234# to log in.
4. Whilst not logged in anywhere, calls to the user receive busy treatment.

Scenario 3: Full Hot Desking

Similar to the scenarios above but this time the user doesn't have a regular phone extension that they use. In order to make and receive calls they must find a phone at which they can log in.

1. The user is given an Extension directory number that is not matched by the extension directory number setting of any existing extension.
2. They are also given a Login Code and a Login Idle Period is set, for this example 3600 seconds (an hour). Forced Login isn't required as the user has no default extension at which they might be automatically logged in by the system.
3. The user can now log in at any available phone when needed.
4. If at the end of the business day they forget to log out, the Login Idle Period will eventually log them off automatically.

Scenario 4: Call Center Hot Desking

In this scenario, the phone extensions have no default extension number. Several phones set like this might be used in a call center where the agents use whichever desk is available at the start of their shift. Alternatively a set of desks with such phones might be provided for staff that are normally on the road but occasionally return to the office and need a temporary desk area to complete paper work.

1. For the extensions, the Extension setting is left blank. This means that those phones will be associated with the NoUser user's settings and display NOT LOGGED ON.
2. The call center agents or road-warrior users are configured with Extension directory numbers that also don't match any existing physical extensions. They are all given Login Code numbers.
3. The users can log in at any of the extensions when required. When they log out or log in elsewhere, the extensions return to the NoUser setting.

9.30.4 Automatic Log Off

Normally a user can either log themselves out or be logged out by another user logging in. The follow methods can be used by the system to automatically log out a user. A remote hot desking user whose home system can no longer be seen by the remote system at which they are logged in is automatically logged out after 24 hours.

The following methods only apply to users with a **Login Code** and set to **Forced Login**.

Idle Timeout

The user Login Idle Period 276 (User | Telephony | Supervisor Settings 276) can be used to automatically log out the user after a set period of phone inactivity. The period can be set between 1 to 99999 seconds and is based on call inactivity other than ringing calls.

Unanswered Calls

Users who are members of hunt groups are presented with hunt group calls when they are logged in and not already on a call. If the user is logged in but not actually present they will continue to be presented with hunt group calls. In this scenario it can be useful to log the user off.

1. **For the hunt group**
   On the Hunt Group | Hunt Group tab, use the Agent's Status on No Answer Applies to setting to select which types of unanswered hunt group calls should change the user's status. The options are:
   - None.
   - Any Calls.
   - External Inbound Calls Only.
2. **For the user**
   The Status on No Answer 276 (User | Telephony | Supervisor Settings 276) can be used. This sets what the user's status should be changed to if they do not answer a hunt group call. The options are:
   - **Logged In**
     If this option is selected, the user's status is not changed.
   - **Busy Wrap-Up**
     If this option is selected the user's membership status of the hunt group triggering the action is changed to disabled. The user can still make and receive calls and will still continue to receive calls from other hunt groups to which they belong.
   - **Busy Not Available**
     If this option is selected the user's status is changed to do not disturb. This is the equivalent of DND and will affect all calls to the user.
   - **Logged Off**
     If this option is selected the users status is changed to logged out. In that state the cannot make calls and cannot receive calls. Hunt group calls go to the next available agent and personal calls treat the user as being busy.

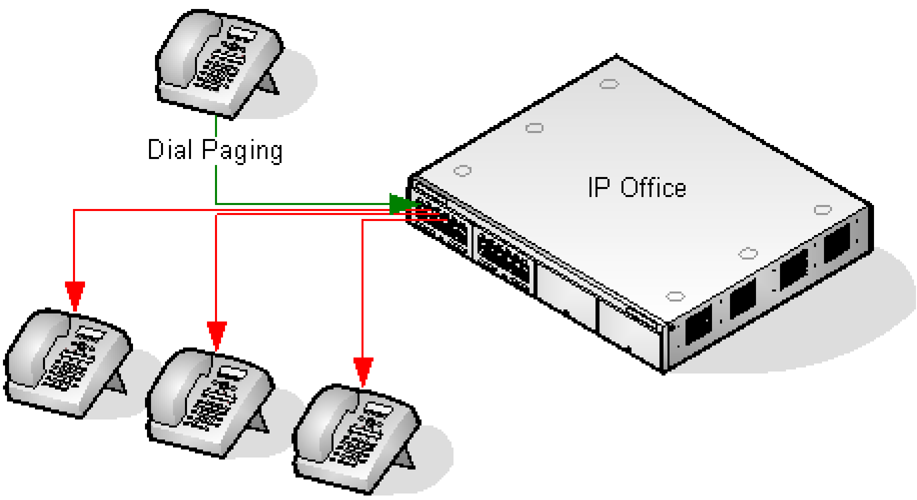
9.31 Paging

Paging Scenario	Paged Device Connects to...	Short Code / Button Feature
Phone to Phone 793 Simple paging to other system extensions.	Digital Station and Avaya H323 Phones	Dial Paging
Mixed Paging 794 This refers to simultaneous paging to phones and a paging speaker.	Analog Extension (Paging Speaker)	Dial Paging
Paging Interface Device 794 This refers to paging to a paging interface device such as a UPAM.	Analog Extension (IVR Port)	Dial Extn
	Analog Trunk	Dial

Paging Limits
The table below lists the maximum recommended size for paging groups.

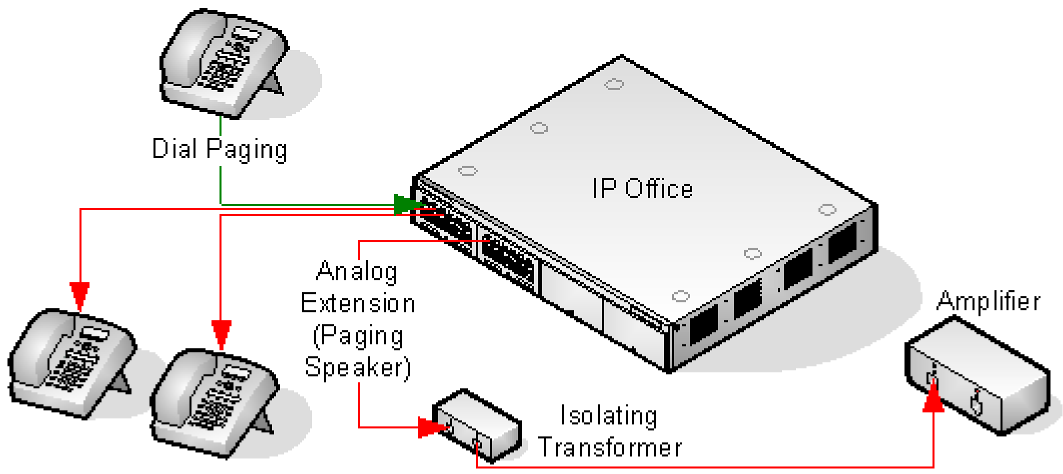
Control Unit	Software Level	
	4.0+	3.2
IP500/IP500v2	64	–
IP412	48	24
IP406 V2	32	16
Small Office Edition	16	16
IP406 V1	–	16
IP403	–	16

Phone to Phone Paging



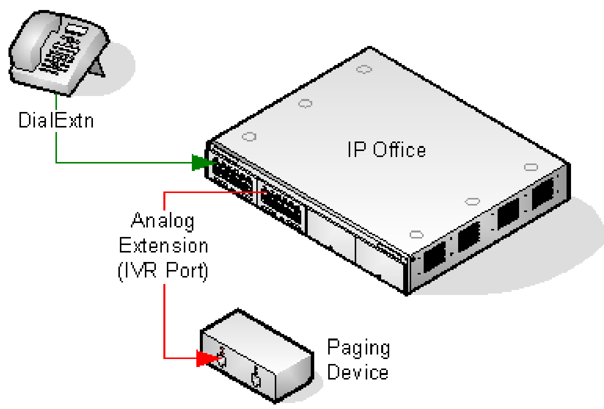
- Paging is supported from all phone types. A page call can be to a single phone or a group of phones.
  - From analog and non-Avaya phones, use a Dial Paging 459 short code.
  - From Avaya feature phones, a programmable button set to Dial Paging 609 can be used.
- Paging is only supported to Avaya phones that support auto answer.
- The page is not heard on phones that are active on another call.
- The page is not heard on phones where the user is set to Do Not Disturb or has Forward Unconditional active.
- On Avaya phones with a dedicated Conference button, the user can answer a page call by pressing that button. This turns the page into a normal call with the pager.

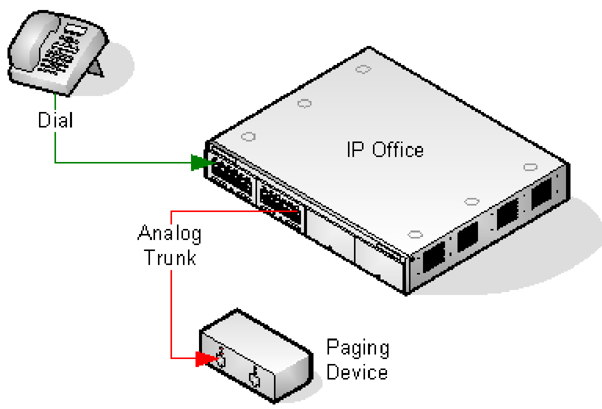
Mixed Paging



- Uses an amplifier connected to an analog extension port via a 600ohm isolating transformer. Some amplifiers include an integral transformer.
  - Avaya/Lucent branded amplifiers are designed for connection to special paging output ports not provided on systems. They are not suitable for supporting mixed paging.
- The transformer and amplifier must be connected when the system is restarted.
- If background music is required between pages, the amplifier must support a separate background music connection and VOX switching.
- The analog extension port is set as a Paging Speaker 254 in the system configuration (Extn | Analog | Equipment Classification) 254.
- Short code/programmable button: Use DialPaging 609.

Paging Interface Device





- Uses a paging interface device such as a UPAM or amplifier with analog trunk/extension interface.
- The device can be connected to an analog trunk port or analog extension port.
- If connected to a trunk port:
  - Short code: Use Dial 455 and the same Line Group ID as the Outgoing Line ID set for the analog trunk.
- If connected to an extension port:
  - Set the analog extension as an IVR Port 254 in the system configuration (Extn | Analog | Equipment Classification 254).
  - Short code/programmable button: Use DialExtn 458.

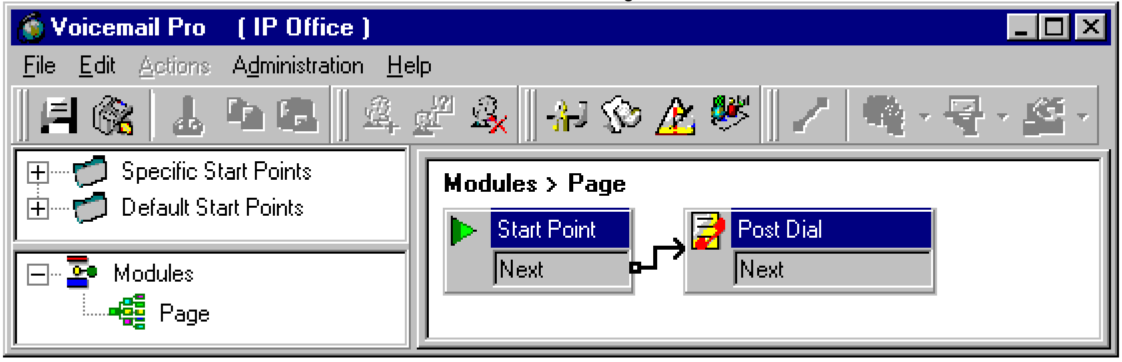
9.31.1 Paging Via Voicemail Pro

Voicemail Pro can be used to deliver pre-recorded announcements. This can be useful when the same announcement is repeated frequently. This method requires the paging port to be an analog extension.

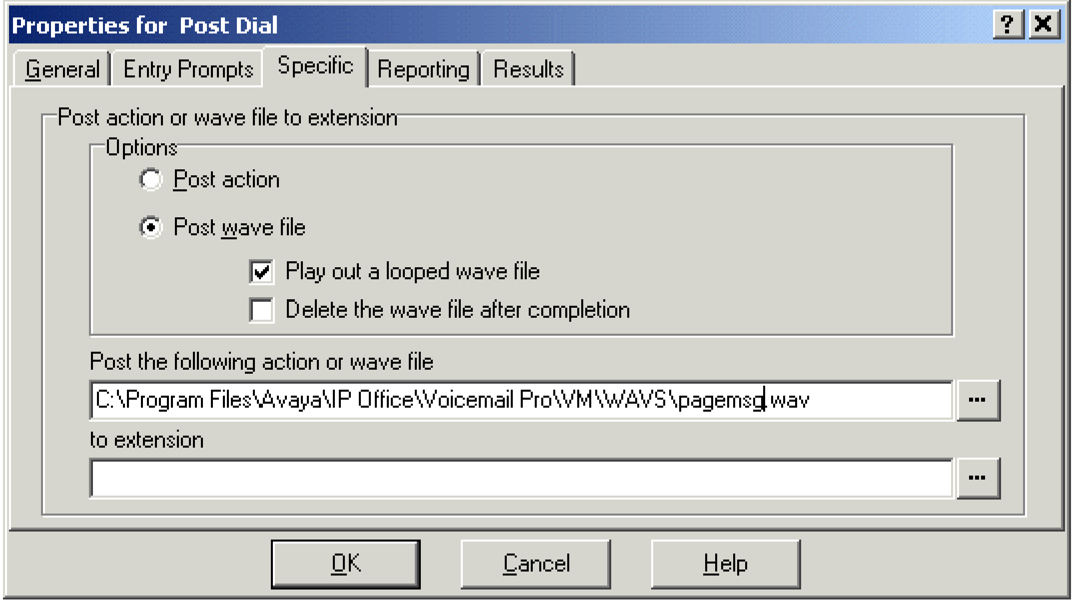
This method also removes the feedback loop that can occur on some sites as the page is first recorded and then played.

Example 1

1. In Voicemail Pro, a new Module was added and named Page.



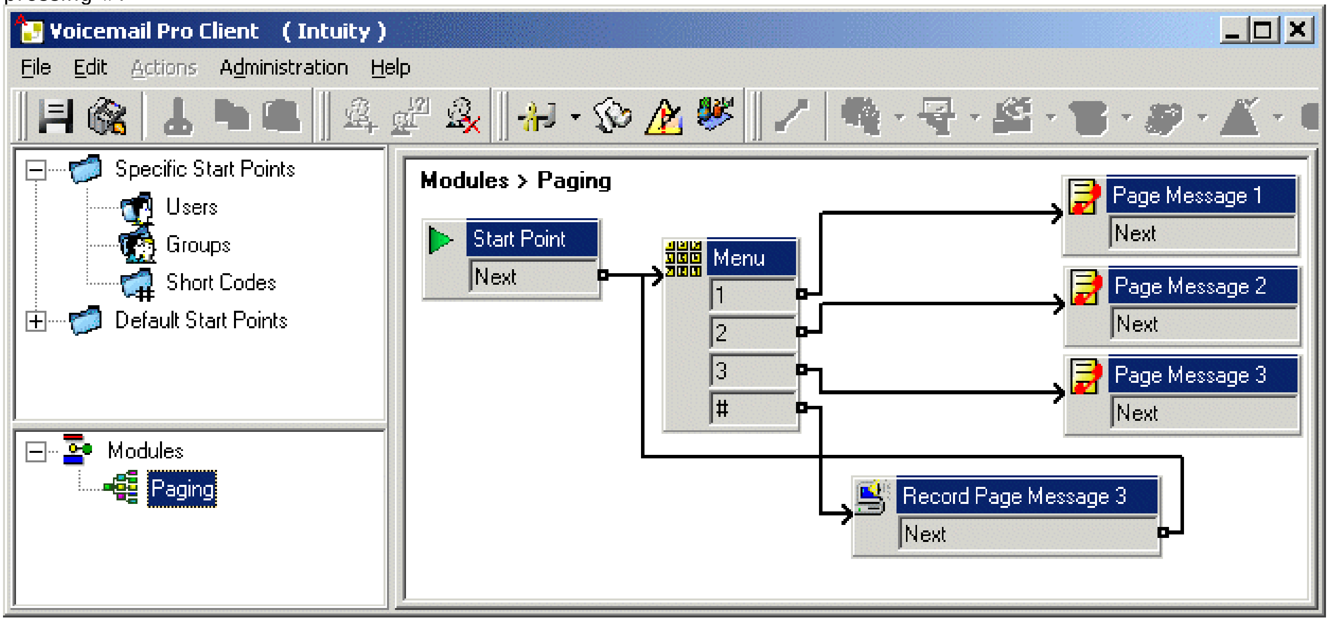
2. A Post Dial action was added to the module. The properties of the Specific tab were set as shown:



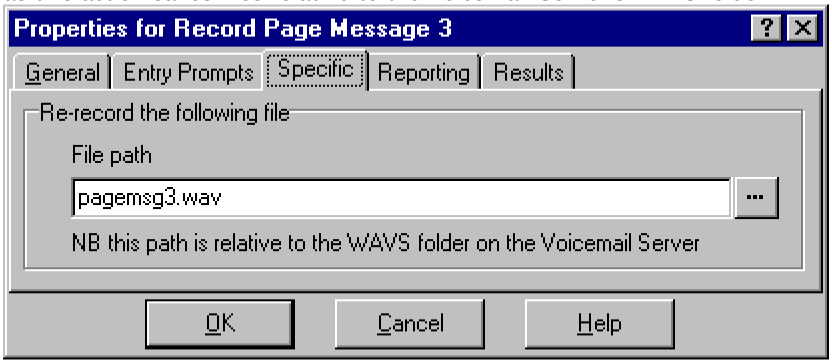
3. We then saved and made live the new Voicemail Pro call flow.
4. In Manager we received the system configuration and created a new short code.
    - Short Code: *80
    - Telephone Number : "Page"
    - Feature: VoicemailCollect.
5. The new system configuration was then merged.

Example 2

This example builds on example 1 by allowing the user to select which message is played from a menu. In this example the user can press 1, 2 or 3 for different messages. They can also re-record the message associated with option 3 by pressing #.



A Play List action was added and in this example set to record pagemsg3.wav. Note that just the file name was specified as this action saves files relative to the Voicemail Server's WAVS folder.



In the Post Dial action that plays back pagemsg3.wav note that the full file path needs to be used.

In Manager, we then added a short code that triggers the module "Paging" using the VoicemailCollect feature.

Chapter 10.
Data Routing

10. Data Routing

The system is a network router. In this role it can connect users on its LAN to remote services by using WAN links and telephone trunk connections. It can also allow users to dial-in and then act as if they were using a PC on the LAN.

As well as being a network router, the system is a telephone system. These dual roles allow it to support a range of functions that involve traffic between the network and telephony interfaces. These functions use internal data channels. The number of internal data channels that can be connected from the system's LAN interface to its telephony interface at any time is restricted.

- An internal data channel is a connection between the system's telephony and LAN interfaces. For example a Voicemail connection, an internet connection or a RAS user.
- Calls using a VCM channel do not use a data channel.
- The number of data channels in use does not necessarily match the number of users:
  - Several LAN network users, browsing the internet using the same service to an ISP would be a single data channel.
  - Several dial-in network users would each have a separate data channel.
- The maximum number of data channels that can be simultaneously in use for voicemail is restricted. These channels also require entry of an appropriate license.

The restriction depends on the type of control unit being used.

System Control Unit	Internal Data Channels	Maximum Data Channels for Voicemail
Small Office Edition	18	10
IP403	18	10
IP406 V1	24	20
IP406 V2	40	20
IP412	100	30
IP500	48[1]	40[2]
IP500v2	48[1]	40[2]

1. Reduced to 44 when an IP500 4-Port Expansion card is installed.
2. Increased from 30 channels for IP Office 5+.

10.1 Network Address Translation (NAT)

NAT allows the addresses used within your LAN to be replaced by a different address when connecting to an external service.

Typically a service provider will allocate you a single IP address to be used when connecting to their service. NAT allows all your user's traffic to appears to be coming from that single address without having to change any of your user's real addresses. This is useful as internally most networks use addresses that have been reserved for public use within networks but are not valid for routing across the internet (since the same addresses may be being used on other networks). Also as stated it allows multiple users to use the same service simultaneously.

The use of NAT is automatically enabled if the system Service being used includes an IP address that is not in the same domain as the its LAN1 IP address.

An exception to the above applies for systems with two LAN's, LAN1 and LAN2. For these units, on each LAN, **Enable NAT** can be selected and then applied to traffic between the two LAN's.

10.2 DHCP

The system can act as a simple DHCP server. When switched on with a defaulted configuration, the Control Unit request IP address information from a DHCP server. If it gets no response it assumes the role of DHCP server for the LAN.

In DHCP Server mode, by default the Control Unit issues itself the address 192.168.42.1. It allocates 200 addresses for DHCP clients, 192.168.42.1 to 19.168.42.200. This leaves 192.168.42.201 to 192.168.42.254 available for any computers that need to be allocated a fixed or static IP address. 192.168.42.255 is not used as this is a broadcast address for the LAN.

10.3 Examples

10.3.1 Simple ISDN Internet Connection

In this example, we want all non-local data traffic to be routed to the Internet. The Internet Service Provider (ISP) has provided the account details required. Using the system's Network Address Translation 798 (NAT), a single account can be used for all users.

1. Select  Service and add a normal service. Change the following settings and click OK.
   - Name: Internet
   - Account Name: As provided by the ISP.
   - Password: As provided by the ISP.
   - Telephone Number: As provided by the ISP.
   - Check Request DNS.
2. Select  IP Route and add a new route. Change the following settings and click OK.
   1. Leave the IP Address and IP Mask blank. This will then match any data traffic that isn't matched by any other IP Route entry.
   2. Select the service created above as the Destination.

Alternate
In the example above, a default IP Route was created which then routed all traffic to the required Service. An alternate method to do this with system is to select Default Route within the Service settings.

10.3.2 ISDN Link Between IP Offices

To create a data link between two sites via ISDN configure the Control Unit as per the following example:

At Site A on IP address 192.168.43.1

1. Create a Normal Service:
   The Service name can be any text and is used to identify this particular Service. The Account Name and password are presented to the remote end, therefore must match the User name and password configured at Site B. The Telephone Number is the number of the remote end.
2. Create an IP Route:
   In the IP Address field enter the network address of the remote end, not the IP address of the Control Unit. Under Destination select the Service created above.
3. Create a User:
   Under the Dial In tab tick Dial In On. This User account is used to authenticate the connection from the Site B. Note that as the Service and User have the same names, these two configuration forms are automatically linked and become an Intranet Service. The User password is displayed at the bottom of the Service tab as the Incoming Password.
4. Setup RAS:
   Check the default RAS settings "Dial In" are available, otherwise create a new one. If the RAS settings are given the same name as the Service and User they are automatically linked and become a WAN Service. Ensure that the Encrypted Password option is not checked when using a WAN Service.
5. Setup an Incoming Call Route:
   Check the default Incoming Call Route is available, otherwise create a new one. If the Incoming Number is left blank, the Incoming Call Route accepts data calls on any number. Under Destination select the RAS service created above. The Bearer Capability should be AnyData.

At Site B on IP address 192.168.45.1

1. Repeat the above process but altering the details to create an route from Site B to Site A.

10.3.3 Using a Dedicated T1/PRI ISP Link

This section shows an example of a dedicated WAN PPP link to an Internet Service Provider (ISP) over a set of T1 or T1 PRI line channels. The ISP must support this mode of connection and will need to provide details of the required settings. If multiple channels are to be used, then the ISP must support Multilink PPP.

1. Create a New WAN Service
   A service is used to define connection settings such as name, password, bandwidth, etc.
   1. Select Service to display the existing services.
   2. Click on and select WAN Service.
   3. Select the Service tab.
      1. In the Name field enter an appropriate name, such as *"Internet"*. Note that the system will also automatically create User entry and a RAS entry with the same name.
      2. Enter the Account Name, Password and Telephone Number details provided by the ISP.
      3. For the Firewall Profile select the firewall created previously.
   4. Click the Bandwidth tab.
      1. Set the Maximum No. of Channels to the maximum number of channels that the service should use. In this example, 12 channels were used.
      2. Leave all the other entries at their default values.
      3. If the ISP has allocated IP address details these are entered through the IP tab. If the IP Address and IP Mask define a different domain from the system LAN, then NAT is automatically applied.
   5. Click the IP tab.
      1. In the IP Address field enter the IP address specified by the ISP.
      2. In the IP Mask field enter the IP Mask specified by the ISP.
      3. The settings shown are typical. The actual settings must match those required by the ISP. For example, if Cisco routers are being used then IPHC needs to be ticked.
   6. Click the PPP tab.
      1. Ensure that the following options are selected. Leave all other options at their default settings.
         - Multilink.
         - Compression Mode: Disable.
         - Callback Mode: Disable.
         - Access Mode: Digital64
   7. Click OK.
2. Create the Virtual WAN Port
   In this stage, a WAN port is defined that actually uses T1 or T1 ISDN trunk channels.
   1. Select WAN Port to display existing ports.
   2. Click on and select WAN Port.
   3. In the Name field, enter either *LINEx.y* where:
      - *LINE* must be in uppercase.
      - *x* is the line number. For a trunk card in Slot A, this will be 1. For a trunk card in Slot B, this will be 5.
      - *y* is the lowest numbered channel number to be used by the WAN link minus 1. For example, if the lowest channel to be used is channel 1 then y = 1 - 1 = 0.
   4. In the Speed field, enter the total combined speed of the maximum number of channels sets in the Service. In this example, 12 channels x 64000 bits = 76800.
   5. Set the Mode to *SyncPPP*.
   6. In the RAS Name field, select the name used for the Service.
   7. Click OK.
3. Create an IP Route
   By creating an IP route with blank IP address details, it becomes the default route for outgoing IP traffic.

1. Select IP Route to display existing routes.

2. Click on and select IP Route.

3. Leave the IP Address and IP Mask fields blank.

4. In the Destination field, select the WAN service.

5. Leave the Metric at default value of *1*.

6. Click OK.

4. Configure the Line Channels
This stage of the process differs according to the type of trunk being used.

5. T1 Trunk
Use the following for a T1 trunk.

1. Click Line to display the existing lines.

2. Double-click on the line previously entered in the WAN Port settings.

3. Check that the Channel Allocation order matches that required by the ISP. Cisco routers typically use 1->24.

4. Select the channels to be used in the WAN PPP link and change their Channel Type to "Clear Channel 64k".

5. Click OK.

6. Click OK again.

7. Send the configuration to the system and reboot.

6. T1 PRI Trunk
Use the following for a T1 PRI trunk.

1. Click on Line to display the list of existing lines.

2. Double-click on the line previously entered in the WAN Port settings.

3. Check that the Channel Allocation order matches that required by the ISP. Cisco routers typically use 1->23.

4. Select the channels to be used in the WAN PPP link and change their Admin to "Out of Service".

5. Click OK.

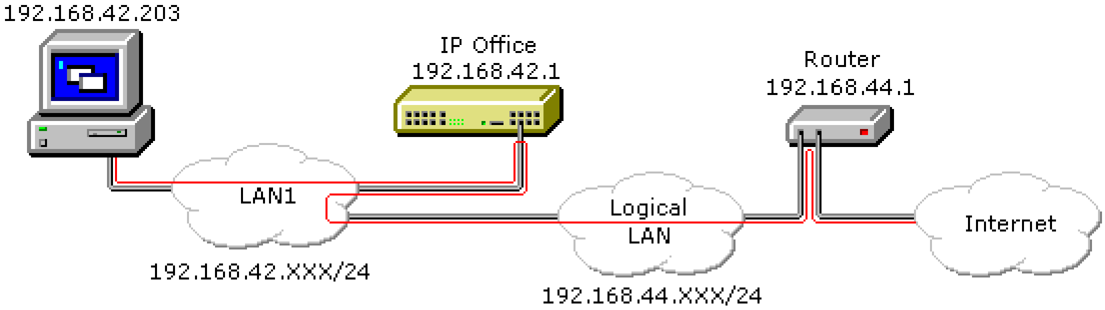
6. Click OK again.

7. Send the configuration to the system and reboot.

10.3.4 Logical LAN Connection

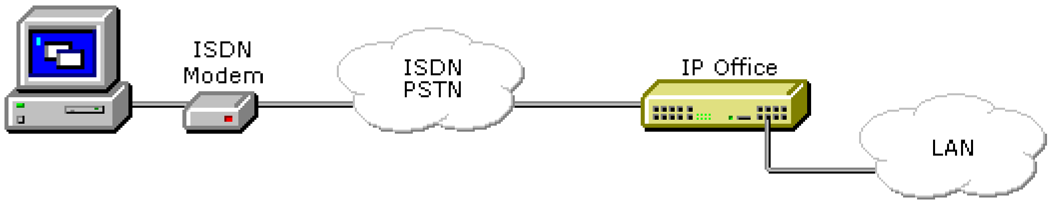
Small Office Edition and IP412 control units support two separate LAN interfaces (LAN1 and LAN2). These are separately addressed and the system's IP route table and firewalls can be used to control traffic between device attached to the two LAN's.

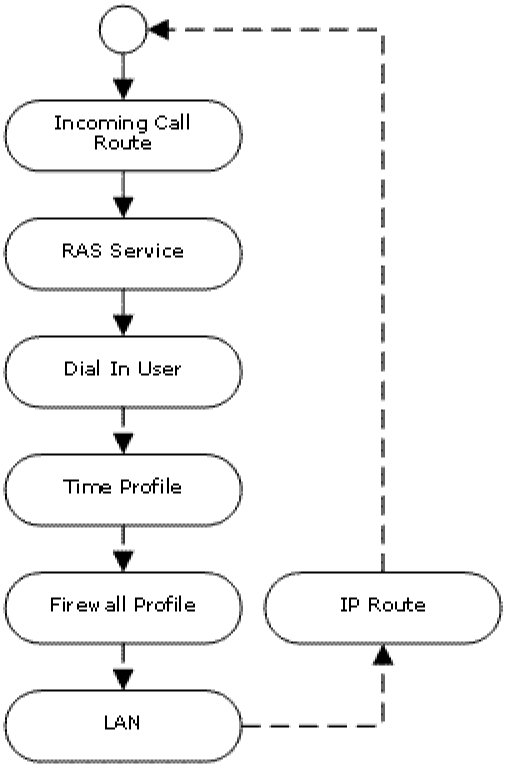
On other control units only a single LAN (LAN1) is available. A logical LAN allows these systems to support a second separately addressed LAN on the same interface. Traffic between the system LAN1 and the logical LAN can then be controlled by the system's IP route table and firewalls.



10.3.5 Remote Access

The system support remote access for incoming data calls on trunks.





To do remote access, an incoming call is passed through the following elements of the system configuration.

-  Incoming Call Route
  A Incoming Call Route is used to match incoming remote access calls and pass them to a RAS service as the destination.

-  RAS Service
  The RAS service defines settings relating to the data traffic methods useable with the call.

-  User
  The user defines the name and password required for the RAS service. The user must have Dial In On enabled.

  - An R setting on the user's Source Numbers 272 tab can be used to define the ICLID from which RAS calls are accepted.

-  Time Profile
  The user settings can specify a time profile. The time profile then controls when remote access is allowed.

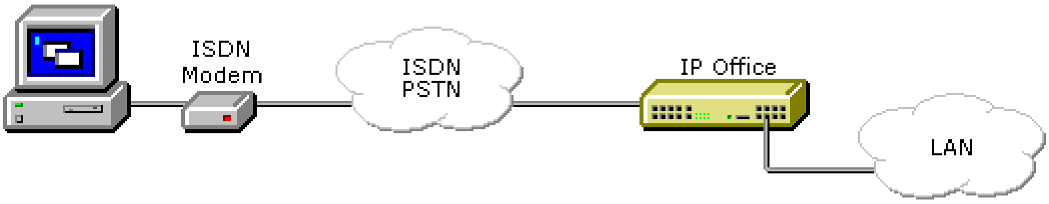
-  Firewall Profile
  The user settings can specify a firewall profile. The firewall profile then controls what traffic is allowed through the remote access connection.

  - Static NAT *(IP Office 4.2+)*
    The system supports the use of Static NAT 366 entries in firewall profiles. These are used to translate external IP addresses to internal IP addresses.

-  System | LAN
  The system can provide DHCP support for remote access connections when it is set to Server or Dial in modes. Alternatively the remote access client can use a static IP address on the system's subnet.

-  IP Route
  If the remote access client uses a IP address that is from a different subnet from the system, then a IP route entry is required for returning data. The RAS service is set as the destination.

ISDN Remote Access Example



1. Create a User
   The required details are:

   · In the User tab:
   Enter a Name and Password. The system is case sensitive. Remember to take care with passwords as this is a remote access link into your network.

   · In the Dial In tab:
   Ensure that Dial In On is ticked. The Firewall Profile and Time Profile are optional.

2. Create a RAS Entry

   · In the RAS tab:
   Enter the same name as the user that you created earlier. Again, remember this is case sensitive.

3. Create an Incoming Call Route

   · Set the Bearer Capability to *Any Data*.

   · In the Destination drop-down list, select the RAS entry created above.

   · The values that you enter for any of the other fields will depend on whether the remote user will be calling in on a particular line, number or from a set ICLID.

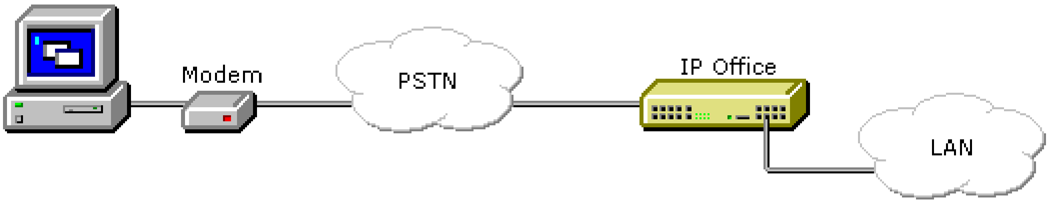
4. Is a Return IP Route Needed ?
   Go to Step 5.

5. Create a IP Route (Optional)
   If the remote user has an IP address that is not in the same domain as the system, then an IP Route is needed for return data. This is not necessary if the remote user's dial-up connection method is set to 'Obtain an IP Address Automatically' and the system's DHCP mode is set to *Server* or *Dial In*.

   · Enter the IP Address and IP Mask of the remote system.

   · In the Destination drop-down list select the RAS entry created above.

Analog Remote Access Example



Configuration for a connection from an analog modem call is very similar to the ISDN example. However the system must be able to answer modem calls. This can be done in the following ways;

- **Modem Cards**
  For control units except the Small Office Edition, IP500 and IP500v2, a modem card can be installed. This module allows the system to answer V.90 analog modem calls. The Internal Modem Card allows the system to support 12 simultaneous modem calls (4 only on the IP403). The Modem 2 card allows the system to support 2 simultaneous modem calls.

- **Analog Trunk Modem Mode**
  On systems with an analog trunk card in the control unit, the first analog trunk can be set to answer V.32 modem calls. This is done by checking the **Modem Enabled** option on the analog line settings or using the default short code *9000* to toggle this service on or off.

- When using an analog modem, the Bearer Capability of the incoming call route used should be Any Voice.

10.3.6 WAN PPP

A VoIP link across a leased line requires the Control Unit at both ends to have a Voice Compression Module installed. These provide for a fixed number of channels to use VoIP at any time. They are used to compress voice down to either 6k3 (G723) or 8k (G729) and provide echo cancellation.

Both ends must using the same version of software and configured to use the same speed and compression.

At Site A on IP address 192.168.42.1

1. Create a Normal Service:
   The Account Name and password is presented to the remote end, therefore must match the User name and password configured at Site B. The Encrypted Password option can only be used if the remote end also supports CHAP.

2. Create a User:
   Under the Dial In tab tick Dial In On. This User account is used to authenticate the connection from the Site B. As the Service and User have the same name these two configuration forms are automatically linked and become an Intranet Service. The User password is displayed at the bottom of the Service tab as the Incoming Password.

   - Name: SiteB
   - Dial In | Dial In On: Enabled.

3. Create a RAS service:
   If CHAP is to be used on this link, then the Encrypted Password option must be checked in the Service and in the RAS service. The name of the RAS service must match the name of the Service at Site B. If the RAS service is given the same name as the Service and User, they are automatically linked and become a WAN Service. Ensure that the Encrypted Password option is not checked when using a WAN Service.

4. Edit the WANPort:
   Note - do not create a new WANPort, this is automatically detected. If a WANPort is not displayed, connect the WAN cable, reboot the Control Unit and receive the configuration. The WANPort configuration form should now be added.

   · RAS Name: SiteA

5. Create an IP Route:
   The IP Address is the network address of the remote end. Under Destination select the Service created above.

6. Create a new Line:
   The Line Number and Line Group ID must be unique, in other words, not used by any other line. The Gateway IP Address is the IP Address of the Control Unit at the remote end. The Compression Mode used is dependent on the Voice Compression Card the Control Unit is running and the speed of the link.

7. Create a Short Code:
   To route all calls where the number dialed starts with 8 via Line Group ID 1, therefore via the VPN Line created above.

   - Short Code: 8N
   - Telephone Number: N
   - Line Group ID: 1
   - Feature: Dial

At Site B on IP address 192.168.45.1

1. Repeat the above steps for VoIP traffic from Site B to Site A.

- Note: For the Small Office Edition Control Unit, enabling Local Tones under the Line and Extension VoIP tabs is recommended.

10.3.7 WAN Frame Relay

To create a VoIP link via the WAN port using frame relay, the first step is to attach a WAN cable and reboot the Control Unit. After this, receive a copy of the configuration.

Both ends must using the same version of software and configured to use the same speed and compression.

At Site A

1. Create a WAN Service:
   - On the Service Tab:
     The Name is "FR_link". The Account Name should be "FR_Link" and all password fields (both Password and Incoming Password) should be left blank.
   - On the PPP Tab:
     Check the MultiLink/QoS box. Set the Header Compression Mode to IPHC.
   - On the Dial In Tab:
     If you are using a WAN 3 module, you must add "WAN" as the Dial In Service number.
2. On the Wan Port Form:
   - In the WanPort Tab
     Set the speed to match the link. Set the RAS Name to Dial In. Set the Mode as SyncFrameRelay.
   - In the FrameRelay Tab
     Set the appropriate Frame Relay Management Type. The other default settings are appropriate for a basic Frame Relay Connection.
   - In the DCLI tab
     Set the RAS Name to "FR_link". Frame Link Type = PPP. DLCI set to the network setting
3. Create a RAS service:
   Encrypted Password option is not checked when using a WAN Service. Have the Name = "FR_Link"
4. Create an IP Route:
   The IP Address is the network address of the remote end. Under Destination select the "FR_link" that was created above.
5. Create a new Line:
   The Line Number and Line Group ID must be unique, in other words, not used by any other line. The Gateway IP Address is the IP Address of the Control Unit at the remote end.
6. Create a Short Code:
   To route all calls where the number dialed starts with 8 via Line Group ID 1, therefore via the VPN Line created above.
   - Short Code: 8N
   - Telephone Number: N
   - Line Group ID: 1
   - Feature: Dial

At Site B

1. Repeat the above steps for VoIP traffic from Site B to Site A.
   - Note: For the Small Office Edition Control Unit, enabling Local Tones under the Line and Extension VoIP tabs is recommended.

Chapter 11.

Small Community Networking

11. Small Community Networking

IP Office systems linked by IP trunks can enable voice networking across those trunks to form a Small Community Network (SCN). Within an SCN, the separate systems automatically learn each other's extension numbers and user names. This allows calls between systems and support for a range of internal call features, see Supported SCN Features 812.

Small Community Networking is only supported by systems running in Manager mode. It is not supported by systems running in IP Office Essential Edition - PARTNER® Version or IP Office Essential Edition - Norstar Version modes.

Capacity

The following are the supported capacity limits for a Small Community Network system.

Software Level	Pre-5.0	5.0+	6.0+
Maximum Number of Systems	16	32	32
Maximum Number of Users	500	500	1000
Maximum H323 SCN Line Hops Between Systems	5	5	5
Star H323 SCN Line Layout	✓	✓	✓
Serial H323 SCN Line Layout	✓	✓	✓
Mesh H323 SCN Line Layout	✗	✓	✓

Configuration Summary

To set up a small community network, the following are required:

- A working H323 trunk between the systems, that has been tested for correct voice and data traffic routing.
  - The arrangement the H323 trunks must meet the requirements detailed in Supported SCN Network Layouts 811.
  - On IP500 and IP500v2 systems, H323 trunks require the entry of **IP500 Voice Networking** licenses.
- VCM channels are required in all systems.
- The extension, user and group numbering on each system must be unique.
- The user and group names on each system must be unique.
- We also recommend that all names and numbers (line, services, etc) on the separate systems are kept unique. This will reduce potential maintenance confusion.
- The **Outgoing Group ID** on the SCN lines should be changed to a number other than the default *0*.
- All systems should use the same set of telephony timers, especially the **Default No Answer Time**.
- Only one system should have its **Voicemail Type** set to *Voicemail Pro/Lite*. All other systems must be set to either *Centralized Voicemail* or *Distributed Voicemail*. No other settings are supported.

Software Level Interoperation

SCN is supported between systems with the same major software level or one level of difference in major software level. For example between 6.1 and 6.0 (same major level) and between 7.0 and 6.0 (one major level of difference).

This option is intended mainly to allow the phased upgrading of sites within a Small Community Network. It is still recommended that all systems within a network are upgraded to the same level where possible. Within a SCN including differing levels of software, the network features and capacity will be based on the lowest level of software within the network.

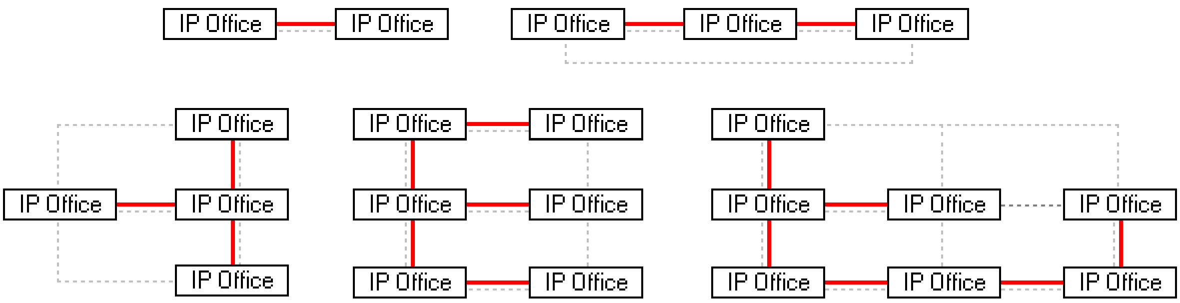
- For a Small Community Network with more than 16 systems, only IP Office 5.0 and higher systems are supported. If a pre-IP Office 5 system is included in a network of more than 16 systems, it will not receive any SCN support.

11.1 Supported SCN Network Layouts

The allowed arrangement of H323 SCN lines between the systems depends on the lowest software level of any system in the network. Note that we are referring to H323 SCN lines configured in the system configurations. The actual IP network configuration, including IP routes in the system configurations, can differ as per the customer network requirements.

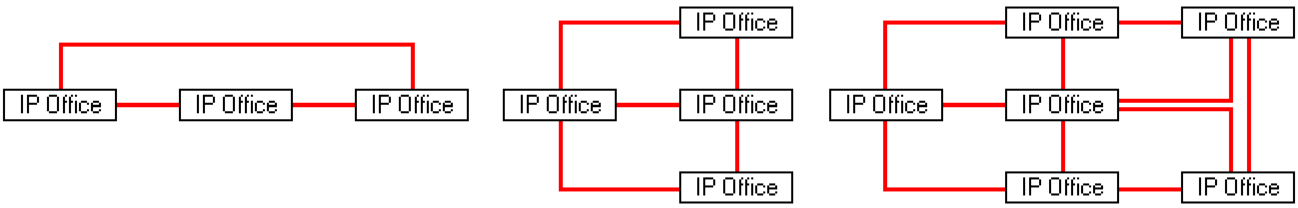
Software Level	Pre-5.0	5.0+	6.0+
Maximum Number of Systems	16	32	32
Maximum Number of Users	500	500	1000
Maximum H323 SCN Line Hops Between Systems	5	5	5
Star H323 SCN Line Layout	✓	✓	✓
Serial H323 SCN Line Layout	✓	✓	✓
Mesh H323 SCN Line Layout	✗	✓	✓

Star/Serial Layouts
The following are examples of star and serial layouts. These are the only types of layouts supported for Small Community Networks containing any pre-IP Office 5 systems.



---- = IP network, | = IP Office H323 SCN Line.

Mesh Layout
The use of 'mesh' layouts connections is only supported for a Small Community Network of IP Office 5.0 or higher systems. A mesh layout is one where there is more than one possible H323 SCN Line route between any two systems. The following are examples of mesh layouts. Mesh, star and serial layouts can be combined.



SCN Signalling
SCN uses a signalling similar to RIP is order to update each other of there presence. This traffic can be seen in the System Monitor application as AVRIP packets. This traffic is is sent to port 50795 on which each system listens.

Each system in the SCN transmits an update every 30 seconds. Additionally BLF updates are transmitted when applicable up to a maximum of every 0.5 seconds. Typically the volume is less than 1Kbps per system.

11.2 Supported SCN Features

The following features are supported between systems in a Small Community Network.

Feature	Description	Minimum Software Level
Extension Dialing	Each system automatically learns the user extension numbers available on other systems and allows routes calls to those numbers.	
Absence Text	Calling an SCN user with absence text set will display the absence text.	
Call Tagging	Call tag text added to a call is retained by calls routed across the Small Community Network.	
Anti-tromboning	Calls routed across the SCN and back to the originating system are turned back into internal calls on the originating system only.	
Hold	Hold and held calls are signalled across the SCN.	
Transfer	Calls can be transferred to SCN extension numbers.	
Forwarding	Calls can be forwarded to SCN extension numbers.	
Paging	Page calls can be made to SCN extension numbers.	
Call Pick-Up Extension	The directed call pickup feature can use SCN extension numbers as its target for the call pickup.	
Callback When Free	When calling a busy or unanswered extension across the SCN, the callback when free function can be used to set a callback.	
Conference	Remote SCN extension can be added to a conference. Note that a conference can only use the conference resources on the system it is started.	
Directory	Inclusion of SCN users and groups in the phone directory of each system. system directory entries on each system are not automatically shared. For IP Office 5, systems can be configured to import directory records from another system. See using HTTP 171.	
User DSS/BLF	Monitoring of user status only. The ability to use additional features such as call pickup via a USER button will differ depending on whether the monitored user is local or remote. Indication of new voicemail messages provided by Phone Manager/SoftConsole user speed dial icon is not supported across an SCN.	
Advertised Hunt Groups	A hunt group can be set to be 'advertised'. Hunt groups that are advertised can be dialed by users on other systems within the Small Community Network (SCN) without the need for short codes.	4.0
Distributed Hunt Groups	Hunt groups on a system can include users located on remote systems within the SCN network. A distributed hunt group can only be edited on the system on which they were created.	4.0
Remote Hot Desking	Users can hot desk between systems within the network. The system on which the user configured is termed their 'home' system, all other systems are 'remote' systems. When a user logs in to a remote system; • The user's incoming calls are rerouted across the SCN. • The user's outgoing calls use the settings of the remote system at which they are logged in. • The user's own settings are transferred. However some settings may become unusable or may operate differently. • Users settings are transferred but user rights are not. However if user rights with the same name exist on the remote system then they will be used. The same applies for user rights applied by time profiles, if a time profile with the same name also exists on the remote system . • Appearance buttons configured for users on the home system will no longer operate.	4.0

Feature	Description	Minimum Software Level
	• Various other settings may either no longer work or may work differently depending on the configuration of the remote system at which the user has logged in. For example T3 phones the personal directory is not transferred with the user. • If a remote user's home system is no longer visible within the SCN, they are automatically logged out after 24 hours.	
Break Out Dialing	This feature allows the user to select a system in the network from a displayed list and then dial a subsequent number as if dialing locally on the select system. This feature is triggered either by a programmable button 581 or short code 441.	4.0
Music on Hold Source Selection	The 4 possible music on hold sources are numbered 1 to 4, with 1 being the System Source. Where the source number to associate with a call is changed, that number is retained when the call is rerouted across the SCN. If the receiving switch has a matching numbered alternate source, that source is used.	4.2
Mobile Call Control	Licensed mobile call control users who remote hot desk to another system within the SCN take their licensed status with them rather than consuming (if available) a license on the system to which they hot desk.	4.2
Fallback	IP Office 5 systems can provide fallback support for Avaya H323 phones, advertised hunt groups and centralized voicemail.	5.0
Centralized Personal Directory	The personal directory numbers stored in the system for each use can now be viewed and called using T3, 1400, 1600 and 9600 Series phone. If the user hot desks to another system in the Small Community Network, their personal directory is still available.	5.0
Centralized Call Log	The centralized call log features include support for users hot desking within a Small Community Network. • The user's call log records are stored by the system that is their home system, ie. the one on which they are configured. When the user is logged in on another system, new call log records are sent to the user's home system, but using the time and date on the system where the user is logged in. • Hunt group call log records are stored on the system on which the hunt group is configured.	5.0
Fax Relay	For an IP500 or IP500v2 system with an IP500 VCM card, Fax relay can be configured on SIP lines and extensions. Within a Small Community Network, fax calls can transition from a trunk or extension with T38 support to a system SCN line with Fax Transport Support enabled.	5.0
User Profile Resilience	When a user hot desks to another switch in the SCN, they retain their Profile settings and rights as validated by the licenses in their home switch's configuration. They do not use a license on the remote switch. This applies even if their home switch is temporarily removed from the SCN.	6.0
Distributed Voicemail Server Support	When using Voicemail Pro 6.0 or higher, each system can support its own Voicemail Pro server in addition to the central Voicemail Pro server. The distributed severs can provide call recording and auto attendant functions to their local system. The central Voice Pro server is still used as the message store for all messages. For full details refer to the Voicemail Pro Installation and Maintenance Manual.	6.0

11.3 Voicemail Support

Within a Small Community Network, a single Voicemail Pro can be used to provide voicemail services for all the systems. For full details of installation and setup refer to the Voicemail Pro documentation. The Voicemail Pro is licensed and hosted by a chosen central system and provides full operation for that system. The voicemail features supported for the other remote systems are listed below:

  - For pre-IP Office 5 IP500 systems, centralized voicemail does not require the remote systems to be running in IP500 Professional mode. Only the IP500 hosting the Voicemail Pro server is required to have an *IP500 Upgrade Standard to Professional* license.
  - The use of additional Voicemail Pro servers is supported by IP Office 6.0 and higher. For full details refer to the Voicemail Pro Installation and Maintenance Manual.

- User mailboxes.
- Call recording.
  Recording of incoming call routes is only supported for destinations on the same system, not for remote SCN destinations.
- Dial by Name.
- Auto Attendants.
- Breakout
  Requires that the numbers used are routable by the system hosting the voicemail server.
- Announcements
  Using IP Office 4.0 announcements. Pre-4.0 announcements are only supported for queues on the system hosting the voicemail server.
- ContactStore
  This application is supported but requires each individual system to have a VMPro Recordings Administrator license.
- UMS Web Services
  Users for UMS Web Services (IMAP and or web voicemail) are licensed through the *UMS Web Services* license on their host system. This applies even if the user remote hot desks to another system in the Small Community Network.

When using Voicemail Pro 6.0 or higher, each system can support its own Voicemail Pro server in addition to the central Voicemail Pro server. The distributed severs can provide call recording and auto attendant functions to their local system. The central Voice Pro server is still used as the message store for all messages. For full details refer to the Voicemail Pro Installation and Maintenance Manual.

11.4 Enabling Small Community Networking

The process below adds an H323 SCN Line to the system configuration. It is assumed that data routing between the systems has already been configured and tested. For Manager 8.1, adding SCN connections between systems can also be done using Manager's SCN Management817 mode.

- A working H323 trunk between the systems, that has been tested for correct voice and data traffic routing.
  - The arrangement the H323 trunks must meet the requirements detailed in Supported SCN Network Layouts811.
  - On IP500 and IP500v2 systems, H323 trunks require the entry of **IP500 Voice Networking** licenses.
- VCM channels are required in all systems.
- The extension, user and group numbering on each system must be unique.
- The user and group names on each system must be unique.
- We also recommend that all names and numbers (line, services, etc) on the separate systems are kept unique. This will reduce potential maintenance confusion.
- The **Outgoing Group ID** on the SCN lines should be changed to a number other than the default *0*.
- All systems should use the same set of telephony timers, especially the **Default No Answer Time**.
- Only one system should have its **Voicemail Type** set to *Voicemail Pro/Lite*. All other systems must be set to either *Centralized Voicemail* or *Distributed Voicemail*. No other settings are supported.

A. Setup the VoIP Line from System A to System B

1. Receive the system configuration for System A. Prepare the system for addition to the SCN:
   a. For IP500 and IP500v2 systems, check that the **Voice Networking** license is present and valid.
   b. Change all extensions numbers and names to values that will be unique within the SCN.
      - For users and extensions this can be done using the **Extension Renumber** tool. That will adjust all users and extension and all items using those numbers, for example hunt group memberships and incoming call routes.
      - For hunt groups, each hunt group must be change individually.
2. Click **Line** to display a list of existing lines.
3. Right-click on the displayed list and select **New** and then **H323 Line**.
4. Select the **Line** tab and set the following:
   - In the **Telephone Number** field, enter a description of the link. For example *System B SCN*.
   - Set the **Outgoing Group ID** to a unique value. For example match the automatically assigned **Line Number** value shown above.
5. Select the **VoIP Settings** tab and set the following:
   - For the **Gateway IP Address**, enter the IP address of the remote System B.
   - In the **Supplementary Services** field select *IP Office SCN*. Use of *IP Office SCN - Fallback* is detailed in SCN Fallback826.
   - Select the preferred **Compression Mode**. The same mode must be used by all VoIP lines and extensions within the network.
   - The other option can be configured as required but must be matched by the other H323 SCN lines in the network.
6. Select **System | Voicemail**.
   a. Only one system should have its **Voicemail Type** set to *Voicemail Pro/Lite*. The **Voicemail IP Address** will be the IP address of the central voicemail server PC.
   b. Any other system with its own Voicemail Pro server PC should have its **Voicemail Type** set to *Distributed Voicemail*. The **Voicemail IP Address** should be the IP address of the distributed voicemail server PC. The **Voicemail Destination** should be set to the **Outgoing Group ID** used for the SCN line to the system that is set as *Voicemail Pro/Lite*.
   c. All other systems should have their Voicemail Type set to Centralized Voicemail. The **Voicemail Destination** should be set to the **Outgoing Group ID** used for the SCN line to the system that is set as *Voicemail Pro/Lite*.
7. Save the configuration and reboot System A.

B. Setup the VoIP Line from System B to System A

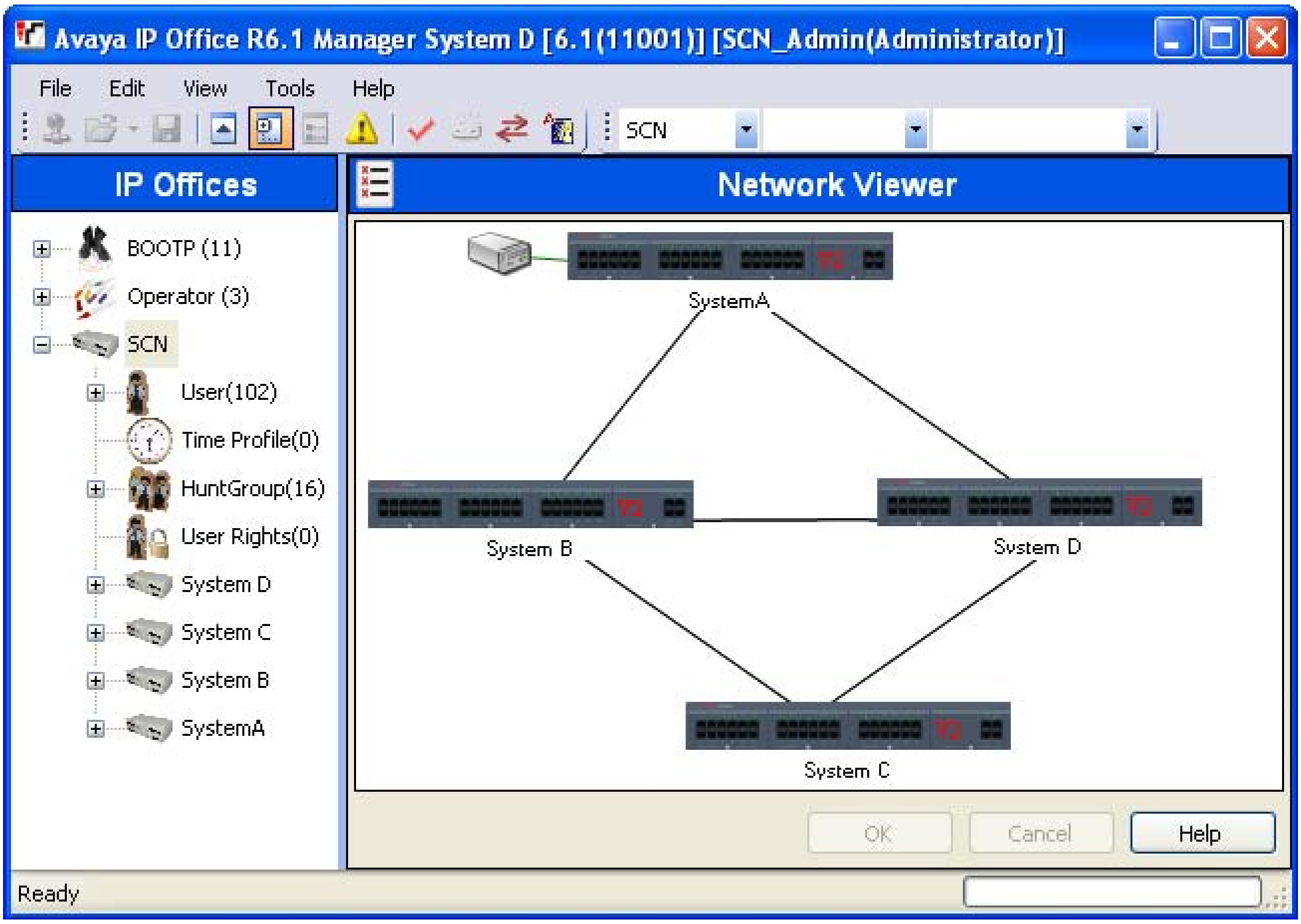
1. On the remote system, repeat the previous steps to create an H323 SCN line to System A.
2. Load the configuration and reboot the remote system.

C. Test Small Community Networking

1. Test by making calls between extensions on the different systems.

11.5 SCN Management

Manager 8.1 and higher supports the ability to load and manage the configurations of the systems in an Small Community Network at the same time. This requires Manager to be enabled for SCN discovery and at least one system in the SCN to have 6.0 or higher software.



When the configurations of the systems in an SCN are loaded, Manager switched to SCN management mode. This differs from normal system configuration mode in a number of ways:

- A network viewer is available. In addition to giving a graphical view of the SCN, the view can be used to add and remove links between the systems in the SCN.
- In the configuration tree, the entries for users and hunt groups on all systems are grouped together.
- Time Profiles and User Right common to all systems are grouped together.
- The configuration settings for each system in the SCN can be accessed and edited.

11.5.1 Enabling SCN Discovery

In order for the Select IP Office menu to groups systems in an SCN and allow loading of all the SCN configurations, Manager must be enabled for SCN discovery.

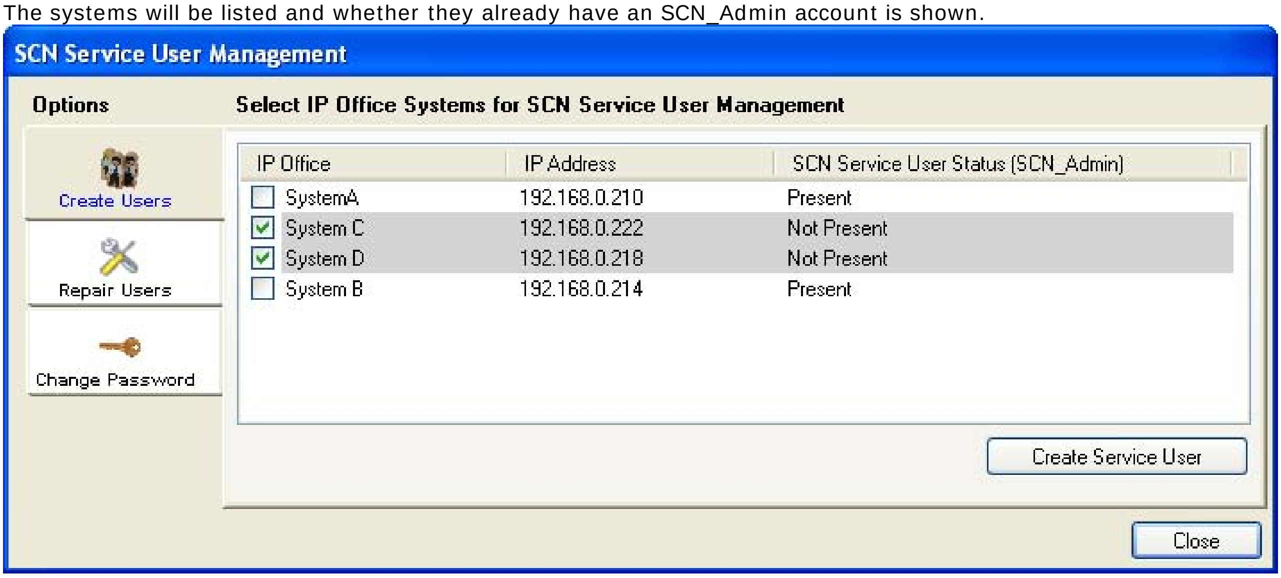
1. Select File | Preferences.
2. Select the Discovery tab.
3. Select the SCN Discovery option.
4. Check that the other discovery setting are sufficient to discover all the systems in the SCN.
5. Click OK.

11.5.2 Creating a Common Admin Account

When managing multiple systems, it may be useful to create a common user name and password on all the systems for configuration access. This tool can be used to create a new service user account, SCN_Admin, for configuration access.

This process requires you to have a user name and password for security configuration access to each of the systems.

1. Select Tools | SCN Service User Management.
   - The option is not shown if a IP Office Essential Edition - PARTNER® Version or IP Office Essential Edition - Norstar Version system configuration is loaded. If no configuration is load, and the option is not shown, select View | Advanced View.
2. The Select IP Office menu will display the list of discoverable systems.
3. Select the systems for which you want to create a common configuration account. Click OK.
4. A user name and password for security configuration access to each system is request. Enter the values and click OK. If the same values can be used for all systems enter those values, select Use above credentials for all remaining, selected IPOs. If each system requires a different security user names and password, deselect Use above credentials for all remaining, selected IPOs.
5. The systems will be listed and whether they already have an SCN_Admin account is shown.



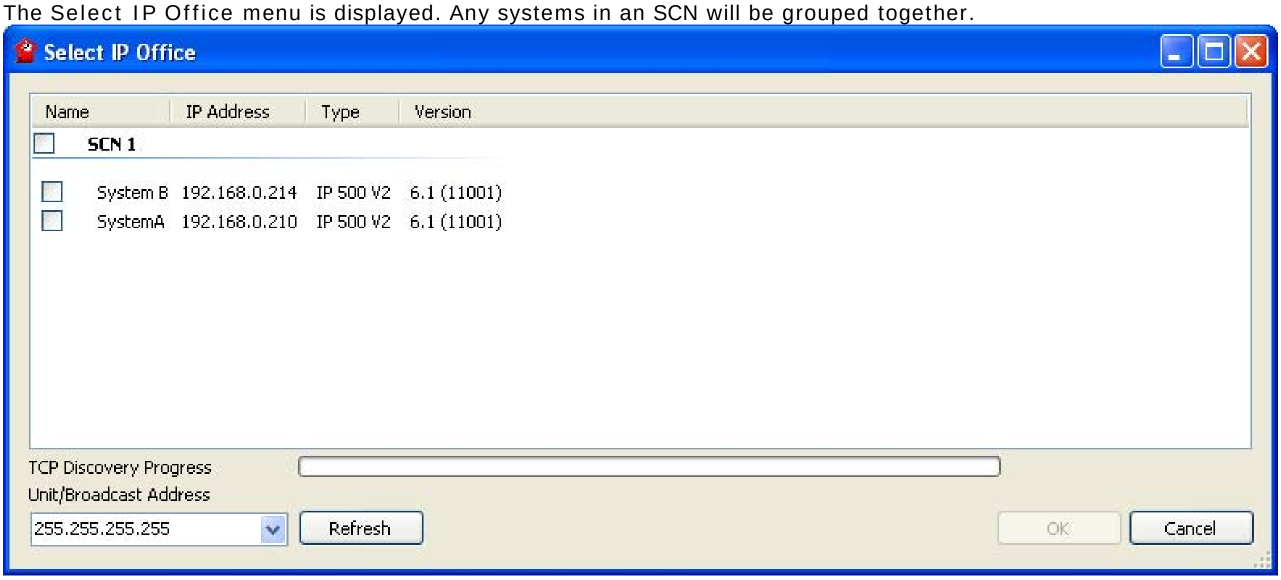
6. To create the SCN_Admin account on each system and set the password for those account click on Create Service User.

7. Enter the common password and click OK.
8. The password can be changed in future using the Change Password option.
9. Click Close.

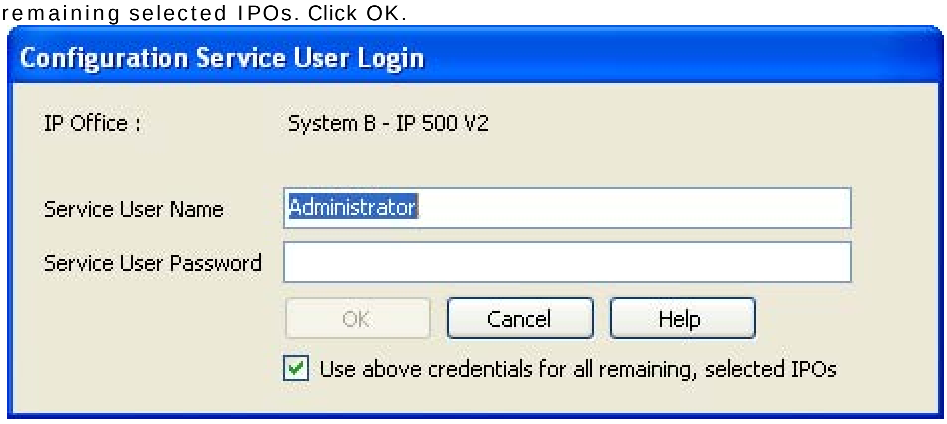
11.5.3 Loading an SCN Configuration

If Manager is configured with SCN Discovery 817 enabled, the Select IP Office menu will display SCNs it discovers.

1. With no configuration loaded, click on or select File | Open Configuration.
2. The Select IP Office menu is displayed. Any systems in an SCN will be grouped together.



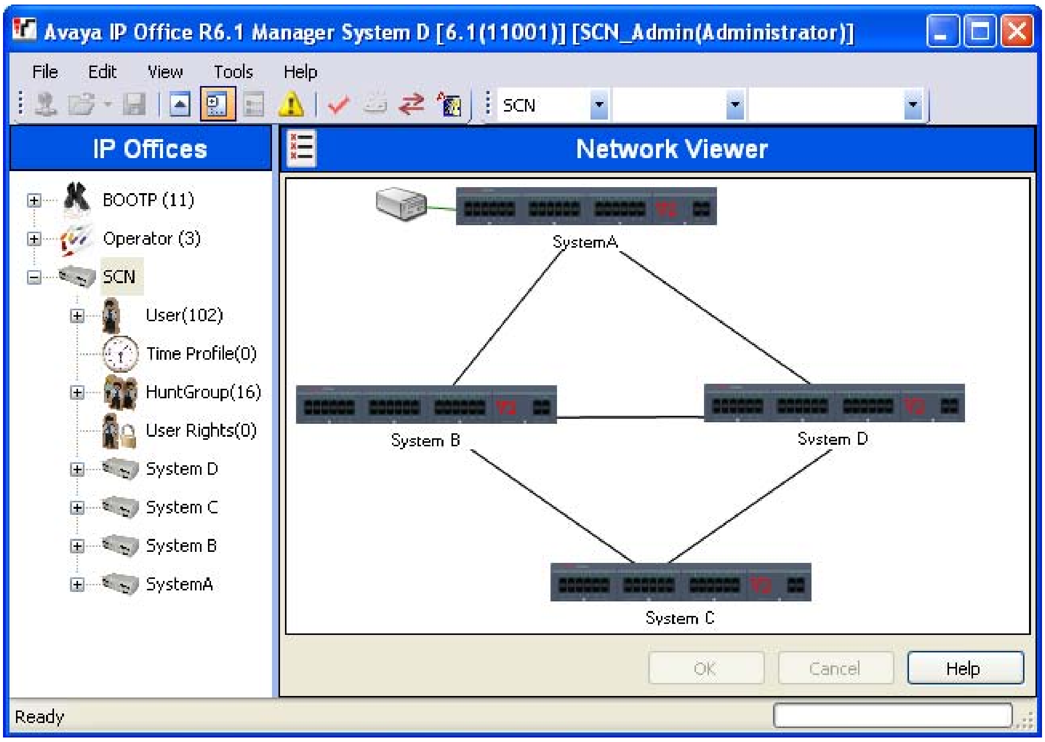
3. To load the configuration of all the systems in the SCN, click the check box next to the SCN name and then click OK .
   - If a warning icon is displayed next to the SCN check box, it indicates that not all the systems known to be in the SCN were discovered. Hovering the cursor over the icon will display details of the missing systems. Loading the SCN configuration at this time would not include the configuration of the missing system or systems. The missing systems:
     - May be disconnected
     - The discovery settings 108 for the Manager PC may be incorrect.
     - The data routing between the Manager PC and the missing systems may be incorrect or blocked.
4. Enter the name and password for configuration access to each system. If the systems all have a common user name and password (see Common Administrator Access 818 below), select Use above credentials for all remaining selected IPOs. Click OK.



5. Manager will load and display the combined configurations in SCN Management mode.

11.5.4 Editing an SCN Configuration

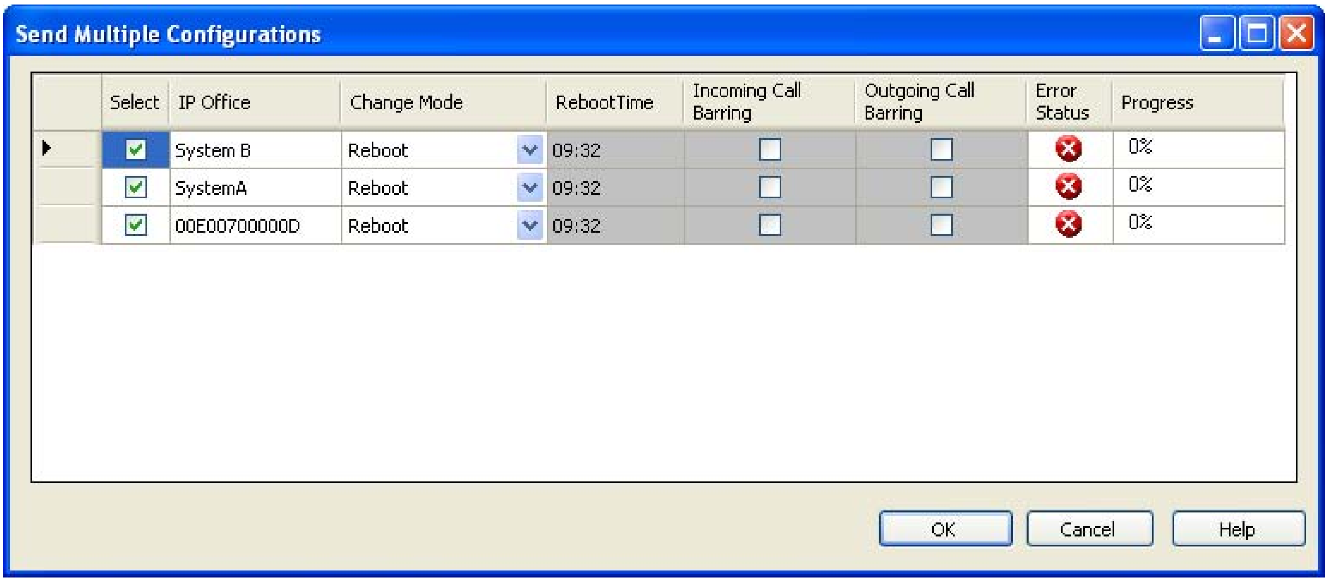
When the configuration of an SCN is loaded, Manager displays the configuration in a different way from when the configuration of a single system is loaded. The main differences are in how configuration entries are grouped in the configuration tree.



- Clicking on the  SCN icon displays the Network Viewer 822 which shows the lines between the systems in the SCN.
- SCN Configuration Entries
  Certain entries from each of the systems in the SCN are grouped together in the configuration tree differently from when just a single system configuration is loaded. There are two types, unique SCN entries and shared SCN entries:
  - Unique Entries
    They can be edited here and the system to which they belong is indicated in the group pane and in the title bar of the details pane. However, to add or delete these types of entry must be done within the configuration entries of the particular system that will host the entry's configuration details.
    - All user in the SCN are shown under the  User icon.
    - All hunt groups in the SCN are shown under the  Hunt Group icon.
  - Shared Entries
    Shared entries are configuration items that exist on all systems in the SCN, having the same name and settings on each system. Editing the shared entry updates the matching copy in the configuration of each system. Similarly, adding or deleting a shared entry adds or deletes from the individual system configurations. If the copy of the shared entry within an individual configuration is edited, it is no longer a shared entry for the SCN though the individual entries on other system will remain. Changing the individual entries back to matching will turn the entries back into a shared entry.
    - Shared time profiles are shown under the  Time Profile icon.
    - Shared user rights are shown under the  User Rights icon.
- Individual System Configurations 
  The full configuration for each system in the SCN can be accessed and edited as required. It is possible to copy and paste configuration entries between systems using the configuration tree.

Saving Changes

When the save icon or File | Save Configuration is selected, the menu for multiple configuration saves is displayed. It provides similar options are for a normal single configuration save. Note that when working in SCN Management mode, after saving configuration changes the Manager will always close the displayed configuration.



- **Change Mode**
  If Manager thinks the changes made to the configuration settings are mergeable, it will select Merge by default, otherwise it will select Reboot.

  - **Merge**
    Send the configuration settings without rebooting the system. This mode should only be used with settings that are mergeable. Refer to Mergeable Settings 51.

  - **Reboot**
    Send the configuration and then immediately reboot the system.

  - **Reboot When Free**
    Send the configuration and reboot the system when there are no calls in progress. This mode can be combined with the Call Barring options.

  - **Timed**
    The same as When Free but waits for a specific time after which it then wait for there to be no calls in progress. The time is specified by the Reboot Time. This mode can be combined with the Call Barring options.

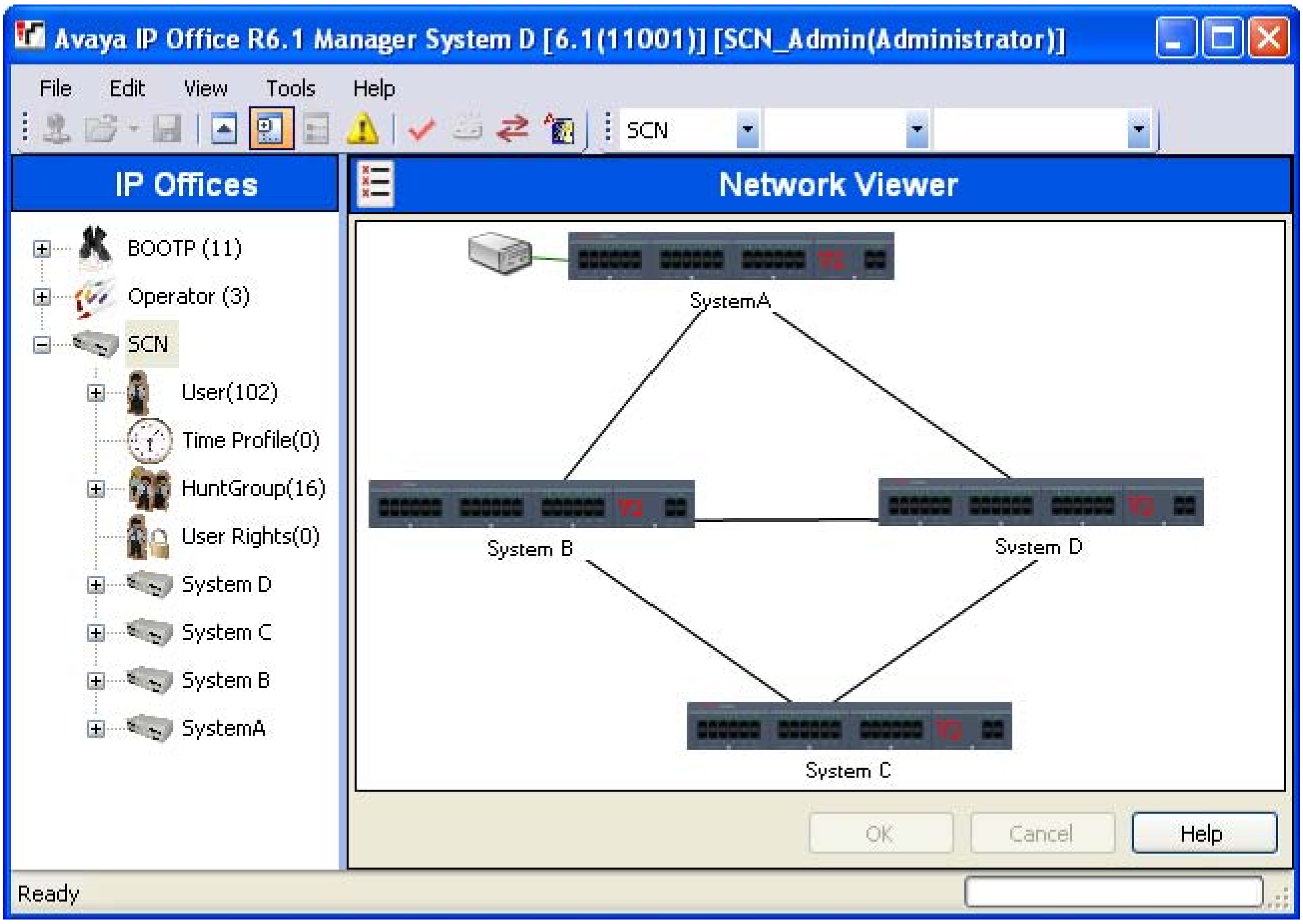
- **Reboot Time**
  This setting is used when the reboot mode Timed is selected. It sets the time for the system reboot. If the time is after midnight, the system's normal daily backup is canceled.

- **Call Barring**
  These settings can be used when the reboot mode Reboot When Free is selected. They bar the sending or receiving of any new calls.

- **Error Status**
  The warning will appear if the configuration being sent contains any validation errors indicated by a icon in the error pane. The configuration can still be sent if required.

11.5.5 Using the Network Viewer

Clicking on SCN in the configuration tree displays the Network Viewer. This shows each of the systems in the SCN and the links between each of the systems. Systems with attached Voicemail Pro servers are also indicated.



- Green
  System with Voicemail Pro system.
- Black
  SCN line between two systems.
- Red
  Incorrect SCN line between systems (probably one-way connection). Right-click on the line and select Repair.

You can use the Network Viewer to perform a range of functions:

- Arrange the View
- Launch System Status
- Launch Voicemail Pro
- Add an SCN Line
- Add a system
- Remove an SCN Line
- Remove a system from the SCN
- Repairing an SCN Line
- Add a Background Image

Arranging the View

You can click and drag items around in order to position them where required. Alternatively if you right click on the view you can select Auto Arrange.

Note that the position of elements in the network view are stored as part of the system configuration. Therefore changes to the view will require the configuration to be saved.

Adding a Line Within the SCN

You can use the network viewer to add an SCN link between two systems in the SCN that are currently linked. This process will add new H323 SCN line entries to the configurations of each of the systems.

1. Note that adding a line between systems will require those systems to reboot when the changes are saved.
2. Right click on the starting system for the link. Select **Connect To** and select the name of the other system in the SCN to which you want to link.
3. Select the type of line, **IP Office SCN** or **IP Office SCN-Fallback**. Click OK.
   - If SCN-Fallback is selected, the actual backup function 826 still need to be configured.
4. The newly added line is displayed in the network viewer.
5. Click OK.

Repairing a Line Within the SCN

A red line in the network viewer indicates a incorrectly configured line between two systems in the SCN. Typically this will be a line configured in one of the systems but not matched by a line configured in the other system. The network viewer can be used to correct this error.

1. Note that adding a line between systems will require those systems to reboot when the changes are saved.
2. Right click on the red line and select **Repair Line**.
3. The line is changed to black.
4. Click OK.

Adding a System to the SCN

You can use the network viewer to add an SCN line to a system not yet in the SCN. This process will add new H323 SCN line entries to the configurations of each of the systems.

1. Note that adding a line between systems will require those systems to reboot when the changes are saved.
2. Right click on the starting system for the link. Select **Connect To** and select **Discovery**.
3. The **Select IP Office** menu will display any discoverable systems not already in the SCN.
   - If the discovery includes systems already in another SCN it will not indicate such. If you want to add such a system in order to join the SCNs you can do so. However after adding the system, you should immediately save the configuration and reload the SCN.
4. Select the required system and click OK.
5. Enter the name and password for configuration access to the selected system and click OK.
6. The newly added system is displayed in the network viewer.
7. Click OK. The configuration of the newly added system is now included in the configuration tree.
8. If the Error List is visible (**View | Error Pane**), check that none of the error are SCN specific errors, for example duplicate names or extension numbers.

Removing an SCN Line

You can use the network viewer to remove the SCN lines between two systems in the SCN.

1. Note that removing a link between systems will require those systems to reboot when the changes are saved.
2. Right click on the link and select **Delete** Line.
3. The line is removed from the network viewer.
4. Click OK.

Removing a System
You can use the network viewer to remove a system from the SCN.

1. Note that removing a system will require previous linked systems to reboot when the changes are saved.
2. Right click on the system and select **Remove From SCN**.
3. Any lines to other system in the SCN are removed.
4. Click OK.

Start System Status
If the System Status Application is also installed on the Manager PC, you can start it for a particular system.

1. Right click on the system and select **System Status**.
2. The application is started and the login form pre filled with the IP address of the system.

Start Voicemail Pro
If the Voicemail Pro client is also installed on the Manager PC, you can start it for the any system with an associated Voicemail Pro server.

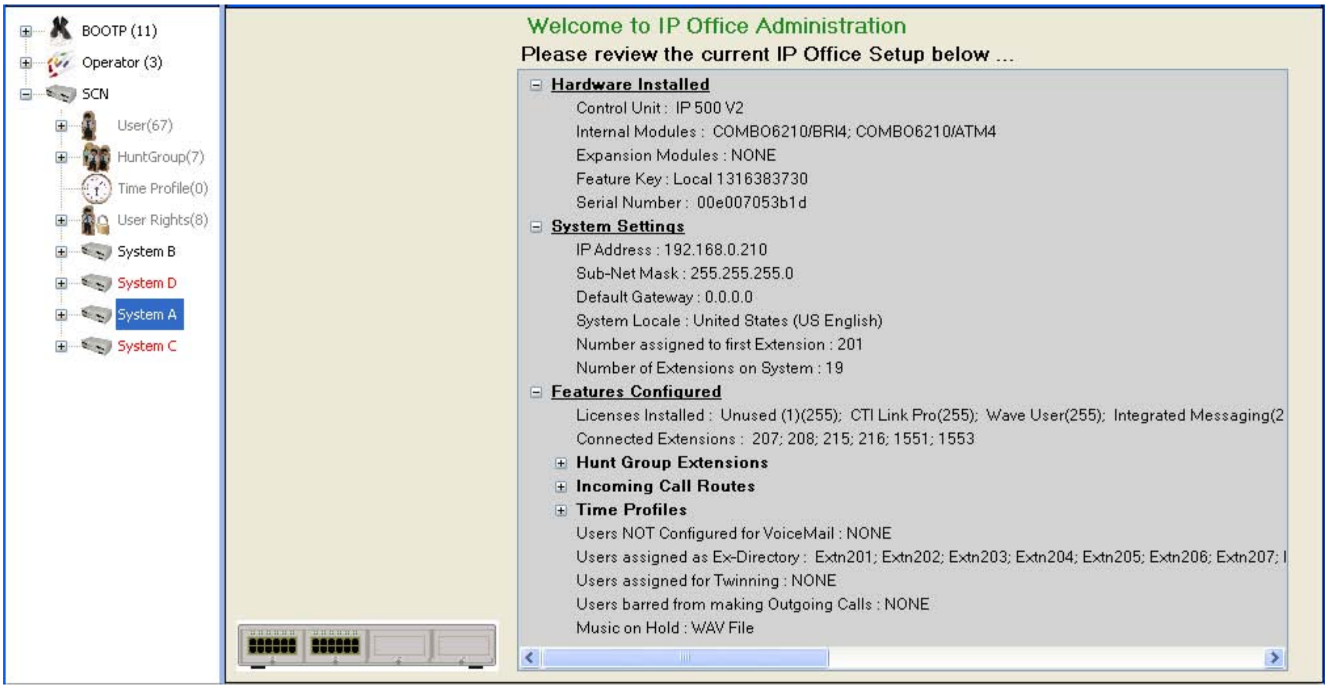
1. Right click on the voicemail server icon and select **Launch VMPro Client**.

Add a Background Image
You can select an image file to be displayed in the background of the Network Viewer display. This file is not saved as part of the configuration in any way, ie. if the image file is moved or deleted it is not longer used by Manager.

1. Right click on the general background area of the network viewer and select **Background Image**.
2. Select **Set Background Image** to browse to the location of the file to be used.
3. The **Visible** option can be used to switch the display of the background image on or off.

11.5.6 System Inventory

When working in SCN Management 817 mode, clicking on the System icon for a particular system displays a system inventory 825 page for that system.



11.6 SCN Remote Hotdesking

IP Office 4.0+ supports hot desking 789 between systems within a Small Community Network 810. In the descriptions below, the system on which the user is configured is termed their 'home' system, all other systems are 'remote' systems. For pre-IP Office 5 systems, this feature requires the systems to have *Advanced Small Community Networking* licenses.

When a user logs in to a remote system:

- The user's incoming calls are rerouted across the SCN.
- The user's outgoing calls uses the settings of the remote system.
- The user's own settings are transferred. However some settings may become unusable or may operate differently.
  - User rights are not transferred to the remote system but the name of any user rights associated with the user are transferred. If user rights with the same name exist on the remote system, then they will be used. The same applies for user rights applied by time profiles, if time profiles with the same name exist on the remote system .
  - Appearance buttons configured for users on the home system will no longer operate.
  - Various other settings may either no longer work or may work differently depending on the configuration of the remote system at which the user has logged in. For example T3 phones the personal directory is not transferred with the user.
  - For IP Office Release 6.0+, the rights granted to the user by their Profile settings are retained by the user. There is no requirement for the remote system to have the appropriate licenses for the Profile.
- If the user's home system is disconnected from the SCN while the user is remotely hot desked, the user will remain remotely hot desked. They can remain in that state for up to 24 hours before being automatically removed from the remote system if their home system has not reconnected to the SCN. Note also that when the user's home system is reconnected to the SCN, the user may be automatically logged on that system.

- Break Out Dialing
  In some scenarios a hot desking user logged in at a remote system will want to dial a number using the system short codes of another system. This can be done using either short codes with the Break Out feature or a programmable button set to Break Out. This feature can be used by any user within the Small Community Network but is of most use to remote hot deskers.

11.7 SCN Fallback

IP Office 5 provides a number of features that allows systems to provide fallback functions for each other. This operation requires the systems to be running IP Office 5+.

Each system in the SCN can include one H323 line where the Supplementary Services is set to *IP Office SCN - Fallback* rather than *IP Office SCN*. The system to which the H323 line connects can then be used to provide fallback support for selected options for the local system.

Important

- Fallback handover takes approximately 3 minutes. This ensure that fallback is not invoked when it is not required, for example when the local system is simply being power cycled to complete a non-mergeable configuration change.
- Fallback is only intended to provide basic call functionality while the cause of fallback occurring is investigated and resolved. If users make changes to their settings while in fallback, for example changing their DND mode, those changes will not apply after fallback.
- If the fallback system is rebooted while it is providing fallback services, the fallback services are lost.
- Fallback features require that the IP devices local to the local system are still able to route data to the fallback system when the local system is not available. This will typically require each system site to be using a separate data router from the system.

Fallback Options

Once a line is set to *IP Office SCN - Fallback*, the following options are available:

- SCN Backup Options: *Software level = 5.0+.*
  These options are only available on when the Supplementary Services option is set to *IP Office - Fallback*. The system to which the trunk connects must be an IP Office 5+ system. The intention of this feature is to attempt to maintain a minimal level of operation while problems with the local system are resolved.

- Backs up my IP Phones: *Default = On.*
  This option is used for Avaya 1600, 4600, 5600 and 9600 Series phones registered with the system. When selected, it will share information about the registered phones and users on those phones with the other system.
  - If the local system is no longer visible to the phones, the phones will reregister with the other system. The users who were currently on those phones will appear on the other system as if they had hot desked.
  - Note that when the local system is restored to the SCN, the phones will not automatically re-register with it. A phone reset via either a phone power cycle or using the System Status Application is required.
  - When phones have registered with the other system, they will show an R on their display.
- Backs up my Hunt Groups: *Default = On.*
  When selected, any hunt groups the local system is advertising to the SCN are advertised from the other system when fallback is required. The trigger for this occurring is Avaya H323 phones registered with the local system registering with the other system, ie. Backs up my IP Phones above must also be enabled.
  - When used, the only hunt group members that will be available are as follows:
    - If the group was a distributed hunt group, those members who were remote members on other systems still visible within the SCN.
    - Any local members who have hot desked to another system still visible within the SCN.
  - When the local system becomes visible to the other system again, the groups will return to be advertised from the local system.
- Backs up my Voicemail: *Default = On.*
  This option can be used if the local system is hosting the Voicemail Pro server being used by the SCN. If selected, when the local system is no longer visible to the voicemail server, the other system will act as host for the voicemail server.
  - This option requires the other system to have licenses for the Voicemail Pro features that are required to operated during any fallback period.
  - This option requires Voicemail Pro 5.0+.

11.8 SCN Short Code Programming

With Small Community Networking enabled, the systems automatically learn each others extension numbers and route calls appropriately. However the same does not apply to dialing of other numbers. Using short codes it is possible to have the dialing of numbers on one system to be redirected to another system and dialed there. The dialing is then matched against the short codes available on the remote system.

Scenario

We want a short code on System A which will correctly route any 3000 range number to System B. This will allow System B group numbers to be dialed from System A. To achieve the above scenario, we will add a new system short code. By using a system short code it becomes available to all users.

1. Receive the configuration from System A.
2. Click **Short code** to display a list of existing system short codes.
3. Right-click on the displayed list and select **New**.
4. Enter the short code settings as follows:
   - **Short Code:** 3XXX
     This will match any four-digit number beginning with 3.
   - **Telephone Number:** .
     The . indicates that the short code should output the digits as dialed.
   - **Line Group ID:** 3000
     This should match the Outgoing Group ID given to the system SCN line connected to System B.
   - **Feature:** Dial
5. Click OK.
6. If the only change made to the configuration was this short code, load the new configuration using merge, otherwise load the configuration and reboot.
7. A similar system short code can be added to System B's configuration to route 2XXX dialing to System A.

Chapter 12.
Licences

12. Licences

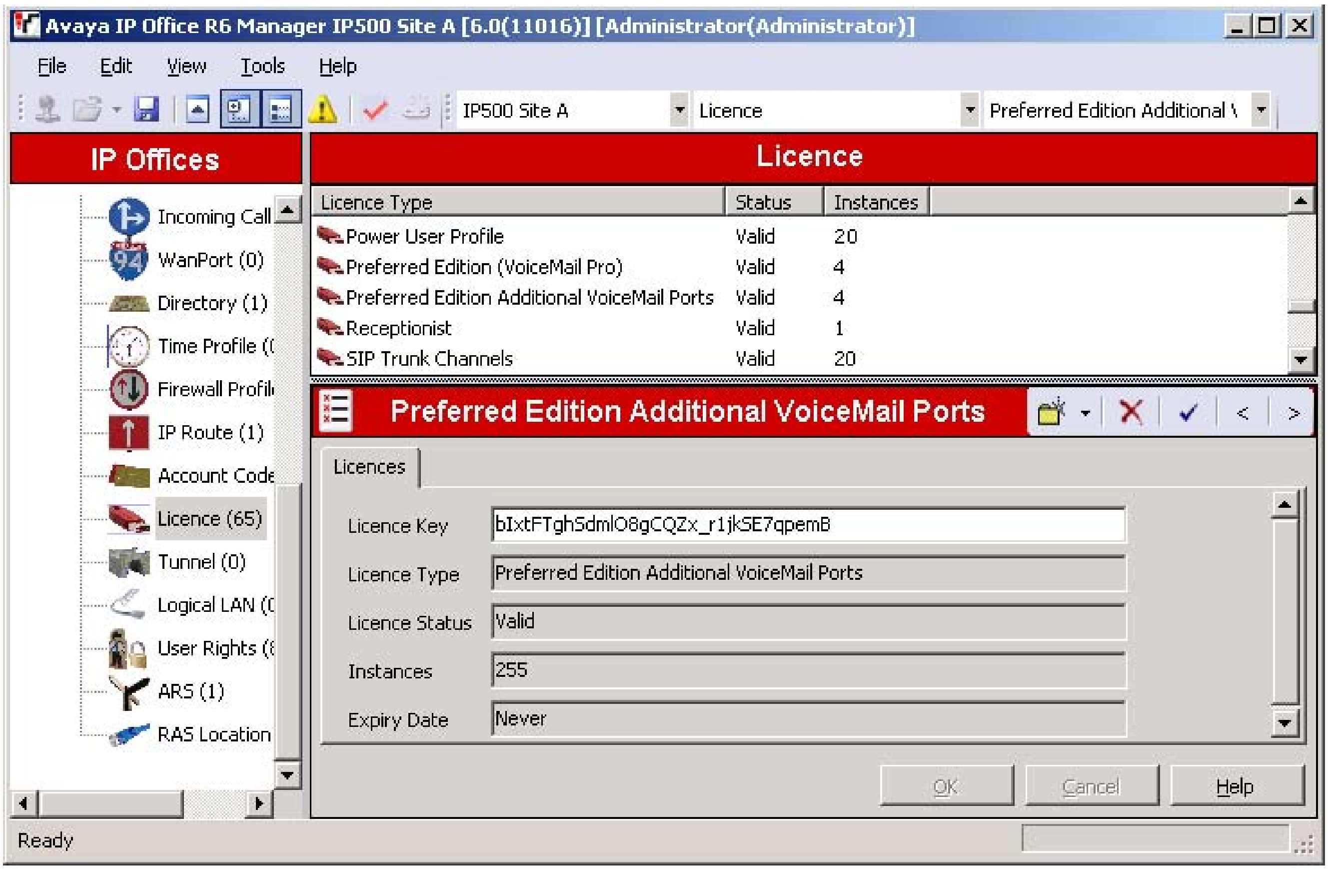
Note that this section covers just the licensing for IP Office 6.1 systems.

Various system features and applications require entry of license keys into the system's configuration. The license keys are unique 32-character codes based on the feature being activated and the serial number of the Feature Key dongle being used by the system.

The serial number is printed on the feature key dongle and prefixed with SN (FK for IP500v2 SD card dongles). It can also be viewed in the system configuration by selecting **System | System | Dongle Serial Number**.

For IP406 V2 and IP412 systems, the Feature Key dongle takes the form of a device attached to the serial port of the control unit. This key is only required if licensed features are needed.

For IP500 and IP500v2 systems, the Feature Key dongle takes the form of a card (smart media or SD card respectively) inserted into the control unit. The card is a mandatory item for these systems even if they use no licensed features.



- **Example 1: Enabling Features**
  In the example above, the system has a valid **Power User Profile** license. In this case the license is for 20 instances. That means that up to 20 users can have their Profile set to Power User. This allows them to be configured for a range of features not available to other users.

- **Example 2: Enabling Applications and Features**
  In the example above the system also has a **Preferred Edition (Voicemail Pro)** license. This licenses enables a range of features including support for the Voicemail Pro application and 4 ports between the system and the voicemail server. Additional ports have also been added using an **Preferred Edition Additional Voicemail Ports** license.

When a license key is entered into the system configuration, the following information is shown.

- Status
  The status, which is Unknown until the configuration file is sent back to the system.
  - *Unknown*
    This status is shown for licenses that have just been added to the configuration shown in Manager. Once the configuration has been sent back to the system and then reloaded, the status will change to one of those below.
  - *Valid*
    The features licensed can be configured and used.
  - *Invalid*
    The license was not recognized. It did not match the serial number of the Feature Key.
  - *Dormant*
    The license is valid but is conditional on some other pre-requisite licenses.
  - *Obsolete*
    The license is valid but is one no longer used by the level of software running on the system.
- License
  The name of the licensed feature. This may differ from the ordered RFA name.
- Instances
  Depending on the license, this may be the number of ports enabled or number of simultaneous users of the licensed feature. Sometime the number of instances is specified in the license name.
- Expires
  Most purchased licenses have no expiry setting. For some features, trial licenses may be available which will have an expiry date.

12.1 System Edition Licenses

These license are used to set what range of features the system supports.

- Essential Edition
  Any system running IP Office 5 or higher software is licensed by default for Essential Edition mode (no license for this is shown in the configuration). The following additional licenses can be added to enable additional services and features.
  - Essential Edition Additional Voicemail Ports : *IPO LIC R6 ESSNTL EDDITION ADD 2CH - 229423.*
    For IP500v2 only. By default the mandatory System SD card is also used to provide 2 ports of embedded voicemail. This license can be used to add additional ports up to the maximum of 6.
- Preferred Edition (Voicemail Pro) : *IP400 LIC PREFRD (VMPRO) - 171991.*
  This license enables support for Voicemail Pro as the system's voicemail server with 4 voicemail ports. The Preferred Edition license allows the voicemail server to provide the services listed below. Additional license can be added for additional voicemail features, these are detailed separately. This license was previously called Voicemail Pro (4 ports).
  - Mailboxes for all users and hunt groups.
  - Announcements for users and hunt groups.
  - Customizable call flows.
    - Call recording to mailboxes.
  - Campaigns.
  - TTS email reading for users licensed to Mobile Worker or Power User profiles.
  - Use of Conference Meet Me functions on IP500 and IP500v2 systems.
- Advanced Edition
  This license enables the additional features listed below. A Preferred Edition license is a pre-requisite for this license.
  - *IPO LIC R6 ADV EDITION RFA LIC:DS - 229424.*
  - *IPO LIC R6 ADV EDITION TRIAL RFA LIC:DS - 229425.*
  - Support for Customer Call Reporter including 1 supervisor.
  - Voicemail Pro database interaction (IVR).
  - Voicemail Pro call flow generic TTS (8 ports).[1]
  - Voicemail Pro Visual Basic Scripting.
  - Voicemail Pro call recording to ContactStore.[2]

  1. Provides up to 8 ports of TTS for use with Speak Text actions within Voicemail Pro call flows. Not used for user TTS email reading.
  2. Note: In a Small Community Network using centralized voicemail, this license only enables ContactStore support for the central system. Remote systems in the network require their own Advanced Edition license or a VMPro Recordings Administrator license.

Upgrade Licenses

Existing systems being upgraded to IP Office 6.0 or higher may require a software upgrade license.

- New IP500v2 Systems
  For the first 90 days, a new IP500v2 control unit will run any IP Office 6.0 or higher without requiring an upgrade license. The highest level run is written into the system's memory (not the SD card) and that becomes a permanent entitlement for the control unit. However, after 90 days the IP500v2 will require an upgrade license if upgraded to a software release higher than any that it has ran in the initial 90 day period.
- Software Upgrade
  Existing systems being upgraded to IP Office 6.0 or higher require an upgrade license. This applies to all system modes: IP Office Standard Version, IP Office Essential Edition - PARTNER® Version and IP Office Essential Edition - Norstar Version.
- Warning
  Systems upgraded without the appropriate license will display *"No license available"* and will not allow any telephony functions. There are two types of upgrade licenses as follows:
  - Small System Upgrade Licenses: *IPO LIC UPG R7.0 SML - 262645.*
    This license can be used to upgrade systems with up to 32 users and no external expansion modules.
  - Large System Upgrade Licenses: *IPO LIC UPG R0.0 - 262644.*
    This license can be used to upgrade system with more than 32 users or with external expansion modules.

12.2 Trunk Licensing

- IP500 Universal PRI (Additional channels)
  These licenses are used to enable additional B-channels above the basic 8 on an IP500 PRI-U card. The IP500 PRI-U card supports E1, T1 and E1-R2 PRI modes. The system supports 8 unlicensed B-channels on each IP500 PRI-U port fitted. Additional B-channels, up to the capacity of ports installed and PRI mode selected require licenses. These additional channels consume the licenses based on which additional channels are configured as in-service from port 9 of slot 1 upwards. D-channels are not affected by licensing.
    - *IP500 T1 CHANNELS ADD 2CH - 215180.*
    - *IP500 T1 CHANNELS ADD 8CH - 215181.*
    - *IP500 T1 CHANNELS ADD 32CH - 215182.*
    - *IP500 E1 CHANNELS ADD 2CH - 215183.*
    - *IP500 E1 CHANNELS ADD 8CH - 215184.*
    - *IP500 E1 CHANNELS ADD 22CH - 215185.*
    - *IP500 E1R2 CHANNELS ADD 2CH - 215186.*
    - *IP500 E1R2 CHANNELS ADD 8CH - 215187.*
    - *IP500 E1R2 CHANNELS ADD 22CH - 215188.*
- SIP Trunk Channels
  These licenses are used to configure the maximum number of simultaneous SIP trunk calls supported.
    - *IPO LIC SIP TRNK RFA 1 - 202967.*
    - *IPO LIC SIP TRNK RFA 5 - 202968.*
    - *IPO LIC SIP TRNK RFA 10 - 202969.*
    - *IPO LIC SIP TRNK RFA 20 - 202970.*
- IP500 Voice Networking
  These licenses are used with the IP500 and IP500v2 systems to enable support for SCN, QSIG and H323 IP trunks.
    - *IP500 VOICE NTWKG ADD LIC RFA - 205650.*
  - For IP Office 5, the additional ports license can be used without requiring a base license first.

12.3 Telephone/Endpoint Licenses

The use of H323 and SIP phones with IP Office 6.0 is controlled by the following licenses. Different licenses are used for Avaya IP phones, non-Avaya phones (including non-Avaya softphones) and Avaya softphones.

For details of the Phone Manager Pro PC SoftPhone refer to the Other Licenses section 842.

- Avaya IP Endpoints License
  On IP500 and IP500v2 systems, these licenses are used to license additional Avaya IP phones. This includes all 1600, 4600, 5600, 9600, IP DECT, DECT R4, T3 IP, Spectralink and VPN phones supported by IP Office 6.0
    - *IPO LIC R6 AV IP ENDPOINT 1 - 229444.*
    - *IPO LIC R6 AV IP ENDPOINT 5 - 229445.*
    - *IPO LIC R6 AV IP ENDPOINT 20 - 229447.*
    - *IPO LIC R6 AV IP ENDPOINT 5 TRIAL - 229449.*
  - The system will automatically license 12 Avaya IP phones for each IP500 VCM 32 or VCM 64 card installed in the system without requiring additional licenses to be added to the configuration.
  - Additional Avaya IP phones are licensed either by the addition of Avaya IP Endpoints licenses above or the conversion of legacy IP500 VCM Channels licenses to Channel Migration licenses (see below).
  - By default licenses are consumed by each Avaya IP phone that registers with the system in the order that they register. The license is released if the phone unregisters. However, it is possible to reserve a license for particular phones in order to ensure that those phones always obtain a license first if available. This is done through the Reserve Avaya IP Endpoint Licence setting of each IP extension.
  - Avaya IP phones without a license will still be able to register but will be limited to making emergency calls only (Dial Emergency short code calls). The associated user will be treated as if logged off and the phone will display *"No license available"* and *"Emergency Calls Only"*. If a license becomes available, it will be assigned to any unlicensed DECT handsets first and then to any other unlicensed Avaya IP phone in the order that the phones registered.
  - For existing IP500 systems being upgraded to IP Office 6.0, the existing VCM channels and IP500 VCM Channels license are treated as follows:
    - For each IP400 VCM card installed in the system, each VCM channel supported by the card allows support for 3 Avaya IP phones.
    - For each IP500 VCM32 and IP500 VCM64 card installed in the system, the 4 unlicensed VCM channels previously provided by each card are converted to allow unlicensed support of 12 Avaya IP phones.
    - For each legacy IP500 VCM Channels license, the license are converted Channel Migration licenses supporting 3 Avaya IP phones. See the Channel Migration license below.
    - The IP500 VCM 32 and IP500 VCM 64 cards will provide their full capacity of VCM channels, ie. providing up to 32 or 64 channels depending on the card type and the codecs being used.
- 3rd Party IP Endpoints License
  These licenses are used for support of non-Avaya IP phones including SIP extensions. The available license are used in order of phone registration. If no licenses are available the phone will not be able to register. Available licenses can reserved for a particular phone using the Reserve 3rd Party IP Endpoint License setting in the each IP extension. This license was previously called the IP End-points license.
    - *IP400 IP ENDPOINTS RFA 1 LIC - 174956.*
    - *IP400 IP ENDPOINTS RFA 5 LIC - 174957.*
    - *IP400 IP ENDPOINTS RFA 20 LIC - 174959.*

Legacy Endpoint Licenses

- Channel Migration
  These licenses were previously called IP500 VCM Channels. In pre-IP Office 6.0 systems, these license were used to enable additional VCM channels on IP500 VCM32 and IP500 VCM 64 cards. For IP Office 6.0 these license are no longer required. Any present in the configuration of systems upgraded to IP Office 6.0 are renamed Channel Migration. Each Channel Migration license instance enables support for 3 Avaya IP phones.
    - *IPO LIC IP500 VCM LIC 4 CH LIC - 202961.*
    - *IPO LIC IP500 VCM LIC 8 CH LIC - 202962.*
    - *IPO LIC IP500 VCM LIC 16 CH LIC - 202963.*
    - *IPO LIC IP500 VCM LIC 28 CH LIC - 202964.*
    - *IPO LIC IP500 VCM LIC 60 CH LIC - 202965.*

12.4 User Licenses

The features available to a basic user can be enhanced by the addition of 'User Profile' licenses. Once these licenses are present in the system configuration, the profiles can be applied to selected users through the User | User | Profile setting in the system configuration.

	Basic User	Office Worker	Teleworker	Mobile Worker	Power User
one-X Portal for IP Office	Yes[1]	Yes	Yes	–	Yes
" Telecommuter options	Yes[1]	–	Yes	–	Yes
UMS Web Services	Yes[1]	Yes	Yes	–	Yes
Mobility Features	Yes[1]	–	–	Yes	Yes
TTS for Email Reading	–	–	–	Yes	Yes
IP Office SoftPhone	–	–	Yes	–	Yes

1. These features are supported for Basic User users on systems with the appropriate pre-IP Office 6.0 legacy licenses.

- Teleworker Profile License
  These licenses set the number of users who can have their profile set as *Teleworker*. For user with this optional, additional settings are enabled in the configuration for the following services: one-X Portal for Manager with Telecommuter option UMS Web Services and Manager SIP SoftPhone.
  - *IPO LIC R6 TELEWORKER 1 - 229430.*
  - *IPO LIC R6 TELEWORKER 5 - 229431.*
  - *IPO LIC R6 TELEWORKER 20 - 229432.*
  - *IPO LIC R6 TELEWORKER 5 TRIAL - 229433.*

- Mobile Worker Profile License
  Thee licenses set the number of users who can have their profile set as Mobile Worker. For user with this optional, additional settings are enabled in the configuration for the following services: Mobility Features and TTS for Email Reading.
  - *IPO LIC R6 MOBILE WORKER 1 - 229434.*
  - *IPO LIC R6 MOBILE WORKER 5 - 229435.*
  - *IPO LIC R6 MOBILE WORKER 20 - 229436.*
  - *IPO LIC R6 MOBILE WORKER 5 TRIAL - 229437.*

- Office Worker Profile License
  These licenses set the number of users who can have their profile set as Office Worker. For user with this optional, additional settings are enabled in the configuration for the following services: one-X Portal for Manager (no telecommuter features) and UMS Web Services. If no *Office Worker Profile* licenses are present, existing legacy *Phone Manager Pro (per seat)* licenses can be used to enable users for the Office Worker profile.
  - *IPO LIC R6 OFF WORKER RFA 1 - 229438.*
  - *IPO LIC R6 OFF WORKER RFA 5 - 229439.*
  - *IPO LIC R6 OFF WORKER RFA 20 - 229440.*
  - *IPO LIC R6 OFF WORKER 5 TRIAL - 229441.*

- Power User Profile License
  These licenses set the number of users who can have their profile set as Power User. For user with this optional, the same additional services as for Teleworker and Mobile Worker are enabled for the user in the configuration plus the following service: SoftPhone.
  - *IPO LIC R6 PWR USER 1 - 229426.*
  - *IPO LIC R6 PWR USER 5 - 229427.*
  - *IPO LIC R6 PWR USER 20 - 229428.*
  - *IPO LIC R6 PWR USER 5 TRIAL - 229429.*

- Receptionist: *IPO LIC RECEPTIONIST RFA - 171987.*
  This license is used to enable support for the SoftConsole application. This license can only be used by users set to Receptionist in the configuration. A maximum of 4 receptionist are supported. This license was previously called SoftConsole.
  - For IP Office 6.0 and 6.1, an instance of this license is consumed by each user configured as a Receptionist. If the user hot desks to another system in an SCN, their license entitlement is retained, ie. the remote system does not require a Receptionist license.

Legacy User Licenses

The following licenses are no longer available from Avaya but are still supported for systems upgraded to IP Office 6.0

- Mobility Features
  These legacy licenses were used to enable mobility features, for example mobile twinning or mobile call control, for users set to the *Basic User* profile.
    - *IPO LIC MOBILE WORKER RFA 1 - 195569.*
    - *IPO LIC MOBILE WORKER RFA 5 - 195570.*
    - *IPO LIC MOBILE WORKER RFA 20 - 195572.*
- one-X Portal for IP Office
  These legacy licenses were used to enable one-X Portal for IP Office support for users set to the *Basic User* profile. The licenses were purchased as part of the IP Office 5 Power User license packages.
- UMS Web Services
  These licenses are used to enable UMS voicemail services support for users set to the *Basic User* profile. Other users are enabled for UMS through their licensed user profile. These licenses are also used to license hunt groups for UMS voicemail services.
    - *IPO LIC VMPRO UMS 1 USER - 217880.*
    - *IPO LIC VMPRO UMS 5 USER - 217881.*
    - *IPO LIC VMPRO UMS 20 USER - 217883.*

12.5 Voicemail Pro Licenses

For IP Office 6.0, support for Voicemail Pro is enable by the addition of a Preferred Edition license.

- Preferred Edition (Voicemail Pro) : IP400 LIC PREFRD (VMPRO) - 171991.
  This license enables support for Voicemail Pro as the system's voicemail server with 4 voicemail ports. The Preferred Edition license allows the voicemail server to provide the services listed below. Additional license can be added for additional voicemail features, these are detailed separately. This license was previously called Voicemail Pro (4 ports).
  - Mailboxes for all users and hunt groups.
  - Announcements for users and hunt groups.
  - Customizable call flows.
    - Call recording to mailboxes.
  - Campaigns.
  - TTS email reading for users licensed to Mobile Worker or Power User profiles.
  - Use of Conference Meet Me functions on IP500 and IP500v2 systems.
- Advanced Edition
  This license enables the additional features listed below. A Preferred Edition license is a pre-requisite for this license.
  - *IPO LIC R6 ADV EDITION RFA LIC:DS - 229424.*
  - *IPO LIC R6 ADV EDITION TRIAL RFA LIC:DS - 229425.*
  - Support for Customer Call Reporter including 1 supervisor.
  - Voicemail Pro database interaction (IVR).
  - Voicemail Pro call flow generic TTS (8 ports).[1]
  - Voicemail Pro Visual Basic Scripting.
  - Voicemail Pro call recording to ContactStore.[2]

  1. Provides up to 8 ports of TTS for use with Speak Text actions within Voicemail Pro call flows. Not used for user TTS email reading.
  2. Note: In a Small Community Network using centralized voicemail, this license only enables ContactStore support for the central system. Remote systems in the network require their own Advanced Edition license or a VMPro Recordings Administrator license.
- Preferred Edition Additional Voicemail Ports
  The required license for Voicemail Pro server support (Preferred Edition (Voicemail Pro)) also enables 4 voicemail ports. These licenses can be used to add additional voicemail ports up to the maximum capacity of the system (IP406 V2 = 20, IP412 = 30, IP500 = 40, IP500v2 =40). This license was previously called Additional Voicemail Pro (ports).
  - *IPO LIC VM PRO RFA 2 LIC - 174459.*
  - *IPO LIC VM PRO RFA 4 LIC - 174460.*
  - *IPO LIC VM PRO RFA 8 LIC - 174461.*
  - *IPO LIC VM PRO RFA 16 LIC - 174462.*
- VMPro Recordings Administrators : *IPO LIC CONTACTSTORE RFA LIC - 187166.*
  To support ContactStore in a Small Community Network, systems other than the central system require either their own Advanced Edition license or this license.
- VMPro Networked Messaging : *IPO LIC NTWKD MSGING RFA LIC - 182297.*
  Enables VPNM (Voicemail Pro Networked Messaging) functionality within Voicemail Pro. This allows message exchange with remote Voicemail Pro systems and Avaya Interchange systems.
- VMPro TTS (Generic) : *IP400 3rd PARTY TTS LIC RFA - 182303.*
  This legacy license enables use of text to speech facilities using third party TTS software with Voicemail Pro. One license per simultaneous instance of TTS usage. For IP Office 6.0 this license is no longer used for user email reading. The IP Office Advance Edition license also enables 8 ports of generic TTS.
- VMPro TTS (ScanSoft) : *IPO LIC AVAYA TTS RFA 1 - 182299.*
  This legacy licence enables use of text to speech facilities using Avaya supplied TTS software with Voicemail Pro. One license per simultaneous instance of TTS usage. For IP Office 6.0 this license is no longer used for user email reading.
- UMS Web Services
  These licenses are used to enable UMS voicemail services support for users set to the *Basic User* profile. Other users are enabled for UMS through their licensed user profile. These licenses are also used to license hunt groups for UMS voicemail services.
  - *IPO LIC VMPRO UMS 1 USER - 217880.*
  - *IPO LIC VMPRO UMS 5 USER - 217881.*
  - *IPO LIC VMPRO UMS 20 USER - 217883.*

Legacy Voicemail Licenses

The following legacy licenses are still supported by IP Office 6.0

- UMS Web Services
  These licenses are used to enable UMS voicemail services support for users set to the *Basic User* profile. Other users are enabled for UMS through their licensed user profile. These licenses are also used to license hunt groups for UMS voicemail services.
    - *IPO LIC VMPRO UMS 1 USER - 217880.*
    - *IPO LIC VMPRO UMS 5 USER - 217881.*
    - *IPO LIC VMPRO UMS 20 USER - 217883.*
- VMPro Database Interface : *IP400 3RD PARTY IVR LIC RFA - 182298.*
  This legacy license enables 3rd party database support within Voicemail Pro call flows. For IP Office 6.0 this is also enabled by the Advanced Edition license.
- VMPro VB Script : *IP400 VB SCRIPTING LIC RFA - 182300.*
  This legacy license enables Visual Basic Script support with Voicemail Pro. For IP Office 6.0 this is also enabled by the Advanced Edition license.

12.6 Customer Call Reporter Licenses

For IP Office 6.0, support for the Customer Call Reporter application is enabled by the presence of Preferred Edition and Advanced Edition licenses in the system configuration. For system being upgraded to IP Office 6.0, Customer Call Reporter can alternately be enabled by a legacy CCR Sup license or a CCC Supervisor and CCR CCC Upg license.

- Preferred Edition (Voicemail Pro) : *IP400 LIC PREFRD (VMPRO) - 171991.*
  This license enables support for Voicemail Pro as the system's voicemail server with 4 voicemail ports. The Preferred Edition license allows the voicemail server to provide the services listed below. Additional license can be added for additional voicemail features, these are detailed separately. This license was previously called Voicemail Pro (4 ports).
  - Mailboxes for all users and hunt groups.
  - Announcements for users and hunt groups.
  - Customizable call flows.
    - Call recording to mailboxes.
  - Campaigns.
  - TTS email reading for users licensed to Mobile Worker or Power User profiles.
  - Use of Conference Meet Me functions on IP500 and IP500v2 systems.
- Advanced Edition
  This license enables the additional features listed below. A Preferred Edition license is a pre-requisite for this license.
  - *IPO LIC R6 ADV EDITION RFA LIC:DS - 229424.*
  - *IPO LIC R6 ADV EDITION TRIAL RFA LIC:DS - 229425.*
  - Support for Customer Call Reporter including 1 supervisor.
  - Voicemail Pro database interaction (IVR).
  - Voicemail Pro call flow generic TTS (8 ports).[1]
  - Voicemail Pro Visual Basic Scripting.
  - Voicemail Pro call recording to ContactStore.[2]

  1. Provides up to 8 ports of TTS for use with Speak Text actions within Voicemail Pro call flows. Not used for user TTS email reading.
  2. Note: In a Small Community Network using centralized voicemail, this license only enables ContactStore support for the central system. Remote systems in the network require their own Advanced Edition license or a VMPro Recordings Administrator license.
- Customer Service Agent
  These licenses enable the configuration of users as CCR agents. Multiple license can be added for up to the maximum of 150 agents. A license is consumed for each CCR agent logged in. If no more license are available, further agents cannot log in. This license was previous called CCR Agent.
  - *IPO LIC CUSTMR SVC AGT RFA 1 - 217650.*
  - *IPO LIC CUSTMR SVC AGT RFA 5 - 217651.*
  - *IPO LIC CUSTMR SVC AGT RFA 20 - 217653.*
- Customer Service Supervisor
  This license is used to enable support for CCR supervisor and wallboard accounts. Each license instance enables both 1 supervisor account and 1 wallboard account. Multiple license can be added for up to 30 supervisors/ wallboards.
  - *IPO LIC R6 CUSTMR SVC SPV 1 - 229442.*
  - *IPO LIC R6 CUSTMR SVC SPV 1 TRIAL - 229443.*

Legacy CCR Licenses

- CCR Sup
  These legacy licenses were used to enable support for the Customer Call Reporter application and CCR supervisors.
    - *IPO CUSTMR CALL REPORTER 1 SPV LIC RFA - 217655.*
    - *IPO CUSTMR CALL REPORTER 10 SPV LIC RFA - 217656.*
    - *IPO CUSTMR CALL REPORTER 20 SPV LIC RFA - 217657.*
- CCR CCC Upg
  This license allows legacy CCC application licenses to be used for Customer Call Reporter.
    - *IPO LIC CUSTMR CALL REPORTER UPG LIC RFA - 217658.*

  - CCC Server - Enables 1 supervisor, 1 wallboard and 5 agents.
  - CCC Supervisors - Enables the equivalent number of supervisors and wallboards.
  - CCC Agents - Enables the equivalent number of agents.

12.7 Trial Licenses

The following trial licenses can be requested. Each is valid for 60 days from the date of issue and can only be issued once for a particular system Feature Key serial number. Apart from that restriction the trial license works the same as a full license.

- Preferred Edition: *IPO LIC PREFERRD (VM PRO) TRIAL RFA LIC:DS (189782).*
- Advanced Edition: *IPO LIC R6 ADV EDITION TRIAL - 229425.*
- Power User (5 Users): *IPO LIC R6 PWR USER 5 TRIAL - 229429.*
- Teleworker (5 Users): *IPO LIC R6 TELEWORKER 5 TRIAL - 229433.*
- Mobile Worker (5 Users): *IPO LIC R6 MOBILE WORKER 5 TRIAL - 229437.*
- Office Worker Profile (5 Users): *IPO LIC R6 OFFICE WORKER 5 TRIAL - 229441.*
- Customer Service Agent: *IPO LIC CUSTMR SVC AGT RFA TRIAL 5 - 227053.*
- Customer Service Supervisor Profile: *IPO LIC R6 CUSTMR SVC SPV 1 TRIAL - 229443.*
- Avaya IP Endpoints (5 Extensions): *IPO LIC R6 AVAYA IP ENDPOINT 5 TRIAL - 229449.*
- Receptionist (Users): *IPO LIC RECEPTIONIST RFA 1 TRIAL LIC:CU - 189783.*
- VMPro Networked Messaging: *IPO LIC NTWKD MSGING TRIAL RFA LIC:DS - 189776.*
- VMPro TTS (ScanSoft): *IPO LIC AVAYA TTS TRIAL RFA 1 LIC:CU - 189778.*
- VM Pro TTS (Generic): *IPO LIC 3RD PRTY TTS TRIAL RFA LIC:CU -189781.*
- Audix Voicemail: *IPO LIC ACM CENTRAL VM TRIAL LIC:DS - 189786.*
- IPSec Tunneling: *IPO LIC IPSec VPN RFA TRIAL LIC:DS - 189806.*
- SIP Trunk Channels: *IPO LIC SIP TRUNKING TRIAL RFA 5 - 205820.*
- IP500 Voice Networking: *IPO LIC IP500 VCE NTWK ADD 4 TRIAL - 205823.*
- CTI Link Pro : *IPO LIC CTI RFA TRIAL - 263128*.

12.8 Other Licenses

- Audix Voicemail : *IPO LIC ACM CENTRAL VM - 177467.*
  Enables the system to use a remote Intuity Audix or Modular Messaging system for voicemail rather than requiring a local voicemail server.

- IPSec Tunneling : *IPO LIC IPSEC VPN RFA - 182301.*
  Enables the system to initiate and terminate IPSec and L2TP tunnels.

Phone Manager Licenses

These licenses are used for the Phone Manager application. In addition to entering Phone Manager licenses, each user is individually configured for the expected Phone Manager type.

- Phone Manager Pro (per seat): *IPO LIC PMGR PRO RFA 1 - 177468.*
  Allows users to be configured as Phone Manager Pro users. The user's Phone Manager mode is set through the system configuration (User | Telephony | Phone Manager Type).
- Phone Manager Pro IP Audio Enabled (per user): *IPO LIC PMGR SFTPHN RFA 1 - 171992.*
  Enables Phone Manager Pro IP softphone operation for a user. Note: Also requires the user to have a Phone Manager Pro license.

CTI Licenses

- CTI Link Pro : *IPO LIC CTI RFA - 171988*
  Enables CTI Link Pro functionality (TAPI Link Pro and DEVLink Pro).

- Wave User : *IPO LIC TAPI WAV RFA 4 - 177466*
  Allows streaming of WAV files, using TAPILink Pro, for 3rd party voice applications. This is a per user license. Note that TAPI WAV calls use system data channels taken from the same pools as used for voicemail ports. The maximum number of simultaneous TAPI WAV user calls and voicemail users is determined by the control unit type; IP406 V2 = 20, IP412 = 30, IP500 = 40, IP500v2 =40.

Chapter 13.
Appendix: Locale Settings

13. Appendix: Locale Settings

The system Locale sets factors such as the default ringing tones and caller display settings. The locale also controls the default language that the system voicemail server will use for prompts.

Users and incoming call routes can also be set to a locale. That locale will then override the system settings for calls to voicemail.

This following table indicates locale settings used for different functions. Note that reference to a locale does not necessarily indicate support, availability or approval for system within that country.

Argentina 849 *(ess)*	France 863 *(fra)*	New Zealand 876 *(enz)*	Sweden 889 *(sve)*
Australia 850 *(ena)*	Germany 865 *(deu)*	Norway 877 *(nor)*	Switzerland (French) 890 *(frs)*
Bahrain 851 *(arh)*	Greece 864 *(ell)*	Oman 878 *(aro)*	Switzerland (German) 890 *(des)*
Belgium 852 *(nlb)*	Hong Kong 866 *(zhh)*	Pakistan 879 *(urd)*	Switzerland (Italian) 890 *(its)*
Belgium 853 *(frb)*	Hungary 867 *(hun)*	Peru 880 *(esr)*	Taiwan 891 *(cht)*
Brazil 854 *(ptb)*	Iceland 868 *(isl)*	Poland 881 *(plk)*	Turkey 892 *(trk)*
Canada 855 *(frc)*	India 869 *(ind)*	Portugal 882 *(ptg)*	United Arab Emirates 893 *(aru)*
Chile 856 *(esl)*	Italy 870 *(ita)*	Qatar 883 *(arq)*	United Kingdom 894 *(eng)*
China 857 *(chs)*	Korea 871 *(kor)*	Russia 884 *(rus)*	USA 895 *(enu)*
Colombia 858 *(eso)*	Kuwait 872 *(ark)*	Saudi Arabia 885 *(ara)*	Venezuela 896 *(esv)*
Denmark 860 *(dan)*	Mexico 873 *(esm)*	Singapore 886 *(zhi)*	
Egypt 861 *(are)*	Morocco 874 *(arm)*	South Africa 887 *(ens)*	
Finland 862 *(fin)*	Netherlands 875 *(nld)*	Spain 888 *(esp)*	

* Locale defaults to best match. For example New Zealand (enz) falls back to UK English (eng). Embedded voicemail supports simple fallback based on the first two letters of the locale TLA. Voicemail Pro employs multiple fallback always ending in UK or US English.

- TLA (Three-Letter Abbreviation):
  These are the three character codes used by pre-6.0 Manager to set locales. In IP Office 3.2 they were replaced by selection of the required country or language by name.
- Locale:
  The country represented by the locale. Teletype (Textphone) is used with Voicemail Pro, refer to the Voicemail Pro documentation.
- Language:
  The voicemail prompt language used for that locale.
- Manager:
  Indicates that the Manager application can run in the specific locale language. Manager uses the best match it has (French, German, Brazilian, Dutch, Italian, Mexican Spanish or US English) for the regional location settings setting of the PC on which it is running, otherwise it defaults to UK English. If required the language used within the Manager screens can be overridden, see Changing the Manager Language 23.
- Telephony:
  The system provides default telephony settings matching the normal expected defaults for the locale.
- Phone Display:
  Indicates that display messages from the system to Avaya phones can be sent using the appropriate language for that locale. Note that the user locale can be used to override the system locale for these messages. Note also some strings displayed on a phone may come from the phone's own software and may differ. Also refer to LPhone Display Language Sets and Fallback 848 for details of how the system loads and resolves phone display strings.
- T3 Phones:
  Menus for T3 Series phones available in the specific language.
- Voicemail:
  These columns indicate for which locales the different voicemail servers can provide the appropriate language prompts. In all cases, the system locale can be overridden by setting a different user locale.
  - EVM:
    Indicates that the locale is recognized by Embedded Voicemail and appropriate language prompts are then used. If an unsupported locale is used, Embedded Voicemail will attempt the best match using the first two characters of the locale.
  - VM Lite: *(Voicemail Lite is not supported on IP Office 5 and higher)*
    Indicates that the locale is recognized by Voicemail Lite and appropriate language prompts are then used. For an unsupported locale is used, or one for which the necessary prompts are not available, Voicemail Lite will attempt the best match using a sequence of alternate locales.

- VM Pro:
  Indicates that the locale is recognized by Voicemail Pro and appropriate language prompts are then used. For an unsupported locale is used, or one for which the necessary prompts are not available, Voicemail Pro will attempt the best match using a sequence of alternate locales. For example French Canadian (frc) fallback to French (fra), then US English (enu) and finally UK English (eng). Note that the languages available are selectable during Voicemail Pro installation. For further details refer to the Voicemail Pro manual.

- All Voicemail
  For calls to voicemail, the locale that is passed to voicemail to determine the prompt to play (if available) is:

  - The user locale, if set, is used if the caller is internal.
  - The incoming call route locale, if set, is used if caller is external.
  - If the possible locales above are not set, the system locale is used.
  - The short code locale, if set, is used and overrides the options above if the call is routed to voicemail using the short code.
  - If the locale requested is not matched by prompts available on the voicemail server, the method of prompt fallback depends on the voicemail type. Refer to the documentation for Embedded Voicemail, Voicemail Lite or Voicemail Pro.

13.1 Locale Defaults

A Locale being covered in this document does not imply approvals or availability.

- Tones
  The table below describes the different system tones. The tones used are determined by the system locale setting. Note that in some locales, the same tone sound may be used for several purposes, for example for Busy and Fast Busy may be the same.

Tone	Description
Dial Tone	Normal dial tone.
Alternate Dial Tone	This tone is also known as 'interrupted', 'broken' or 'stutter' dial tone. It is used on extensions when the user has DND, Follow-Me or Forward Unconditional set. It is also used on analog phones when the user has a call on hold, for example during a transfer.
Secondary Dial Tone	Used when accessing an external trunk using a short code or ARS that specifies secondary dial tone. If no specific tone is defined for the locale, normal dial tone is used.
Busy Tone	Used when the number called is busy.
Fast Busy Tone	No channel.
Intercept Busy Tone	Unallocated number.
Ring Tone	Other end is ringing. This tone is also known as 'ringback'.
Call Waiting Tone	Used when a user has a call waiting enable and a is call waiting.
NU Tone	Number Unobtainable.

- Frequency
  All tone frequencies are in Hertz (Hz). Where a tone uses a combination of frequencies, the frequencies are shown separated by a + symbol. Where a tone uses alternating frequencies, the frequencies are shown separated by a / symbol.

- Cadence
  The tone cadence is indicated as either a sequence of on/off times or as a sequence of alternating frequency 1/ frequency 2 times. Where a portion of the sequence is enclosed in ( ) symbols, it indicates a repeating sequence.

- POT Port Settings
  These settings are used for analog phone extensions. The settings used are determined by the system locale.

Analog Phone Settings	
Ring Current Frequency	The frequency of ring current.
Minimum Flash Hook Time	The minimum time that loop current has to be broken to detect a timed break recall (TBR). Anything shorter is regarded as a glitch.
Maximum Flash Hook Time	The maximum time the loop current can be broken for it to be detected as a time break recall (TBR). Anything longer is regarded as clearing.
Default Caller Display Type	The type of caller display used when Extn | Caller Display Type is set to *On*.
Default MWI Type	The type of message waiting indication used when Extn | Message Waiting Indication Type is set to *On*. Note that the setting may also be dependant on the equipment providing the analog port.

Locale Specific Behavior	
Companding	The typical companding method employed for telephone systems in that locale. Companding operation can be adjusted through the System | Telephony | Telephony 160 tab.
Dialing Timeout	The method used by the system to determine when to disconnect an extension that does not complete dialing a valid routable number.
Disconnect Tone	The tone used when disconnect indication is provided to an extension.

Display Language	The default language used for messages sent to extensions.
Feature Phone Clearing	The locale specific action applied to digital and IP phones when the far end of a call disconnects. This action can be overridden by the Disconnect Tone option on the System | Telephony | Telephony 160 tab.
Voicemail Truncation Time	On analog trunks, call disconnection can occur though busy tone detection. When such calls go to voicemail to be recorded or leave a message, when the call ends the system indicates to the voicemail system how much to remove from the end of the recording in order to remove the busy tone segment. For some systems it may be necessary to override the default if analog call recordings are being clipped or include busy tone. That can be done by adding a VM_TRUNCATE_TIME= setting with the required value in the range 0 to 7 seconds.
Default Currency	Used for Advice of Charge 740 operation where supported on ISDN lines.
Default Time Zone	This value is used to set the default time offset and daylight saving settings for a system when using SNTP to obtain the UTC time value.

13.2 Phone Display Language Sets and Fallback

The phone display strings loaded by the system are based on the system Locale setting of the system. These are the languages used for display information sent to the phone from the system. Individual phones may have support for differing languages through the phones own menus.

System Locale	Languages Loaded
Belgium (Flemish), Belgium (French), Denmark, Finland, France, German, Greece, Hungarian, Italian, Netherlands, Norway, Poland, Portugal, Spain, Sweden, United Kingdom.	Danish, Dutch, English (UK), Finnish, French, German, Italian, Norwegian, Portuguese, Spanish, Swedish.
Argentina, Brazil, Chile, Colombia, Mexico, Peru, Venezuela.	Brazilian Portuguese, English (US), Mexican Spanish.
Russia.	English (US), Finnish, French, Russian, Swedish.
Canada, United States.	English (US), French Canadian, Mexican Spanish.
All others.	English (UK), French, German, Portuguese, Russian, Spanish.

If the user locale setting differs from the system locale, the set of loaded languages are searched for the best match. If no match is possible then the loaded variant of English is used.

13.3 Argentina

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(2.0/4.0) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25) alternating tones.
Ring Tone	440+480	(1.0/3.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.070s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Latin Spanish
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	0 seconds.
Default Currency	ARS
Default Time Zone	UTC-03:00 - Buenos Aires

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Latin Spanish
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Latin Spanish
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.4 Australia

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.375/0.375) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.010s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	5 seconds.
Default Currency	AUD
Default Time Zone	UTC+10:00 - Canberra, Melbourne, Sydney

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.5 Bahrain

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	440	Continuous.
Alternate Dial Tone	440	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.4/0.35/0.225/0.525) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+03:00 -Abu Dhabi.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.6 Belgium - Flemish

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	(0.5/0.5) on/off.
Ring Tone	425	(1.0/3.0) on/off.
Call Waiting Tone	400	(0.08/0.175/0.08/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Dutch.
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+01:00 - Brussels

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Dutch
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Dutch
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.7 Belgium - French

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	(0.5/0.5) on/off.
Ring Tone	425	(1.0/3.0) on/off.
Call Waiting Tone	400	(0.08/0.175/0.08/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	French.
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+01:00 - Paris

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	French
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	French/ Canadian French
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	French
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.8 Brazil

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(2.0/4.0) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.25/0.25) on/off.
Fast Busy Tone	425	(0.25/0.25) on/off.
Intercept Busy Tone	425	(0.1/0.1/0.1/0.1) on/off.
Ring Tone	425	(1.0/4.0) on/off.
Call Waiting Tone	425	(0.06/0.25/0.06/5) on/off.
Number Unobtainable Tone	425	IP Office 4.2+: (0.25/0.25/0.75/0.25) on/off. Pre-IP Office 4.2: Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.100s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	DTMFD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	A-Law/Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Brazilian Portuguese
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	2 seconds.
Default Currency	BRL
Default Time Zone	UTC-03:00 - Brasilia

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Brazilian	Portuguese
Phone Display		
- DS Phones	✓	✓
- T3 Phones	✗	✗
Voicemail Prompts		
- Voicemail Pro	✓	✓
- Embedded Voicemail	✓	✓

Language	Brazilian	Portuguese
Manager	✓	✗
System Status Application	✓	✓
Phone Manager	✓	✓
Soft Console	✓	✗
one-X Portal for IP Office	✓	✓
IP Office Customer Call Reporter	✓	✓

13.9 Canada - French

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(0.25/0.25) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25) alternating tones.
Ring Tone	440+480	(2.0/4.0) on/off.
Call Waiting Tone	480+620	(0.0/0.1/0.2/1200) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Canadian French
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	0 seconds.
Default Currency	CAD
Default Time Zone	UTC-05:00 - Eastern Standard Time (EST)

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	French
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	French/ Canadian French
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	French
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.10 Chile

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(2.0/4.0) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25) alternating tones.
Ring Tone	480+620	(1.0/3.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Mexican Spanish.
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	2 seconds.
Default Currency	CLP
Default Time Zone	UTC-04:00 - Santiago

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Latin Spanish
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Latin Spanish
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.11 China

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(0.4/0.04) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	440	(0.35/0.35) on/off.
Fast Busy Tone	440	(0.7/0.7) on/off.
Intercept Busy Tone	440	(0.1/0.1/0.1/0.1/0.1/0.1/0.4/0.4) on/off.
Ring Tone	440	(1.0/4.0) on/off.
Call Waiting Tone	440	(0.4/4.0) on/off.
Number Unobtainable Tone	440	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	0.900s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	5 seconds.
Default Currency	CNY
Default Time Zone	UTC+08:00 - Beijing, Hong Kong.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Simplified Chinese	Madarin	Cantonese
Phone Display			
- DS Phones	✗	–	–
- T3 Phones	✗	–	–
Voicemail Prompts			
- Voicemail Pro	–	✓	✓
- Embedded Voicemail	–	✓	✓

Language	Simplified Chinese	Madarin
Manager	✓	–
System Status Application	✓	–
Phone Manager	✓	–
Soft Console	✓	–
one-X Portal for IP Office	✓	–
IP Office Customer Call Reporter	✓	–

13.12 Colombia

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(2.0/4.0) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25 alternating tones.
Ring Tone	440+480	(1.0/3.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Latin Spanish
Feature Phone Clearing	Disconnect tone.
Voicemail Truncation Time	2 seconds.
Default Currency	COP
Default Time Zone	UTC-05:00 - Bogota.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Latin Spanish
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Latin Spanish
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.13 Customize

IP Office 4.0 Q2 2007 maintenance release and higher: On the System | System tab, the Locale can be set to *Customize*. This option is intended for those locales which use a mix of telephony equipment from a range of other locales. Testing and use of this locale option is entirely the responsibility of the installer.

The Customize locale matches the Saudi Arabia 885 locale but with the following additional controls:

- Tone Plan: *Default = Tone Plan 1*
  The tone plan control dial and ringing tone. The options are:
  - *Tone Plan 1:* United States.
  - *Tone Plan 2:* United Kingdom.
  - *Tone Plan 3:* France.
  - *Tone Plan 4:* Germany.
  - *Tone Plan 5:* Spain.
- CLI Type: *Default = FSK V23*
  This is the method used for CLI signalling on analog lines.
- Busy Tone Detection: *Default = Off.*
  Enables or disables the use of busy tone detection for call clearing. This is a system wide setting.

13.14 Denmark

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.25/0.25) on/off.
Fast Busy Tone	425	(0.25/0.25) on/off.
Intercept Busy Tone	425	(0.25/0.25) on/off.
Ring Tone	425	(0.75/7.5) on/off.
Call Waiting Tone	425	(0.08/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	DTMFC
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Danish
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	DKK
Default Time Zone	UTC+01:00 - Coppenhagen.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Danish
Phone Display	
- DS Phones	✓
- T3 Phones	✗
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Danish
- Manager	✗
- System Status Application	✗
- Phone Manager	✓
- Soft Console	✓
- one-X Portal for IP Office	✗
- IP Office Customer Call Reporter	✗

13.15 Egypt

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	425+450	(1.0/4.0) on/off.
Fast Busy Tone	425+450	(0.5/0.5) on/off.
Intercept Busy Tone	425+450	Continuous.
Ring Tone	425+450	(2.0/1.0) on/off.
Call Waiting Tone	425+450	(0.1/30.0) on/off.
Number Unobtainable Tone	425+450	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+02:00
Emergency Numbers	122, 122, 180

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.16 Finland

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.25/0.25) on/off.
Fast Busy Tone	425	(0.25/0.25) on/off.
Intercept Busy Tone	425	(0.25/0.25) on/off.
Ring Tone	425	(1.0/5.0) on/off.
Call Waiting Tone	425	(0.08/120.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	DTMFA
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Finnish (Suomi)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+02:00 - Helsinki

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Finnish
Phone Display	
- DS Phones	✓
- T3 Phones	✗
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Finnish
Manager	✗
System Status Application	✗
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✗
IP Office Customer Call Reporter	✗

13.17 France

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	330	Continuous.
Alternate Dial Tone	330	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	440	(0.5/0.5) on/off.
Fast Busy Tone	440	(0.5/0.5) on/off.
Intercept Busy Tone	440	(0.5/0.5) on/off.
Ring Tone	440	(1.5/3.5) on/off.
Call Waiting Tone	440	(0.1/8.0) on/off.
Number Unobtainable Tone	440	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	French
Feature Phone Clearing	Disconnect Tone
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+01:00 - Paris

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	French
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	French/ Canadian French
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	French
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.18 Greece

The tones and cadences below are applicable for the IP Office 3.2 4Q7+ and 4.1 1Q8+ maintenance releases and for IP Office 4.2+. Releases prior to those used the United Kingdom 894 set of tones and cadences.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous
Alternate Dial Tone	425	(0.5/1.0) on/off
Secondary Dial Tone	425	(0.2/0.3/0.7/0.8) on/off
Busy Tone	425	(0.3/0.3) on/off
Fast Busy Tone	425	(0.15/0.15) on/off
Intercept Busy Tone	425	(0.1/0.1/0.2/0.2) on/off
Ring Tone	425	(1.0/4.0) on/off
Call Waiting Tone	425	(0.3/8.0) on/off
Number Unobtainable Tone	425	(0.1/0.2) on/off

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+02:00 - Athens.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Greek
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✓
- Embedded Voicemail	✓	✗

Language	English	Greek
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	✗

13.19 Germany

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	(0.16/0.16/0.16/0.16/0.16/0.8) on/off.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	425	Continuous
Busy Tone	425	(0.48/0.48) on/off.
Fast Busy Tone	425	(0.48/0.48) on/off.
Intercept Busy Tone	425	(0.48/0.48) on/off.
Ring Tone	425	(0.945/4.05) on/off.
Call Waiting Tone	425	(0.08/0.2/0.08/10) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	Busy Tone.
Display Language	German.
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+01:00 - Berlin.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	French
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	French
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.20 Hong Kong

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(0.25/0.25) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	480/620	(0.25/0.25) on/off.
Ring Tone	440/620	(2.0/4.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	–
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	7 seconds.
Default Currency	CNY
Default Time Zone	UTC+08:00 - Hong Kong.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Simplified Chinese	Madarin	Cantonese
Phone Display			
- DS Phones	✗	–	–
- T3 Phones	✗	–	–
Voicemail Prompts			
- Voicemail Pro	–	✓	✓
- Embedded Voicemail	–	✓	✓

Language	Simplified Chinese	Madarin
Manager	✓	–
System Status Application	✓	–
Phone Manager	✓	–
Soft Console	✓	–
one-X Portal for IP Office	✓	–
IP Office Customer Call Reporter	✓	–

13.21 Hungary

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	(0.5/0.5) on/off.
Ring Tone	425	(1.0/4.0) on/off.
Call Waiting Tone	425	(0.15/0.15/0.15/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone
Voicemail Truncation Time	7 seconds.
Default Currency	HUF
Default Time Zone	UTC+01:00 - Belgrade.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Hungarian
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✓
- Embedded Voicemail	✓	✗

Language	English	Hungarian
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	✗

13.22 Iceland

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+450	Continuous.
Alternate Dial Tone	350+450	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.375/0.375) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	UK20
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	ISK
Default Time Zone	UTC+00:00 - Reykjavik

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.23 India

Support for this locale was added as part of IP Office 4.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	400	Continuous.
Alternate Dial Tone	400	(0.25/0.25/0.25/3.25) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.25/0.25) on/off.
Intercept Busy Tone	400	(0.75/0.75) on/off.
Ring Tone	400	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.2/0.1/0.2/7.5) on/off.
Number Unobtainable Tone	400	(2.8/0.2) on/off.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	0.300s
Default Caller Display Type	DTMFA
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	7 seconds.
Default Currency	INR
Default Time Zone	UTC+05:30 - Chennai, Mumbai, New Dehli.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.24 Italy

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+425	Continuous.
Alternate Dial Tone	350+425	(1.0/0.5) on/off.
Secondary Dial Tone	425	(0.2/0.2/0.6/1.0) on/off.
Busy Tone	400	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	(0.5/0.5) on/off.
Ring Tone	425	(1.0/4.0) on/off.
Call Waiting Tone	400	(0.1/4.9) on/off.
Number Unobtainable Tone	400	(0.1/0.1) on/off.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	10 seconds with no digit dialed.
Disconnect Tone	NU
Display Language	Italian
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	2 seconds.
Default Currency	EUR
Default Time Zone	UTC+01:00 - Rome.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Italian
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Italian
- Manager	✓
- System Status Application	✓
- Phone Manager	✓
- Soft Console	✓
- one-X Portal for IP Office	✓
- IP Office Customer Call Reporter	✓

13.25 Korea

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	(1.0/0.25) on/off.
Alternate Dial Tone	350+440	(0.25/0.25) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.3/0.2) on/off.
Intercept Busy Tone	480+620	(0.125/0.025/0.125/1.5) on/off.
Ring Tone	440+480	(1.0/2.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone
Voicemail Truncation Time	3 seconds.
Default Currency	KRW
Default Time Zone	UTC+09:00 - Seoul.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Korean
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✓
- Embedded Voicemail	✓	✓

Language	English	Korean
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✓
Soft Console	✓	✓
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	✗

13.26 Kuwait

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.25/0.25) on/off.
Intercept Busy Tone	425	Continuous.
Ring Tone	425	(1.0/4.0) on/off.
Call Waiting Tone	425	(0.1/4.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+03:00
Emergency Numbers	112

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.27 Mexico

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(2.0/4.0) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25) alternating tones.
Ring Tone	440+480	(1.0/3.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Mexican Spanish
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	2 seconds.
Default Currency	MXN
Default Time Zone	UTC-06:00 - Guadalajara, Mexico City.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Latin Spanish
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Latin Spanish
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.28 Morocco

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	Continuous.
Ring Tone	425	(1.7/3.3) on/off.
Call Waiting Tone	425	(0.1/30.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	French
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+00:00
Emergency Numbers	19, 177, 15

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.29 Netherlands

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.48/0.48) on/off.
Fast Busy Tone	425	(0.48/0.48) on/off.
Intercept Busy Tone	425	(0.48/0.48) on/off.
Ring Tone	425	(0.945/4.05) on/off.
Call Waiting Tone	400	(0.08/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	DTMFD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	Busy.
Display Language	Dutch.
Feature Phone Clearing	Go Idle.
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+01:00 - Amsterdam.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Dutch
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Dutch
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.30 New Zealand

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+350	Continuous.
Alternate Dial Tone	350+350	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.375/0.375) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	400	(0.075/0.1/0.075/0.1/0.075/0.1/0.075/0.4) on/off.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	Busy Tone.
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	5 seconds.
Default Currency	NZD
Default Time Zone	UTC+12:00 - Auckland, Wellington.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.31 Norway

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	(0.5/0.5) on/off.
Ring Tone	425	1.0/1.5/(1.0/4.0) on/off.
Call Waiting Tone	425	(0.08/0.6/0.08/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	UK
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Norwegian
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	NOK
Default Time Zone	UTC+01:00 - Oslo.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Norwegian
Phone Display	
- DS Phones	✓
- T3 Phones	✗
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Norwegian
Manager	✗
System Status Application	✗
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✗
IP Office Customer Call Reporter	✗

13.32 Oman

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	425	(0.375/0.375) on/off.
Fast Busy Tone	425	(0.4/0.35/0.225/0.525) on/off.
Intercept Busy Tone	425	Continuous.
Ring Tone	425	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	425	(0.3/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+04:00
Emergency Numbers	999

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.33 Pakistan

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	400	(0.5/0.5) on/off.
Fast Busy Tone	400	(0.25/0.25) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400	(1.0/2.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+05:00
Emergency Numbers	15, 1122, 115, 16.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.34 Peru

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(2.0/4.0) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25) alternating tones.
Ring Tone	440+480	(1.0/3.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Mexican Spanish
Feature Phone Clearing	Disconnect Tone
Voicemail Truncation Time	7 seconds.
Default Currency	PEN
Default Time Zone	GMT-05:00 - Lima

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Latin Spanish
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Latin Spanish
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.35 Poland

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	(0.5/0.5) on/off.
Ring Tone	425	(1.0/4.0) on/off.
Call Waiting Tone	425	(0.15/0.15/0.15/10.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect tone.
Voicemail Truncation Time	7 seconds.
Default Currency	PLN
Default Time Zone	UTC+01:00 - Warsaw.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Polish
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✓
- Embedded Voicemail	✓	✗

Language	English	Polish
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	✗

13.36 Portugal

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+450	Continuous
Alternate Dial Tone	350+450	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.375/0.375) on/off.
Intercept Busy Tone	400	Continuous
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	400	Continuous

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	UK20
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Portuguese
Feature Phone Clearing	Disconnect Tone
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+00:00 - Lisbon.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Brazilian	Portuguese
Phone Display		
- DS Phones	✓	✓
- T3 Phones	✗	✗
Voicemail Prompts		
- Voicemail Pro	✓	✓
- Embedded Voicemail	✓	✓

Language	Brazilian	Portuguese
Manager	✓	✗
System Status Application	✓	✓
Phone Manager	✓	✓
Soft Console	✓	✗
one-X Portal for IP Office	✓	✓
IP Office Customer Call Reporter	✓	✓

13.37 Qatar

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	400	(0.8/0.8) on/off.
Fast Busy Tone	400	(0.4/0.35/0.22/0.52) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.38/0.25/0.38/2.0) on/off.
Call Waiting Tone	400	(0.2/0.6/0.2/5.0) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+03:00
Emergency Numbers	999

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.38 Russia

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	Continuous.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.35/0.35) on/off.
Fast Busy Tone	425	(0.2/0.2) on/off.
Intercept Busy Tone	425	(0.35/0.35) on/off.
Ring Tone	425	(1.0/4.0) on/off.
Call Waiting Tone	425	(0.2/5.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	None
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Russian
Feature Phone Clearing	Disconnect Tone
Voicemail Truncation Time	7 seconds.
Default Currency	RUR
Default Time Zone	UTC+03:00 - Moscow.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Russian
Phone Display	
- DS Phones	✓
- T3 Phones	✗
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Russian
Manager	✗
System Status Application	✗
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

Cyrillic display support for the Russian locale is limited on phones as follows:

- 4600 Series/5600 Series: All display strings if using the double-byte build of phone firmware.
- EU24/EU24BL: No Cyrillic language support.

13.39 Saudi Arabia

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(.25/.25) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(.5/.5) on/off.
Fast Busy Tone	480+620	(.25/.25) on/off.
Intercept Busy Tone	480+620	(.25/.25/.25/.25) on/off.
Ring Tone	440+480	(2.0/4.0) on/off.
Call Waiting Tone	480+620	(.1/.2/1200.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	5 seconds.
Default Currency	SAR
Default Time Zone	UTC+03:00 - Kuwait, Riyadh.
Emergency Numbers	997, 998, 999

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.40 Singapore

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	425+50	(1.0/4.0) on/off.
Fast Busy Tone	450	(0.5/0.5) on/off.
Intercept Busy Tone		
Ring Tone	425+50	(2.0/1.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+03:00
Emergency Numbers	122, 122, 180

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.41 South Africa

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+450	Continuous.
Alternate Dial Tone	350+450	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.375/0.375) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	UK20
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	ZAR
Default Time Zone	UTC+02:00 - Pretoria.
Emergency Numbers	10111, 10177

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.42 Spain

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.25/0.25) on/off.
Fast Busy Tone	425	(0.25/0.25) on/off.
Intercept Busy Tone	425	(0.25/0.25) on/off.
Ring Tone	425	(1.5/3.0) on/off.
Call Waiting Tone	425	(0.15/0.15/0.15/30.0) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	A-Law.
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Spanish.
Feature Phone Clearing	Disconnect tone.
Voicemail Truncation Time	7 seconds.
Default Currency	EUR
Default Time Zone	UTC+01:00 - Madrid.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Latin Spanish	Spanish
Phone Display		
- DS Phones	✓	✓
- T3 Phones	✓	✓
Voicemail Prompts		
- Voicemail Pro	✓	✓
- Embedded Voicemail	✓	✓

Language	Latin Spanish	Spanish
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✓
Soft Console	✓	✓
one-X Portal for IP Office	✓	✓
IP Office Customer Call Reporter	✓	✓

13.43 Sweden

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.25/0.25) on/off.
Fast Busy Tone	425	(0.25/0.25) on/off.
Intercept Busy Tone	425	(0.25/0.25) on/off.
Ring Tone	425	(1.0/5.0) on/off.
Call Waiting Tone	425	(0.08/120) on/off.
Number Unobtainable Tone	425	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	DTMFA
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Swedish
Feature Phone Clearing	Disconnect Tone
Voicemail Truncation Time	7 seconds.
Default Currency	SEK
Default Time Zone	UTC+01:00 - Stockholm

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Swedish
Phone Display	
- DS Phones	✓
- T3 Phones	✗
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Swedish
Manager	✗
System Status Application	✗
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✗
IP Office Customer Call Reporter	✗

13.44 Switzerland

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	425	Continuous.
Alternate Dial Tone	425	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	425	(0.5/0.5) on/off.
Fast Busy Tone	425	(0.5/0.5) on/off.
Intercept Busy Tone	425	(0.2/0.2) on/off.
Ring Tone	425	(1.0/4.0) on/off.
Call Waiting Tone	425	(0.2/0.2/0.2/4.0) on/off.
Number Unobtainable Tone	425	(0.2/0.2) on/off.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	UK20
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	Busy
Display Language	French, Italian or German depending on specific selected locale.
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	CHF
Default Time Zone	UTC+01:00 - Bern.

13.45 Taiwan

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(0.1/0.1) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	480/620	(0.25/0.25)/(0.25/0.25) on/off.
Ring Tone	440+480	(1.2/2.0) on/off.
Call Waiting Tone	350+440	(0.25/0.25/0.25/5.25) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.150s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	DTMFD/FSKD (auto detected)
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	TWD
Default Time Zone	UTC+08:00 - Taipei.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Simplified Chinese	Madarin	Cantonese
Phone Display			
- DS Phones	✗	–	–
- T3 Phones	✗	–	–
Voicemail Prompts			
- Voicemail Pro	–	✓	✓
- Embedded Voicemail	–	✓	✓

Language	Simplified Chinese	Madarin
Manager	✓	–
System Status Application	✓	–
Phone Manager	✓	–
Soft Console	✓	–
one-X Portal for IP Office	✓	–
IP Office Customer Call Reporter	✓	–

13.46 Turkey

Support for this locale was added as part of IP Office 4.2.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(0.25/0.25) on/off.
Secondary Dial Tone	350+440	Continuous.
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	480+620	Continuous.
Ring Tone	440+480	(2.0/4.0) on/off.
Call Waiting Tone	480+620	(0.0/0.1/0.2/1200) on/off.
Number Unobtainable Tone	480+620	(0.1/0.2) on/off.

Analog Phone Settings	
Ring Current Frequency	25Hz.
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle.
Voicemail Truncation Time	0 seconds.
Default Currency	YTL
Default Time Zone	UTC+02:00 - Istanbul.
Emergency Numbers	155, 112, 110

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.47 United Arab Emirates

Support added in IP Office 6.1.

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	440	Continuous.
Alternate Dial Tone	440	(1.0/0.5) on/off.
Secondary Dial Tone	440+350	Continuous.
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.4/0.35/0.225/0.525) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30.0) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	81V

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	-
Default Currency	-
Default Time Zone	UTC+04:00 -Abu Dhabi.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English	Arabic
Phone Display		
- DS Phones	✓	✗
- T3 Phones	✓	✗
Voicemail Prompts	(UK/US English)	
- Voicemail Pro	✓	✗
- Embedded Voicemail	✓	✓

Language	English	Arabic
Manager	✓	✗
System Status Application	✓	✗
Phone Manager	✓	✗
Soft Console	✓	✗
one-X Portal for IP Office	✓	✗
IP Office Customer Call Reporter	✓	

13.48 United Kingdom

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+450	Continuous.
Alternate Dial Tone	350+450	(1.0/0.5) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	400	(0.375/0.375) on/off.
Fast Busy Tone	400	(0.375/0.375) on/off.
Intercept Busy Tone	400	Continuous.
Ring Tone	400+450	(0.4/0.2/0.4/2.0) on/off.
Call Waiting Tone	400	(0.1/30) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Ring Current Cadence	(0.4/0.2/0.4/2.0s) on/off.
Minimum Flash Hook Time	0.025s
Maximum Flash Hook Time	0.350s
Default Caller Display Type	UK
Default Message Waiting Indication Type	On = *101V* on Phone V2 modules and IP500 Phone cards, otherwise *81V*.

Locale Specific Behavior	
Companding	A-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	English (UK)
Feature Phone Clearing	Go Idle
Voicemail Truncation Time	7 seconds.
Default Currency	GBP
Default Time Zone	UTC+00:00 - London.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.49 United States

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone	350+440	Continuous.
Alternate Dial Tone	350+440	(0.25/0.25) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25) alternating tone.
Ring Tone	440+480	(2.0/4.0) on/off.
Call Waiting Tone	480+620	(0.0/0.1/0.2/1200) on/off.
Number Unobtainable Tone	400	Continuous.

Analog Phone Settings	
Ring Current Frequency	20Hz
Minimum Flash Hook Time	0.300s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped

Locale Specific Behavior	
Companding	Mu-Law.
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	Silence.
Display Language	English (US).
Feature Phone Clearing	Go Idle.
Voicemail Truncation Time	0 seconds.
Default Currency	USD
Default Time Zone	UTC-05:00 - Eastern Standard Time (EST)

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	English
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	(UK/US English)
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	English
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

13.50 Venezuela

Tone	Frequency (Hz)	Cadence (seconds)
Dial Tone Tone	425	Continuous.
Alternate Dial Tone	425	(2.0/4.0) on/off.
Secondary Dial Tone	Use Dial Tone.	
Busy Tone	480+620	(0.5/0.5) on/off.
Fast Busy Tone	480+620	(0.25/0.25) on/off.
Intercept Busy Tone	440/620	(0.25/0.25) alternating tone.
Ring Tone	440+480	(1.0/3.0) on/off.
Call Waiting Tone	480+620	(0.06/0.25/0.06/5.0) on/off.
Number Unobtainable Tone	480+620	Continuous.

Analog Phone Settings	
Ring Current Frequency	25Hz
Minimum Flash Hook Time	0.050s
Maximum Flash Hook Time	1.000s
Default Caller Display Type	FSKD
Default Message Waiting Indication Type	51V Stepped.

Locale Specific Behavior	
Companding	Mu-Law
Dialling Timeout	30 seconds from start of dialing.
Disconnect Tone	NU
Display Language	Mexican Spanish.
Feature Phone Clearing	Disconnect Tone.
Voicemail Truncation Time	7 seconds.
Default Currency	VEB
Default Time Zone	UTC-04:30 - Caracas.

These are the language support options for the IP Office 6.1 suite of software. The language support options will differ for other system releases.

Language	Latin Spanish
Phone Display	
- DS Phones	✓
- T3 Phones	✓
Voicemail Prompts	
- Voicemail Pro	✓
- Embedded Voicemail	✓

Language	Latin Spanish
Manager	✓
System Status Application	✓
Phone Manager	✓
Soft Console	✓
one-X Portal for IP Office	✓
IP Office Customer Call Reporter	✓

Chapter 14.
Appendix: SMDR

14. Appendix: SMDR

IP Office 4.2+: The control unit is able to send SMDR (Station Message Detail Reporting) records to a specified IP address and port.

Typically an SMDR record is output for each call between two parties (internal and or external) that is handled by the system. In some scenarios, for examples transfers, where a call involves multiple parties then multiple SMDR records may be output for each part of the call. See SMDR Examples 903.

Each SMDR record contains call information in a comma-separated format (CSV) format, that is variable-width fields with each field separated by commas. See SMDR Fields 900.

Unless otherwise stated, the SMDR referred to in this documentation is the SMDR output direct from the system rather than that output via the Delta Server application.

- **SMDR Buffer Persistence**
  For IP Office 6.0, the operation of SMDR on IP500 and IP500v2 control units has been enhanced to store any buffered SMDR records during any controlled system power downs or reboots.

Enabling SMDR

1. Receive the configuration from the system.
2. Select **System** and then select the **CDR/SMDR** tab.
3. Use the **Output** drop down box to select ***SMDR only***.
4. In the SMDR settings, enter the required **IP Address** and **TCP Port**.

SMDR Records

An SMDR record is generated for each call between two devices on the system. Devices include extensions, trunk lines (or channels on a trunk), voicemail channels, conference channels and system tones.

Calls which are not presented to another device do not generate an SMDR record. For example internal users dialing short code that simply changes a configuration setting.

The SMDR record is generated when the call ends, therefore the order of the SMDR records output does not match the call start times.

Each record contains a call ID which is increased by 1 for each subsequent call.

When a call moves from one device to another, an SMDR record is output for the first part of the call and an additional SMDR record will be generated for the subsequent part of the call.

Each of these records will have the same Call ID.

Each record for a call indicates in the Continuation field if there will be further records for the same call.

Call Times

Each SMDR record can include values for ringing time, connected time, held time and parked time. The total duration of an SMDR record is the sum of those values.

The time when a call is not in any one of the states above, for example when one party to the call has disconnected, is not measured and included in SMDR records.

Where announcements are being used, the connected time for a call begins either when the call is answered or the first announcement begins.

All times are rounded up to the nearest second.

Each SMDR record has a Call Start time taken from the system clock time. For calls being transferred or subject to call splitting, each of the multiple SMDR records will have the same Call Start time as the original call.

Delta Server SMDR Differences

Previously SMDR records were obtained using the Delta Server application running on a PC connected to the system via the LAN. This method was used for pre-IP Office 4.2 systems. Delta Server SMDR output is still supported with IP Office 4.2+ systems. However, it is not supported if also outputting direct SMDR records from the system.

There are a number of differences between the system SMDR output and that output by the Delta Server application. The key differences are:

- **Advice of Charge and Authorization Code Fields are Included as Standard**
  These fields are optional in the Delta Server SMDR output. They are included as standard in the SMDR output.
- **Additional Fields are Included for External Outgoing Call Routing**
  The SMDR output can include additional external targeting fields for calls routed out via external lines. These fields indicate what device targeted the call externally, how, and to what number.
- **Call ID Starting Number**
  The Call ID is always reset whenever the system is restarted and then incremented by 1 for each subsequent call that is not a continuation of a previous call.
  - The Call ID used by the Delta Server applications starts from 1.
  - The Call ID used for system SMDR starts from 1,000,000.
- **Small Community Network**
  The following differences apply for calls to and from SCN destinations.
  - System SMDR indicates calls across an SCN as internal.
  - System SMDR inserts the hunt group or user name of the remote SCN party rather than the local line used.
- **Parked and Held Time**
  For parked calls, the Delta Server included the time a call was parked in both the **Park Time** and **Hold Time** fields. The system SMDR uses just the **Park Time** field.
- **Additional Records Through Call Splitting**
  The system supports Call Splitting for Diverts 179 which produces separate initial call and forwarded call records. This applies for calls forwarded by forward unconditional, forward on no answer, forward on busy, DND or mobile twinning. It also applies to calls forwarded off-switch by an incoming call route. The two sets of records will have the same Call ID. The call time fields of the forward call record are reset from the moment of forwarding on the external trunk.

14.1 SMDR Fields

The SMDR output contains the following fields. Note that time values are rounded up to the nearest second.

1. Call Start
   Call start time in the format YYYY/MM/DD HH:MM:SS. For all transferred call segment this is the time the call was initiated, so each segment of the call has the same call start time.

2. Connected Time
   Duration of the connected part of the call in HH:MM:SS format. This does not include ringing, held and parked time. A lost or failed call will have a duration of 00:00:00. The total duration of a record is calculated as *Connected Time + Ring Time + Hold Time + Park Time.*

3. Ring Time
   Duration of the ring part of the call in seconds.

   - For inbound calls this represents the interval between the call arriving at the switch and it being answered, not the time it rang at an individual extension.
   - For outbound calls, this indicates the interval between the call being initiated and being answered at the remote end if supported by the trunk type. Analog trunks are not able to detect remote answer and therefore cannot provide a ring duration for outbound calls.

4. Caller
   The callers' number. If the call was originated at an extension, this will be that extension number. If the call originated externally, this will be the CLI of the caller if available, otherwise blank.

5. Direction
   Direction of the call – *I* for Inbound, *O* for outbound. Internal calls are represented as *O* for outbound. This field can be used in conjunction with Is_Internal below to determine if the call is internal, external outbound or external inbound.

6. Called Number
   This is the number called by the system. For a call that is transferred this field shows the original called number, not the number of the party who transferred the call.

   - Internal calls: The extension, group or short code called.
   - Inbound calls: The target extension number for the call.
   - Outbound calls: The dialed digits.
   - Voice Mail: Calls to a user's own voicemail mailbox.

7. Dialled Number
   For internal calls and outbound calls, this is identical to the Called Number above. For inbound calls, this is the DDI of the incoming caller.

8. Account
   The last account code attached to the call. Note: System account codes may contain alphanumeric characters.

9. Is Internal
   0 or *1*, denoting whether both parties on the call are internal or external (*1* being an internal call). Calls to SCN destinations are indicated as internal.

Direction	Is Internal	Call Type
I	0	Incoming external call.
O	1	Internal call.
O	0	Outgoing external call.

10. Call ID
    This is a number starting from 1,000,000 and incremented by 1 for each unique call. If the call has generates several SMDR records, each record will have the same Call ID. Note that the Call ID used is restarted from 1,000,000 if the system is restarted.

11. Continuation
    1 if there is a further record for this call id, *0* otherwise.

12. Party1Device
    The device 1 number. This is usually the call initiator though in some scenarios such as conferences this may vary. If an extension/hunt group is involved in the call its details will have priority over a trunk. That includes remote SCN destinations.

Type	Party Device	Party Name
Internal Number	E*<extension number>*	<name>
Voicemail	V*<9500 + channel number>*	VM Channel *<channel number>*
Conference	V*<1><conference number>+<channel number>*	CO Channel *<conference number.channel number>*

Type	Party Device	Party Name
Line	T<*9000+line number*>	Line <*line number*>.<*channel if applicable*>
Other	V<*8000+device number*>	U<*device class*> <*device number*>.<*device channel*>
Unknown/Tone	V8000	U1 0.0

13. Party1Name
The name of the device – for an extension or agent, this is the user name.

14. Party2Device
The other party for the SMDR record of this call segment. See Party1Device above.

15. Party2Name
The other party for the SMDR record of this call segment. See Party1Name above.

16. Hold Time
The amount of time in seconds the call has been held during this call segment.

17. Park Time
The amount of time in seconds the call has been parked during this call segment.

18. AuthValid
This field is used for authorization codes. This field shows *1* for valid authorization or *0* for invalid authorization.

19. AuthCode
This field shows either the authorization code used or *n/a* if no authorization code was used.

20. User Charged
This and the following fields are used for ISDN Advice of Charge (AoC) 740. The user to which the call charge has been assigned. This is not necessarily the user involved in the call.

21. Call Charge
The total call charge calculated using the line cost per unit and user markup.

22. Currency
The currency. This is a system wide setting set in the system configuration.

23. Amount at Last User Change
The current AoC amount at user change.

24. Call Units
The total call units.

25. Units at Last User Change
The current AoC units at user change.

26. Cost per Unit
This value is set in the system configuration against each line on which Advice of Charge signalling is set. The values are 1/10,000th of a currency unit. For example if the call cost per unit is £1.07, a value of 10700 should be set on the line.

27. Mark Up
Indicates the mark up value set in the system configuration for the user to which the call is being charged. The field is in units of 1/100th, for example an entry of 100 is a markup factor of 1.

The following additional fields are provided by system SMDR. They are not provided by Delta Server SMDR.

28. External Targeting Cause
This field indicates who or what caused the external call and a reason code. For example *U FU* indicates that the external call was caused by the Forward Unconditional setting of a User.

Targeted by		Reason Code	
HG	Hunt Group.	fb	Forward on Busy.
U	User.	fu	Forward unconditional.
LINE	Line.	fnr	Forward on No Response.
AA	Auto Attendant.	fdnd	Forward on DND.
ICR	Incoming Call Route.	CfP	Conference proposal (consultation) call.
RAS	Remote Access Service.	Cfd	Conferenced.
?	Other.	MT	Mobile Twinning.
		TW	Teleworker.
		XfP	Transfer proposal (consultation) call.
		Xfd	Transferred call.

29. External Targeter Id
The associated name of the targeter indicated in the External Targeting Cause field. For hunt groups and users this will be their name in the system configuration. For an Incoming Call Route this will be the Tag if set, otherwise *ICR*.

30. External Targeted Number
This field is used for forwarded, Incoming Call Route targeted and mobile twin calls to an external line. It shows the external number called by the system as a result of the off switch targeting where as other called fields give the original number dialled.

14.2 SMDR Examples

The following are examples of system SMDR records for common call scenarios.

Basic Examples

Lost incoming Call

In this record, the Call duration is zero and the Continuation field is 0, indicating that the call was never connected. The Ring Time shows that it rang for 9 seconds before ending.

```
2008/06/28 09:28:41,00:00:00,9,8004206,I,4324,4324,,0,1000014155,0,E4324,Joe Bloggs,T9161,LINE 5.1,0,0,,,,,,,,,,,,
```

Call Answered by Voicemail

In this example, 215 has made a call to 211. However the Party2Device and Party2Name show that the call was answered by voicemail.

```
2008/10/20 06:43:58,00:00:10,21,215,O,211,211,,I,28,0,E215,Extn215,V9051,VM Channel 1,0,0,,,,,,,,,,,,
```

Call Transferred to Voicemail

In this example, the Continuation field in the first record tells us that it wasn't the end of the call. The matching Call ID identifies the second record as part of the same call. The change in Party 1 details between the two records show that the call was transferred to voicemail.

```
2008/06/28 09:30:57,00:00:13,7,01707392200,I,299999,299999,,0,1000014160,1,E4750,John Smith,T9002,LINE 1.2,11,0,,,,,,,,,,,,
2008/06/28 09:30:57,00:00:21,0,01707392200,I,299999,299999,,0,1000014160,0,V9502,VM Channel 2,T9002,LINE 1.2,0,0,,,,,,,,,,,,
```

External Call

The Is Internal field being 0 shows this to be a external call. The Direction field as I shows that it was an incoming call. The Ring Time was 7 seconds and the total Connected Time was 5 seconds.

```
2008/08/01 15:14:19,00:00:05,7,01707299900,I,403,390664,,0,1000013,0,E403,Extn403,T9001,Line 1.2,0,0,,,,,,,,,,,,
```

Internal call

The Is Internal field being 1 shows this to be a internal call. The Ring Time was 4 seconds and the total Connected Time was 44 seconds.

```
2008/06/26 10:27:44,00:00:44,4,4688,O,4207,4207,,1,1000013898,0,E4688,Joe Bloggs,E4207,John Smith,0,0,,,,,,,,,,,,
```

Outgoing Call

The combination of the Direction field being outbound and the Is Internal field be 0 show that this was a outgoing external call. The line (and in this case channel) used are indicated by the Party2 Name and being a digital channel the Ring Time before the call was answered is also shown.

```
2008/06/28 08:55:02,00:08:51,9,4797,O,08000123456,08000123456,,0,1000014129,0,E4797,Joe Bloggs,T9001,LINE 1.1,0,0,,,,,,,,,,,,
```

Voicemail Call

The two records below show calls to voicemail. The first shows the Dialed Number as*17, the default short code for voicemail access. The second shows the Dialed Number as VoiceMail, indicating some other method such as the Message key on a phone was used to initiate the call.

```
2008/06/28 09:06:03,00:00:19,0,4966,O,*17,*17[1],,1,1000014131,0,E4966,John Smith,V9501,VM Channel 1,0,0,,,,,,,,,,,,
2008/06/28 09:06:03,00:00:19,0,4966,O,VoiceMail,VoiceMail,,1,1000014134,0,E4966,John Smith,V9501,VM Channel 1,0,0,,,,,,,,,,,,
```

Parked Call

In this example the first record has a Park Time showing that the call was parked. The Continuation field indicates that the call did not end this way and there are further records. The second record has the same Call ID and shows a change in the Party2Name [4], indicating that party unparked the call. Note also that both records share the same call start time.

```
2008/10/20 07:18:31,00:00:12,3,215,O,210,210,,1,38,1,E215,Extn215,E210,Extn210,0,7,,,,,,,,,,,,
2008/10/20 07:18:31,00:00:10,0,215,O,210,210,,1,38,0,E215,Extn215,E211,Extn211,0,0,,,,,,,,,,,,
```

Incoming call with Account Code

In this example, at some stage as the call was made or during the call, an Account Code has been entered. In this specific case it is a text account code which can be selected and entered by the user using Phone Manager.

```
2008/06/28 11:29:12,00:00:02,2,5002,I,1924,1924,Support,0,1000014169,0,E1924,Extn1924,T9620,LINE 8.20,0,0,,,,,,,,,,,,
```

Conference Using Conference Add Short Code

In this example 2101 has made a call and put put it on hold (record 2), then made another call and put it on hold (record 1) and then dialled the default short code *47 to conference all their held calls (record 3). The records for the first two calls have the Continuation field set as 1 indicating that the calls continued in further records.

Record 3 shows 2101 making a new call in which they dial *47, which places them and their held calls into a conference. This is shown by the Party Device and Party Name details as being a conference (100) and the conference channel used for each.

For both the Continuation fields show that the calls do not end but rather have subsequent records.

```
2008/07/09 17:55,00:00:03,3,2101,O,8262623#,8262623#,,0,1000024,1,E2101,Extn2101,T9002,Line 2.1,8,0,,,,,,,,,,,,
2008/07/09 17:54,00:00:29,7,2101,O,2121,2121,,1,1000023,1,E2101,Extn2101,E2121,Extn2121,23,0,,,,,,,,,,,,
2008/07/09 17:55,00:00:46,0,2101,O,*47,*47,,1,1000026,0,E2101,Extn2101,V11001,CO Channel 100.1,0,0,,,,,,,,,,,,
2008/07/09 17:54,00:00:49,0,,O,71234567890,71234567890,,1,1000023,0,E2121,Extn2121,V11003,CO Channel 100.3,0,0,,,,,,,,,,,,
2008/07/09 17:55,00:00:49,0,,O,8262623#,8262623#,,0,1000024,0,V11002,CO Channel 100.2,T9002,Line 2.1,0,0,,,,,,,,,,,,
```

Conference Using Conference Button

In this example, an extension user answers a call and then brings in another user by using the Conference button on their phone. Again we see records for the initial call, the conference proposal call and then for the 3 parties in the conference that is created.

```
2008/07/09 15:05:41,00:00:04,3,203,O,201,201,,1,1000009,1,E203,Extn203,E201,Extn201,0,0,,,,,,,,,,,,
2008/07/09 15:05:26,00:00:09,3,207,O,203,203,,1,1000008,1,E207,Extn207,E203,Extn203,10,0,,,,,,,,,,,,
2008/07/09 15:05:41,00:00:08,0,,O,,,,1,1000009,0,E201,Extn201,V11001,CO Channel 100.1,0,0,,,,,,,,,,,,
2008/07/09 15:05:50,00:00:10,0,203,O,201,201,,1,1000010,0,E203,Extn203,V11002,CO Channel 100.2,0,0,,,,,,,,,,,,
2008/07/09 15:05:26,00:00:10,0,207,O,203,203,,1,1000008,0,E207,Extn207,V11003,CO Channel 100.3,0,0,,,,,,,,,,,,
```

Adding a Party to a Conference

This example is a variant on that above. Having started a conference, extension 203 adds another party.

```
2008/07/09 15:08:31,00:00:03,3,203,O,201,201,,1,1000014,1,E203,Extn203,E201,Extn201,0,0,,,,,,,,,,,,
2008/07/09 15:08:02,00:00:22,6,207,O,203,203,,1,1000013,1,E207,Extn207,E203,Extn203,9,0,,,,,,,,,,,,
2008/07/09 15:08:45,00:00:02,4,203,O,403,403,,0,1000016,1,E203,Extn203,E403,Libby Franks,0,0,,,,,,,,,,,,
2008/07/09 15:08:02,00:00:24,0,207,O,203,203,,1,1000013,0,E207,Extn207,V11003,CO Channel 100.3,0,0,,,,,,,,,,,,
2008/07/09 15:08:39,00:00:17,0,203,O,201,201,,1,1000015,0,E203,Extn203,V11002,CO Channel 100.2,8,0,,,,,,,,,,,,
2008/07/09 15:08:31,00:00:26,0,,O,,,,1,1000014,0,E201,Extn201,V11001,CO Channel 100.1,0,0,,,,,,,,,,,,
2008/07/09 15:08:45,00:00:12,0,,O,403,403,,0,1000016,0,E403,Libby Franks,V11004,CO Channel 100.4,0,0,,,,,,,,,,,,
```

Transfer

In this example 2126 has called 2102. The record (1) for this has the Continuation set a 1 indicating that it has further records. In the following record (3) with the same Call ID it can be seen that the Party 2 Device and Party 2 Name fields have changed, indicating that the call is now connected to a different device, in this example 2121. We can infer the blind transfer from the intermediate record (2) which shows a call of zero Connected Time between the original call destination 2102 and the final destination 2121.

```
2008/07/09 17:51,00:00:38,18,2126,O,2102,2102,,1,1000019,1,E2126,Extn2126,E2102,Extn2102,19,0,,,,,,,,,,,,
2008/07/09 17:52,00:00:00,7,2102,O,2121,2121,,1,1000020,0,E2102,Extn2102,E2121,Extn2121,0,0,,,,,,,,,,,,
2008/07/09 17:51,00:00:39,16,2126,O,2102,2102,,1,1000019,0,E2126,Extn2126,E2121,Extn2121,0,0,,,,,,,,,,,,
```

In this second example extension 402 answers an external call and then transfers it to extension 403. Again the two legs of the external call have the same time/date stamp and same call ID.

```
2008/08/01 15:23:37,00:00:04,7,01707299900,I,4001,390664,,0,1000019,1,E402,Extn402,T9001,Line 1.1,6,0,,,,,,,,,,,,
2008/08/01 15:23:46,00:00:00,3,402,O,403,403,,1,1000020,0,E402,Extn402,E403,Extn403,0,0,,,,,,,,,,,,
2008/08/01 15:23:37,00:00:04,4,01707299900,I,4001,390664,,0,1000019,0,E403,Extn403,T9001,Line 1.1,0,0,,,,,,,,,,,,
```

Busy/Number Unavailable Tone

In this example 2122 calls 2123 who is set to DND without voicemail. This results in 2122 receiving busy tone.

The records shows a call with a Connected Time of 0. The Call Number field shows 2123 as the call target but the Party 2 Device and Party 2 Name fields show that the connection is to a virtual device.

```
2008/07/09 17:59,00:00:00,0,2122,O,2123,2123,,1,1000033,0,E2122,Extn2122,V8000,U1 0.0,0,0,,,,,,,,,,,,
```

Call Pickup

The first record shows a call from 2122 to 2124 with a Connected Time of zero but a Ring Time of 8. The Continuation field indicates that the call has further records.

The second record has the same Call ID but the Party 2 Device and Party 2 Name details show that the call has been answered by 2121.

```
2008/07/09 18:00,00:00:00,8,2122,O,2124,2124,,1,1000038,1,E2122,Extn2122,E2124,Extn2124,0,0,,,,,,,,,,,,
2008/07/09 18:00,00:00:38,1,2122,O,2124,2124,,1,1000038,0,E2122,Extn2122,E2121,Extn2121,0,0,,,,,,,,,,,,
```

Internal Twinning

The records for scenarios such as internal call forwarding or follow me indicate the rerouting in a single record by having Caller and Called Number details that differ from the final Party 1 and Party 2 details. Internal twinning differs is showing a call answered at the twin exactly the same as having been answered at the primary.

203 is internally twinned to 201. Call from 207 to 203 but answer at 201.

```
2008/07/09 16:25:26,00:00:03,7,207,O,203,203,,1,1000037,0,E207,Extn207,E203,Extn203,0,0,,,,,,,,,,,,
```

Park and Unpark

Parking and unparking of a call at the same extension is simply shown by the Park Time field of the eventual SMDR record. Similarly calls held and unheld at the same extension are shown by the Held Time field of the eventual SMDR record for the call. The records below however show a call parked at one extension and then unparked at another.

The records show a call from 207 to 203. 203 then parks the call shown by the Park Time. The call is unparked by 201, hence the first record is indicated as continued in its Continuation field. The matching Call ID indicates the subsequent record for the call.

```
2008/07/09 16:39:11,00:00:00,2,207,O,203,203,,1,1000052,1,E207,Extn207,E203,Extn203,0,4,,,,,,,,,,,,
2008/07/09 16:39:11,00:00:02,0,207,O,203,203,,1,1000052,0,E207,Extn207,E201,Extn201,0,0,,,,,,,,,,,,
```

Distributed Hunt Group Call

An incoming call to site A is targeted to a distributed hunt group member on site B. They transfer the call back to a hunt group member on site A.

```
2008/08/01 15:32:52,00:00:10,19,01707299900,I,4002,390664,,0,1000024,1,E209,Luther-209,T9001,Line 1.2,0,0,,,,,,,,,,,,
2008/08/01 15:33:19,00:00:00,2,209,I,403,403,,0,1000025,0,E209,Luther-209,E403,Extn403,0,0,,,,,,,,,,,,
2008/08/01 15:32:52,00:00:03,3,01707299900,I,4002,390664,,0,1000024,0,E403,Extn403,T9001,Line 1.2,0,0,,,,,,,,,,,,
```

Voicemail Supervised Transfer

A call is routed to a voicemail module that performs a supervised transfer.

```
2008/08/01 16:36:04,00:00:09,0,01707299900,I,xfer,390664,,0,1000061,1,T9001,Line 1.1,V9508,VM Channel 8,0,0,,,,,,,,,,,,
2008/08/01 16:36:07,00:00:03,4,,I,402,402,,0,1000062,0,E402,Extn402,V8000,U12 0.8,0,0,,,,,,,,,,,,
2008/08/01 16:36:04,00:00:09,0,01707299900,I,402,390664,,0,1000061,0,E402,Extn402,T9001,Line 1.1,0,0,,,,,,,,,,,,
```

Outgoing External Call

The External Targeting Cause indicates that the external call was caused by a user. The lack of specific reason implies that it was most likely dialed. The External Targeter ID is the user name in this example

```
… 16:23:06,00:00:04,5,203,O,9416,9416,,0,1000035,0,E203,Extn203,T9005,Line 5.1,0,0,,,Extn203,,,,,,,U,Extn203,,
```

Rerouted External Call

In this example an incoming external call has been rerouted back off switch, shown by the Party 1 fields and the Party 2 fields being external line details. The External Targeter Cause shows that rerouting of the incoming call was done by an incoming call route (ICR). The External Targeter ID in this case is the Tag set on the incoming call route. The External Targeted Number is the actual external number call.

```
… 08:14:27,00:00:03,5,392200,I,9416,200,,0,1000073,0,T9005,Line 5.1,T9005,Line 5.2,0,0,,,,0000.00,,0000.00,0,0,618,0.01,ICR,Main ICI
```

External Forward Unconditional

In this example, user 203 has a forward unconditional number set for calls. This is indicated by the External Targeting Cause showing user and forward unconditional. The External Targeter ID shows the source of the call being forwarded, in this example user 207. The External Targeted Number shows the actual external number called by the system.

```
… 16:22:41,00:00:02,5,207,O,203,203,,0,1000034,0,E207,Extn207,T9005,Line 5.1,0,0,,,Extn203,0000.00,,0000.00,0,0,618,1.00,U fu,Extn2
```

Transferred Manually

In this example the internal user transfers a call to an external number. The External Targeting Cause in the first record indicates that this external call is the result of a user (U) transfer proposal (XfP) call. The Continuation field indicates that another record with the same Call ID will be output.

The additional records are output after the transferred call is completed. The first relates to the initial call prior. The second is the transferred call with the External Targeting Cause now indicating user (U) transferred (Xfd).

```
… 16:33:19,00:00:05,3,203,O,9416,9416,,0,1000044,1,E203,Extn203,T9005,Line 5.1,0,0,,,,,,,,,,U XfP,Extn207,,
… 16:33:09,00:00:02,2,207,O,203,203,,1,1000043,0,E207,Extn207,E203,Extn203,11,0,,,,,,,,,,,,,,
… 16:33:19,00:00:04,0,207,O,9416,9416,,0,1000044,0,E207,Extn207,T9005,Line 5.1,0,0,,,Extn207,,,,,,,,U Xfd,Extn203,,
```

Mobile Twinned Call Answered Internally

For this example user 203 has mobile twining enabled to the external number 9416 as twin. Their mobile dial delay is set to 2 seconds. The call is answered at the user's internal extension.

In this scenario the record for the external call part of twinning is output immediately the call is answered internally. The Call Start for this record differs dues to the user's Mobile Dial Delay setting. The External Targeting Cause indicates the external call was the result of user (U) mobile twinning (MT) settings. If the call had been answered before the mobile dial delay expired, no external call and therefore no record would be produced. When the call is completed the second record is output.

```
… 16:17:59,00:00:00,7,,O,9416,9416,,0,1000028,0,E203,Extn203,T9005,Line 5.1,0,0,,,,,,,,,,U MT,Extn203,9416,
… 16:17:58,00:00:07,9,207,O,203,203,,1,1000027,0,E207,Extn207,E203,Extn203,0,0,,,,,,,,,,,,,
```

Mobile Twinned Call Answered at the Mobile Twin

This is the same scenario as the example above except that the call is answered at the external mobile twinning destination. Unlike the previous example the external call record has a non-zero Call Time showing that the call was also answered externally.

```
… 16:17:04,00:00:06,9,,O,9416,9416,,0,1000026,0,E203,Extn203,T9005,Line 5.1,0,0,,,,,,,,,,U MT,Extn203,9416
… 16:17:02,00:00:06,11,207,O,203,203,,1,1000025,0,E207,Extn207,E203,Extn203,0,0,,,,,,,,,,,,,
```

Mobile Twinned Call Picked Up Using the Twinning Button

This is the same scenario as the example above, however after answering the call on the external twinned device, the user has picked it up internally by using a twinning button. The first two records are for the answered external call and are output when that call is picked up by the internal extension. The third record is output when the call is ended internally.

```
… 16:19:18,00:00:05,11,207,O,203,203,,1,1000029,1,E207,Extn207,E203,Extn203,0,0,,,,,,,,,,,,,
… 16:19:20,00:00:05,9,,O,9416,9416,,0,1000030,0,E203,Extn203,T9005,Line 5.1,0,0,,,,,,,,,,U MT,Extn203,9416
… 16:19:18,00:00:05,0,207,O,203,203,,1,1000029,0,E207,Extn207,E203,Extn203,0,0,,,,,,,,,,,,,
```

External Conference Party

This is similar to internal conferencing (see examples above) but the conference setup and progress records include External Targeting Cause codes for user (U) conference proposal (CfP) and user (U) conferenced (Cfd).

```
… 16:48:58,00:00:02,2,203,O,9416,9416,,0,1000066,1,E203,Extn203,T9005,Line 5.1,0,0,,,,,,,,,,U CfP,Extn203,,
… 16:48:37,00:00:04,3,203,O,207,207,,1,1000064,1,E203,Extn203,E207,Extn207,7,0,,,,,,,,,,,,,
… 16:49:04,00:00:08,0,203,O,9416,9416,,1,1000067,0,E203,Extn203,V11002,CO Channel 100.2,0,0,,,,,,,,,,,,,
… 16:48:37,00:00:13,0,,O,,,,1,1000064,0,E207,Extn207,V11003,CO Channel 100.3,0,0,,,,,,,,,,,,,
… 16:48:58,00:00:13,0,,O,9416,9416,,0,1000066,0,V11001,CO Channel 100.1,T9005,Line 5.1,0,0,,,Extn203,,,,,,,,U Cfd,Extn203,
```

Call Routed by Incoming Call Route

Call from external number 403 rerouted by incoming call route (ICR) for incoming line group 701 back out to 404.

```
2008/08/01 11:45:36,00:00:01,2,403,I,9404,,,0,1000007,0,T9001,Line 1.0,T9010,Line 10.0,0,0,n/a,0,,,,,,,,,ICR,ICR701,404
```

Two Outgoing External Calls Transferred Together

This scenario shows an outgoing call which is then transferred to another outgoing call.

```
2009/02/19 11:13:26,00:00:06,0,203,O,9403,9403,,0,1000012,1,E203,Extn203,T9001,Line 1.0,8,0,n/a,0,,,,,,,,,U,Extn203,,
2009/02/19 11:13:36,00:00:02,0,203,O,8404,8404,,0,1000013,0,E203,Extn203,T9002,Line 2.0,0,0,n/a,0,,,,,,,,,U XfP,Extn203,,
2009/02/19 11:13:26,00:00:11,0,8404,I,404,,,0,1000012,0,T9002,Line 2.0,T9001,Line 1.0,0,0,n/a,0,,,,,,,,,LINE Xfd,0.1038.0 13 Alog Tr
```

Chapter 15.
Appendix: CDR Records

15. Appendix: CDR Records

Note that CDR output is not supported by IP Office 5+.

There are a number of formats available for CDR output. Each format consist of two types of records; date records and call detail records.

- Date Records
  A date record is sent each time a CDR connection is started and then once a day (at midnight). The date can be in month/day or day/month format, as selected on the System | CDR 179 tab.
- Call Detail Records
  Call detail records are sent at the termination of a call. For some formats, additional fields can be selected using the Normal, Enhanced, or ISDN options on the System | CDR 179 tab.

CDR Record Formats			
Record Format	Record Options		
	Normal	Enhanced	ISDN
Printer 924	✓	✓	✓
59-Character 914	✓	✗	✗
Expanded 915	✓	✓	✗
LSU 920	✓	✓	✓
LSU Expanded 923	✓	✗	✗
INT Direct 917	✓	✗	✗
INT ISDN 918	✓	✗	✗
INT Process 919	✓	✗	✗
Teleseer 927	✓	✓	✓
Unformatted 930	✓	✓	✗

15.1 CDR Record Fields

The following list describes the fields which, depending on the selected report format and options, may be included in the CDR records.

Those fields shown in italics are not supported by CDR. Where the report format includes such a field, the data is replaced by a space or spaces. Similarly fields not appropriate to the call type are replaced by a space or spaces as appropriate.

- Access Code Dialed
  The access code the user dialed to place an outgoing call. This will be the digit used to trigger secondary dial tone if used.
- Access Code Used
  The number of the line used for an outgoing call.
- Account Code
  This field may contain a number to associate call information with projects or account numbers. For some formats, a long account code overwrites spaces on the record that are assigned to other fields.
- *Attendant Console - Not supported.*
- *Authorization Code - Not supported.*
- *Bandwidth - Not supported.*
- BCC (Bearer Capability Class)
  This field identifies the type of ISDN call. Any one of the following may appear in this field.
  - 0 = Voice Grade Data and Voice.
  - 1 = Mode 1 (56 Kbps synchronous data).
  - 2 = Mode 2 (less than 19.2 Kbps synchronous or asynchronous data).
  - 3 = Mode 3 (64 Kbps data for LDAP protocol).
  - 4 = Mode 0 (64 Kbps data clear).
- Calling Number
  For outgoing or intra-switch calls, this field contains the extension number of the originating telephone user. For incoming and tandem calls, this field contains the trunk access code in standard formats. The fifth digit is the first digit of a 5-digit dialing plan. In formats where the field is less than 7 digits, this also shows the trunk access code of the incoming call.

  This field shows the calling party number in Unformatted or Expanded records. If the calling party number is not available, this field is blank for both formats.
- Calling Number/Incoming Trunk Access Code
  For incoming calls this field contains the incoming trunk access code. For outgoing calls, this field contains the calling extension.
- Carriage Return
  The ASCII carriage return character followed by a line feed indicates the end of a call record.
- Condition Code
  The condition code indicates what type of call the record describes. For example, condition code C indicates a conference call, 7 indicates an ARS call, etc. The table below shows condition codes for most record formats. The 59-character format uses different condition codes from those used for other record types.

Code	59	Description
0	-	Identifies an outgoing intra-switch call (a call that originates and terminates on the switch).
9	I	Identifies an incoming external call.
A	-	Identifies an outgoing external call.
C	L	Identifies a conference call.
E	N	An incomplete external call, due to all trunks being busy or out of service. Incoming trunk calls to a busy phone do not generate a CDR record.
G	-	Indicates a call terminating to a ringing station.
H	-	Indicates that a ringing call that was then abandoned.
I	-	Indicates a call attempt to a busy station.

CDR can also record the ring time to answer or abandon for incoming calls originated by the trunk group. In addition, CDR indicates if the incoming destination is busy. This record is separate from the normal call duration record printed for an answered call. This information is indicated by the condition code.

When an incoming call is terminated to an internal destination, the call is tracked from the time ringing feedback is given to the originator. If the call is answered, a CDR record is printed with the condition code G and the duration reflects the time between the start of ringing and the answer of the call. If the call is abandoned before being answered, the system prints a record with the condition code H and the duration reflects the time between the start of ringing and the time the call was abandoned. If the destination is busy, a CDR record is printed with the condition code I and a duration of 0.

- Dialed Number
  This field contains the number dialed. If it is an outgoing call, the field contains the number dialed by a system user. If it is an incoming call, the field contains the extension that was dialed. If more than 18 digits are dialed, the least significant digits (starting from the right) are truncated.
- Duration
  This is the duration of the call or call segment. It is recorded in hours, minutes and tenths of minutes. Calls are rounded down in 6-second increments. Therefore, a call of 5-second duration will be indicated as 0 duration. If 9999 appears in this field, this call was in progress when a time change was made in the switch.
- Feature Flag
  1 for a data call, 0 for voice calls.
- *Format Code - Not supported.*
- *FRL - Not supported.*
- Incoming Circuit Id.
  This field identifies the trunk used for an incoming call. For outgoing calls this field is blank.
- Incoming Trunk Access Code
  This field contains the access code of the incoming trunk group.
- *ISDN Network Service - Not supported.*
- *ISDN CC - Not supported.*
- *IXC (Interexchange Carrier Code) - Not supported.*
- Line Feed
  The ASCII line feed character follows a carriage return to terminate CDR records.
- MA-UUI (Message Associated User-to-User Signaling) - *Not supported.*
- *Node Number - Not supported.*
- Null
  Used to terminate and divide CDR Records (usually in triplets) when needed.
- Outgoing Circuit Id.
  For outgoing calls, this field identifies the trunk used.
- *Packet Count - Not supported.*
- *PPM (Periodic Pulse Metering) - Not supported.*
- *Resource Flag - Not supported.*
- Space
  The ASCII space character separates other CDR fields or fills unused record locations.
- *TSC Flag - Not supported.*
- Time
  This fields contains the time that the call ended, or the time that a user dropped from a multi-party call.

15.2 Call Splitting

Call splitting keeps track of calls where more than two parties are involved. These can be calls that are transferred or conferenced. When any of these situations arise, CDR produces a separate record for each new party involved in the call.

Conference

Caller A makes an incoming trunk call to switch party B (201). They talk for 2 minutes, then B conferences in C (202), and D (203). The entire group talks for another 8 minutes, at which point B drops off the call. This produces a record for segment A–B.

A, C and D continue to talk for another 5 minutes. All remaining parties drop, producing two more records; A–C and A–D. Note that each record shows the incoming trunk ID as the calling number.

Segment	Duration	Condition Code	Calling Number	Dialed Number
A-B	0:10:0	C	123	201
A-C	0:13:0	C	123	202
A-D	0:13:0	C	123	203

Transfer

A calls B (201). They talk for 1 minute, then B transfers the call to C (202). CDR generates a record for segment A–B. A and C talk for 5 minutes. CDR then generates a record for segment A–C.

Segment	Duration	Condition Code	Calling Number	Dialed Number
A-B	0:01:0	9	123	201
A-C	0:05:0	9	123	202

Trunk to Trunk Transfer

A calls switch party B (201), they talk for one minute. B transfers the call to public-network party E (5665555), they talk for 4 minutes. Note that the duration of the original incoming trunk call includes the time after the call was transferred to an outgoing trunk, until all trunk parties drop.

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
A-B	0:05:0	9		123	201
A-E	0:04:0	9	345	123	5665555

15.3 Record Formats

15.3.1 59 Character (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-2	Time of day-hours
3-4	Day *	3-4	Time of day-minutes
5	Carriage return	5	Duration-hours
6	Line feed	6-7	Duration-minutes
7-9	Null	8	Duration-tenths of minutes
*Leading 0 added if needed.		9	Condition code 911
		10-12	Access code dialed
		13-15	Access code used
		16-30	Dialed number
		31-35	Calling number
		36-50	Account code
		51	*FRL*
		52	*IXC*
		53-55	Incoming circuit ID
		56-58	Outgoing circuit ID
		59	Carriage return
		60	Line feed
		61-63	Null

15.3.2 Expanded (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	**Field Description**	**Position**	**Field Description**
1-2	Month *	1-2	Time of day-hours
4-5	Day *	3-4	Time of day-minutes
6	Carriage return	6	Duration-hours
7	Line feed	7-8	Duration-minutes
8-10	Null	9	Duration-tenths of minute
*Leading 0 added if needed.		11	Condition code 911
		13-16	Access code dialed
		18-21	Access code used
		23-37	Dialed number
		39-48	Calling number
		50-64	Account code
		66-72	*Authorization code*
		77	*FRL*
		79-81	Incoming circuit ID
		83-85	Outgoing circuit ID
		87	Feature flag
		89-90	*Attendant console*
		92-95	Incoming trunk access code
		97-98	*Node number*
		100-102	ISDN NSV
		104-106	*IXC*
		108	Bearer Capability Class
		110	*MA-UUI*
		112	*Resource flag*
		114-117	*Packet count*
		119	TSC flag
		121-129	Reserved
		131	Carriage return
		132	Line feed
		133-135	Null

15.3.3 Expanded (Enhanced)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	**Field Description**	**Position**	**Field Description**
1-2	Month *	1-2	Time of day-hours
4-5	Day *	3-4	Time of day-minutes
6	Carriage return	6	Duration-hours
7	Line feed	7-8	Duration-minutes
8-10	Null	9	Duration-tenths of minutes
*Leading 0 added if needed.		11	Condition code 911
		13-16	Access code dialed
		18-21	Access code used
		23-37	Dialed number
		39-48	Calling number
		50-64	Account code
		66-72	*Authorization code*
		74-75	Time in queue
		77	*FRL*
		79-81	Incoming circuit ID
		83-85	Outgoing circuit ID
		87	Feature flag
		89-90	*Attendant console*
		92-95	Incoming TAC
		97-98	*Node number*
		100-102	ISDN NSV
		104-107	*IXC*
		109	Bearer Capability Class
		111	*MA-UUI*
		113	*Resource flag*
		115-118	*Packet count*
		120	TSC flag
		122-123	*Bandwidth*
		125-130	*ISDN CC (digits 1–6)*
		131-135	*ISDN CC (digits 7–11) /PPM count (1–5)*
		136-146	Reserved for future use
		147	Carriage return
		148	Line feed
		149-151	Null

15.3.4 INT-Direct (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	**Field Description**	**Position**	**Field Description**
1-2	Month *	1-2	Day of month
3-4	Day *	3-4	Month
5	Carriage return	5-6	Year
6	Line feed	8-9	Time of day-hours
7-9	Null	10-11	Time of day-minutes
*Leading 0 added if needed.		13	Duration-hours
		14-15	Duration-minutes
		16	Duration-tenths of minutes
		18	Condition code 911
		20-22	Access code dialed
		23-25	Access code used
		27-44	Dialed number used
		46-50	Calling number
		52-66	Account code
		68-72	*PPM count*
		74-75	Incoming circuit ID
		77-78	Outgoing circuit ID
		79	Carriage return
		80	Line feed

15.3.5 INT-ISDN (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-2	Time of day-hours
3-4	Day *	3-4	Time of day-minutes
5	Carriage return	5	Space
6	Line feed	6	Duration-hours
7-9	Null	7-8	Duration-minutes
*Leading 0 added if needed.		9	Duration-tenths of minutes
		11	Condition code 911
		13-16	Access code dialed
		18-21	Access code used
		23-37	Dialed number
		39-48	Calling number
		50-64	Account code
		66-72	*Authorization code*
		74	Line feed
		76	*FRL*
		78	Incoming circuit ID (hundreds)
		79	Incoming circuit ID (tens)
		80	Incoming circuit ID (units)
		82-84	Outgoing circuit ID
		86	Feature flag
		88-89	*Attendant console (1st digit)*
		91-94	Incoming trunk access code
		96-97	*Node number*
		99-101	ISDN NSV
		103-106	*IXC*
		108	Bearer Capability Class
		110	*MA-UUI*
		112	*Resource flag*
		114-119	Reserved
		120-124	*PPM*
		132	Carriage return
		133	Line feed
		134-136	Null

15.3.6 INT-Process (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	**Field Description**	**Position**	**Field Description**
1-2	Month *	*1-2*	*Format code*
3-4	Day *	3-4	Time of day-hours
5	Carriage return	5-6	Time of day-minutes
6	Line feed	7	Duration-hours
7-9	Null	8-9	Duration-minutes
*Leading 0 added if needed.		10	Duration-tenths of minutes
		12	Condition code 911
		14-16	Access code dialed
		17-19	Access code used
		21-38	Dialed number (digits 1–18)
		39-43	Calling number (digits 1–5)
		45-59	Account code (digits 1–15)
		61	*IXC*
		62	*FRL*
		66-67	Incoming circuit ID (digits 1–2)
		71-72	Outgoing circuit ID (digits 1–2)
		74-78	*PPM (digits 1–5)*
		79	Carriage return
		80	Line feed
		81-83	Null

15.3.7 LSU (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Hour *	1	Duration-hours
3	Colon (:)	2-3	Duration-minutes
4-5	Minute *	4	Duration-tenths of minutes
6	Blank	5	Condition code 911
7-8	Month *	6-8	Access code dialed
9	Slash (/)	9-11	Access code used
10-11	Day *	12-26	Dialed number
12	Carriage return	27-30	Calling number (digits 2–5)
13	Line feed	31-35	Account code (first 5 digits)
14-16	Null	36-42	*Authorization code* or digits 6–12 of account code
*Leading 0 added if needed.		43-44	Space or digits 13–14 of account code
		45	*FRL* or digit 15 of account code
		46	Calling number (1st digit)
		47-48	Incoming circuit ID (tens, units)
		49	Feature flag
		50-52	Outgoing circuit ID (tens, units, hundreds)
		53	Incoming circuit ID (hundreds)
		54	*IXC*
		55	Carriage return
		56	Line feed
		57-59	Null

15.3.8 LSU (Enhanced)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	**Field Description**	**Position**	**Field Description**
1-2	Hour *	1	Duration-hours
3	Colon (:)	2-3	Duration-minutes
4-5	Minute *	4	Duration-tenths of minutes
6	Blank	5	Condition code 911
7-8	Month *	*6-9*	*IXC*
9	Slash (/)	10-12	Access code used
10-11	Day *	13-27	Dialed number
12	Carriage return	28-31	Calling number
13	Line feed	32-35	Account code (digits 1–4)
14-16	Null	36-42	*Authorization code* or digits 6–12 of account code
*Leading 0 added if needed.		43-45	ISDN NSV
		46	1st digit of a 5-digit calling number
		47-48	Incoming circuit ID (tens, units)
		49	Feature flag
		50-52	Outgoing circuit ID (tens, units, hundreds)
		53	Incoming circuit ID (hundreds)
		54	*FRL*
		55	Carriage return
		56	Line feed
		57-59	Null

15.3.9 LSU (ISDN)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Hour *	1	Duration-hours
3	Colon (:)	2-3	Duration-minutes
4-5	Minute *	4	Duration-tenths of minutes
6	Blank	5	Condition code 911
7-8	Month *	*6-8*	*IXC*
9	Slash (/)	9-11	Access code used
10-11	Day *	12-26	Dialed number
12	Carriage return	27-30	Calling number (digits 2–5)
13	Line feed	31-35	Account code (digits 1–5)
14-16	Null	36-42	*Authorization code* or digits 6–12 of account code
*Leading 0 added if needed.		43-44	ISDN NSV or digits 13–14 of account code
		45	ISDN NSV (3rd digit) or *FRL*, or digit 15 of account code
		46	Calling number (1st digit)
		47-48	Incoming circuit ID (tens, units)
		49	Feature flag
		50-52	Outgoing circuit ID (tens, units, hundreds)
		53	Incoming circuit ID (hundreds)
		54	*FRL*
		55	Carriage return
		56	Line feed
		57-59	Null

15.3.10 LSU-Expanded

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Hour *	1-2	Time of day-hours
3	Colon (:)	3-4	Time of day-minutes
4-5	Minute *	6	Duration-hours
6	Blank	7-8	Duration-minutes
7-8	Month *	9	Duration-tenths of minutes
9	Slash (/)	11	Condition code 911
10-11	Day *	13-15	Access code dialed
12	Carriage return	16-18	Access code used
13	Line feed	20-34	Dialed number
14-16	Null	36-39	Calling number
*Leading 0 added if needed.		41-45	Account code
		47-53	*Authorization code*
		58	*FRL*
		60	Calling number (1st digit)
		62-63	Incoming circuit ID (tens, units)
		65	Feature flag
		67-68	Outgoing circuit ID (tens, units)
		70	Outgoing circuit ID (hundreds)
		72	Incoming circuit ID (hundreds)
		73	*IXC*
		74	Carriage return
		75	Line feed
		76-78	Null

15.3.11 Printer (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-2	Time of day-hours
4-5	Day *	3-4	Time of day-minutes
6	Carriage return	6	Duration-hours
7	Line feed	7-8	Duration-minutes
8-10	Null	9	Duration-tenths of minutes
*Leading 0 added if needed.		11	Condition code 911
		13-15	Access code dialed
		17-19	Access code used
		21-35	Dialed number
		37-41	Calling number
		43-57	Account code
		59-65	*Authorization code*
		70	*FRL*
		72	*IXC*
		74-76	Incoming circuit ID
		78-80	Outgoing circuit ID
		82	Feature flag
		83	Carriage return
		84	Line feed

15.3.12 Printer (Enhanced)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-2	Time of day-hours
4-5	Day *	3-4	Time of day-minutes
6	Carriage return	6	Duration-hours
7	Line feed	7-8	Duration-minutes
8-10	Null	9	Duration-tenths of minutes
*Leading 0 added if needed.		11	Condition code 911
		13-16	*IXC*
		18-21	Access code used
		23-37	Dialed number
		39-43	Calling number
		45-59	Account code
		61-67	*Authorization code*
		69-71	ISDN NSV
		73	*FRL*
		75-77	Incoming circuit ID
		79-81	Outgoing circuit ID
		83	Feature flag
		84	Carriage return
		85	Line feed

15.3.13 Printer (ISDN)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-2	Time of day-hours
4-5	Day *	3-4	Time of day-minutes
6	Carriage return	6	Duration-hours
7	Line feed	7-8	Duration-minutes
8-10	Null	9	Duration-tenths of minutes
*Leading 0 added if needed.		11	Condition code 911
		13-15	*IXC*
		17-19	Access code used
		21-35	Dialed number
		37-41	Calling number
		43-57	Account code
		59-65	*Authorization code*
		67-68	ISDN NSV (hundreds, tens)
		70	ISDN NSV (units)
		72	*FRL*
		74-76	Incoming circuit ID
		78-80	Outgoing circuit ID
		82	Feature flag
		83	Carriage return
		84	Line feed

15.3.14 Teleseer (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-3	Space
3-4	Day *	4-5	Time of day-hours
5	Carriage return	6-7	Time of day-minutes
6	Line feed	8	Duration-hours
7-9	Null	9-10	Duration-minutes
*Leading 0 added if needed.		11	Duration-tenths of minutes
		12	Condition code 911
		13-15	Access code dialed
		16-18	Access code used
		19-33	Dialed number
		34-38	Calling number
		39-53	Account code
		54	*FRL*
		55	*IXC*
		56-58	Incoming circuit ID
		59-61	Outgoing circuit ID
		62	Feature flag
		63-69	*Authorization code*
		70-76	Space
		77	Carriage return
		78	Line feed
		79-81	Null

15.3.15 Teleseer (Enhanced)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-3	Space
3-4	Day *	4-5	Time of day-hours
5	Carriage return	6-7	Time of day-minutes
6	Line feed	8	Duration-hours
7-9	Null	9-10	Duration-minutes
*Leading 0 added if needed.		11	Duration-tenths of minutes
		12	Condition code 911
		13-16	*IXC*
		17-19	Access code used
		20-34	Dialed number
		35-39	Calling number
		40-54	Account code
		55	ISDN NSV (units)
		56	*FRL*
		57-59	Incoming circuit ID
		60-62	Outgoing circuit ID
		63	Feature flag
		64-70	*Authorization code*
		71-72	ISDN NSV (hundreds, tens)
		77	Carriage return
		78	Line feed
		79-81	Null

15.3.16 Teleseer (ISDN)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Month *	1-3	Space
3-4	Day *	4-5	Time of day-hours
5	Carriage return	6-7	Time of day-minutes
6	Line feed	8	Duration-hours
7-9	Null	9-10	Duration-minutes
*Leading 0 added if needed.		11	Duration-tenths of minutes
		12	Condition code 911
		13-15	*IXC*
		16-18	Access code used
		19-33	Dialed number
		34-38	Calling number
		39-53	Account code
		54	ISDN NSV (units)
		55	*FRL*
		56-58	Incoming circuit ID
		59-61	Outgoing circuit ID
		62	Feature flag
		63-69	*Authorization code*
		70-71	ISDN NSV (hundreds, tens)
		77	Line feed
		78-80	Null

15.3.17 Unformatted (Normal)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Hour *	1-2	Time of day-hours
3	Colon (:)	3-4	Time of day-minutes
4-5	Minute *	5	Duration-hours
6	Blank	6-7	Duration-minutes
7-8	Month *	8	Duration-tenths of minutes
9	Slash (/)	9	Condition code 911
10-11	Day *	10-13	Access code dialed
12	Carriage return	14-17	Access code used
13	Line feed	18-32	Dialed number
14-16	Null	33-42	Calling number
*Leading 0 added if needed.		43-57	Account code
		58-64	*Authorization code*
		67	*FRL*
		68-70	Incoming circuit ID
		71-73	Outgoing circuit ID
		74	Feature flag
		75-76	*Attendant console*
		77-80	Incoming TAC
		81-82	*Node number*
		83-85	ISDN NSV
		86-88	*IXC*
		89	Bearer Capability Class
		90	*MA-UUI*
		91	*Resource flag*
		92-95	*Packet count*
		96	TSC flag
		97-100	Reserved
		101	Carriage return
		102	Line feed
		103-105	Null

15.3.18 Unformatted (Enhanced)

Those fields shown in italics are not supported by CDR and are replaced by a space or spaces. Those positions not listed are spaces.

Date Record		Call Details Record	
Position	Field Description	Position	Field Description
1-2	Hour *	1-2	Time of day-hours
3	Colon (:)	3-4	Time of day-minutes
4-5	Minute *	5	Duration-hours
6	Blank	6-7	Duration-minutes
7-8	Month *	8	Duration-tenths of minutes
9	Slash (/)	9	Condition code 911
10-11	Day *	10-13	Access code dialed
12	Carriage return	14-17	Access code used
13	Line feed	18-32	Dialed number
14-16	Null	33-42	Calling number
*Leading 0 added if needed.		43-57	Account code
		58-64	*Authorization code*
		65-66	Time in queue
		67	*FRL*
		68-70	Incoming circuit ID
		71-73	Outgoing circuit ID
		74	Feature flag
		75-76	*Attendant console number*
		77-80	Incoming TAC
		81-82	*Node number*
		83-87	ISDN NSV
		88-89	*IXC*
		90	Bearer Capability Class
		91	*MA-UUI*
		92	*Resource flag*
		93-96	*Packet count*
		97	TSC flag
		98-99	*Bandwidth*
		100-105	*ISDN CC (digits 1–6)*
		106-110	*ISDN CC (digits 7–11)/PPM*
		111-114	Reserved for future use
		115	Carriage return
		116	Line feed
		117-119	Null

Chapter 16.
Appendix: Release History

16. Appendix: Release History

16.1 What was New in Release 6.1

IP Office 6.1 is supported on IP406v2, IP412, IP500 and IP500v2 systems.

Phones

The following additional phones and phone features are supported by IP Office 6.1:

- **1000 Series**
  The 1010 and 1040 SIP devices are HD video softphone devices. The main units provided connections for a variety of video and audio inputs and outputs.
- **1100/1200 Series**
  The following phones from the Avaya range of SIP phones are supported: 1120E, 1140E, 1220 and 1230.
- **Automatic Call Log Expiry**
  For call log entries written into a user's centralized call log, a expiry time can be set after which the call log entry is automatically deleted from the user's call log.
- **Mobile Twinning Handover** (IP Office Release 6.1)
  When on a call on the primary extension, pressing the **Twinning** button will make an unassisted transfer to the twinning destination. This feature can be used even if the user's **Mobile Twinning** setting was not enabled.
  - During the transfer process the button will wink.
  - Pressing the twinning button again will halt the transfer attempt and reconnect the call at the primary extension.
  - The transfer may return if it cannot connect to the twinning destination or is unanswered within the user's configured **Transfer Return Time** (if the user has no **Transfer Return Time** configured, a enforced time of 15 seconds is used).
- **Show Last Call Duration**
  1400, 1600 and 9600 phones will briefly display the duration of a call after it is ended. This setting is a user accessible option through the phone menu **Features | Call Settings | Show Last Call Duration** (except on 1403 and 1603 phones where it is on by default).

Small Community Network Management

Manager 8.1 is able to load and display the configuration of all systems in a Small Community Network. The configurations can be edited and saved. When working in SCN management mode, Manager can also display a graphic view of the network and allow the adding of additional systems and SCN lines between systems.

one-X Portal for IP Office

IP Office 6.1 supports one-X Portal for IP Office 6.1. one-X Portal for IP Office 6.1 provides the following new features:

- **Adjustable Layout**
  The gadgets provided on the one-X Portal for IP Office's main page can now be moved, resized and minimized by the user. The users positioning of the gadgets is retained between one-X Portal for IP Office sessions.
- **Multi Directory Search**
  It is now possible to display results for a search across all directories (System, Personal and External).
- **Call Assistant**
  The one-X Portal for IP Office Call Assistant is now supported for Windows based users of one-X Portal for IP Office. The Call Assistant provides pop up messages about calls even when not logged into one-X Portal for IP Office, screen popping to Outlook and hot key dialing of numbers.

IP Office Customer Call Reporter

- **Microsoft SQL 2008 Support**
  IP Office Customer Call Reporter is now supported using Microsoft SQL 2008.
- **Visual Redesign**
  Some elements of the browser display have been redesigned.
  - Many of the tabs used by Supervisors to move between the available pages have been replaced by icons.
  - The color used for warnings has been changed from yellow to orange.
- **New Supervisor Pages**
  Supervisors are able to access two new pages of call information in addition to the views that they share with their agents.
  - **Dashboard Display**
    The dashboard display is the default page displayed to a supervisor when they login. They can customize it to display a combination of up to three graphs and data display elements.
  - **Customer Map**
    Supervisors can display and configure a map that will plot calls based on caller ID numbers. The map can be used to show a combination of historical and realtime calls and can be overlayed onto Google or Yahoo maps.
- **Force Agent State**
  Supervisors can now force force a change to an agent's status. For example log an agent out or enable/disable an agents queue membership. The IP Office Customer Call Reporter administrator configures which supervisors have this function. This feature requires the IP Office Customer Call Reporter server to have access to a one-X Portal for IP Office server.
- **New Statistics**
  The following new statistics are available for use in views.
  - Agent Productivity Factor
  - Talk Average
  - Talk Inbound
  - Talk Inbound Average
  - Talk Internal
  - Talk Outbound
  - Talk Outbound Average
  - Talk Total
- **Report Templates**
  The follow changes have been made for historical reporting.
  - The **Call Summary Report** now includes **Average Answer Time** when reporting on agents (previously this value was blank when reporting on an agent or agents).
  - A new template called the **Agent Report Card** template has been added. It provides historical reporting on the Talk Time statistics.
- **Message Color**
  When scheduling a wallboard message, supervisors can now select the color for the message.
- **IP Office Customer Call Reporter Help**
  IP Office Customer Call Reporter help is now also available in French and Latin Spanish.
- **Wallboard Controls**
  The wallboard **Background and Content Settings** now include options to adjust the animation effect applied to changing statistic values and the aspect ratio used for the display of the wallboard elements.

Embedded Voicemail

- **Skip Your Mailbox Greeting**
  Caller's can skip your mailbox greeting by pressing 1. Instead they immediately hear the tone for the start of recording.

IP Office Essential Edition - PARTNER® Version

- **Phantom Users**
  Previously user settings were only created and configurable for the physical extensions present in the system (and excluding ports 7 and 8 on ETR6 cards). User settings are now created for all possible user extensions regardless of whether a matching physical extension port is present or not.
  - Calls to a phantom extension number go directly to that user's mailbox. This applies for normal dialing, DDI call routing, line coverage, transfers and routing from an auto-attendant.
- **Auto Attendant Enhancements**
  The following changes have been made to auto attendant support for IP Office Essential Edition - PARTNER® Version 6.1.

- **9 Auto Attendants**
  Up to 9 auto attendants are now supported. Where a line is associated with an auto attendant the auto attendant required can be selected.
  - **Transfer to Auto Attendant Action**
    This additional menu action allows calls to be routed from one auto attendant to another. When this occurs, only the menu prompt of the new auto attendant is played.
- **Language Selection**
  Each auto attendant can be configured with a language selection. The selected language controls the prompts used by the auto attendant actions where applicable.
- **Time Profile Dependant Menu Actions**
  In addition to controlling with initial greeting is played to callers, each auto attendants time profiles now also control which set of menu actions are available to the callers with the appropriate time dependant menu actions greeting.
- **Selectable Night Service Mode**
  Previously when the system was put into night service, the auto attendant switched to its out of hours greeting. Each auto attendant now has a **Follow Night Service** setting. If selected, the previously behavior still applies. If not selected, when the system is put into night service, the auto attendant continues to follow its own time profile settings.
- **Picking Up Auto Attendant Calls**
  Bridging into a call being handled by an auto attendant drops the auto attendant from the call.
- **Emergency Greeting**
  Each auto attendant can have a recorded emergency greeting and a setting for whether that greeting is active or not. When active, the emergency greeting is played before any of the other auto attendant greetings. When an emergency greeting is active, a warning is displayed on extensions 10 and 11.
  - **Transfer to Greeting Action**
    This additional menu action allows the caller to play and record the emergency greeting. It also allows them to select whether the greeting is active or not. If a system password has been set, it is used to restrict access to this option.

- **Line Enhancements**
  - **Unique Line Ringing**
    Each incoming line, channel or DID can be assigned a ringing pattern. That pattern is then used for incoming calls on that line unless overridden by the user's setting.
  - **Assigning Lines to Auto Attendants**
    Previously assignment of a line to an auto attendant could only be done through the Manager. This option can now be selected through the administrator menus on extensions 10 and 11.
- **User Setting Enhancements**
  The following changes have been made to user support for IP Office Essential Edition - PARTNER® Version 6.1.
  - **Immediate Voicemail**
    The user setting for VMS Cover Ring can now be set to *0* for immediate voicemail.
  - **Transfer Return Extension**
    For users who have transfer return enabled, a different extension destination for the return calls can now be specified.
  - **Override Line Ringing**
    For selected users the new Unique Line Ringing can be overridden.
- **Language Control**
  Previously only individual users could be configured with a language selection. For IP Office Essential Edition - PARTNER® Version 6.1 the default language selection can now be done at the system level. In addition a language setting can be applied to each auto attendant.
- **Embedded Voicemail Enhancements**
  The following changes have been made to the embedded voicemail provided to IP Office Essential Edition - PARTNER® Version 6.1 users.
  - **Caller Post Message Options**
    After leaving a mailbox message, callers can now press # rather than hanging up immediately. The caller will hear a prompt informing them whether the message has been saved or whether the messages was too short (less than 3 seconds) and so was not saved.
  - **Skip Your Mailbox Greeting**
    Caller's can skip your mailbox greeting by pressing 1. Instead they immediately hear the tone for the start of recording.
  - **Mark Messages as New**
    You can now change an old or saved message's status back to new while it is being played or just after it has played by dialing ***06**. The message waiting light is relit. However if voicemail email is being used no new message email is sent.

IP Office Essential Edition - Norstar Version

To replace Avaya Norstar systems in the middle eastern market, IP500v2 systems can be configured to run in IP Office Essential Edition - Norstar Version mode. This is done using a IP Office Essential Edition - Norstar Version System SD card. Operation is similar to IP Office Essential Edition - PARTNER® Version except with no support for ETR phones.

- The locales supported for IP Office Essential Edition - Norstar Version are: Bahrain, Egypt,, Kuwait, Morocco, Oman, Pakistan, Qatar and the United Arab Emirates.
- Automatic impedance matching is also available for the locales above.
- For embedded voicemail, Arabic language prompts are supported.

Voicemail Pro

The following features have been added to Voicemail Pro 6.1.

- **Additional Generic Action String Manipulation Options**
  The **String Manipulation** command has two additional options. They are:
  - **Copy**
    This action can be used to copy the value of one variable to another variable. The command can copy the whole value or can, treating the value as a string, copy a section to or from a specified matching character.
  - **Length**
    This action can be used to return the length of variable. It can return the full length or the length from or to a specified matching character.
- **Post Call Completion Call Flows**
  Call flows can be configured to continue running even after the caller has disconnected. If the current action which the call had reached has a *Timeout* or *Next* result, the connection from that result is followed immediately until the call flow either reaches a **Disconnect** action or an unconnected result.
- **Automatic Call Recording for Internal Calls**
  The user and hunt group options for call recording can now be set to work for internal calls. Previously they only applied to external calls.

Music on Hold

The following changes have been made for hold music support:

- **Analog Extension Source**
  For alternate music on hold sources 2 to 4, the audio input to an analog extension port can be specified as the source.
- **Short Code Music on Hold Selection**
  For calls routed by short codes, including outgoing external calls, the hold music source for the call can be specified using the h character in the short code telephone number.

IP Office Application Server

The IP Office Application Server is a Linux based server running server applications, currently Voicemail Pro and one-X Portal for IP Office. It does not require Linux knowledge to install. The sever and the applications are managed remotely via web browser and the existing Voicemail Pro client.

Miscellaneous Features

- **Restrict Analog Extension Ringer Voltage**
  If this system setting is selected, the available message waiting lamp indication settings for analog extensions are restricted to *None* and *Line Reversal*.
- **Time Support**
  A number of changes have been made to the way that the systems time and date can be set.
  - The existing RFC868 Time method using the Voicemail Pro server or Manager as the source is still supported. Use of other RFC868 server sources is not supported.
  - Using SNTP requests to a NTP server is now supported. When using SNTP, additional fields for the local time offset and for daylight savings time (DST) settings are also available.
  - If the time is being set manually through a system phone, the system can be configured with a manually adjustable offset and or with automatic daylight savings time settings.

- The time settings, including the time server being used and whether DST is in use or not, can be viewed through system phones.

- **Restart IP Phones on System Upgrade**
  The Manager Upgrade Wizard now includes an option to restart all connected IP phones following a system upgrade. This will cause those phones to check and, if necessary, upgrade their phone firmware following the system upgrade.

- **Multiple SIP Proxies**
  The use of multiple SIP proxy servers in SIP line settings is now supported. The multiple proxies can be specified by IP address or obtained through DNS response (RFC3263). When specified by IP address, weighting can be applied to each address.

16.2 What was New in Release 6.0

This section summarizes the major changes introduced with IP Office 6.0

IP500v2

The IP500v2 control unit is similar to the existing IP500 control unit with 4 slots for IP500 base cards. Key differences are:

- SD Memory Card and Feature Key Dongle
  The Feature Key dongle used is an Avaya SD memory card with a unique serial number on which the IP500v2 systems licenses are based. This card must be present for the correct operation of the control unit.
  - The cards are supplied with a set of core software and phone binary files. When a system is started, it will compare its current software (plus that of phones and expansion modules) and if necessary will upgrade the system using the files on the card.
  - In addition the card can contain a license file which will be merged with the systems current configuration during a restart.
  - The card provides embedded voicemail. It provides 2 channels by default but can be licensed for up to 6 channels total.
  - The System Status application is present on the card and can be run from the card.
  - An additional SD card slot on the control unit can be used for an additional memory card to or from which files can be copied to the system card.
  - Various commands within Manager, System Status and on system phones can be used to perform memory card management function such as backing up the System card to an Optional card.
- Partner Version Mode
  In addition to A-Law and Mu-Law variants of the Feature Key, a Partner Version variant exists. IP500v2 systems using this card run in Partner Version mode, see below.
- IP500v2 Combination Cards
  These cards are a pre-built pairing of IP500 base card and daughter card. They provide 6 digital station ports, 2 analog phone ports, 10 VCM channels and either 4 analog trunk ports or 4 BRI channels. A maximum of 2 combination cards can be installed in an IP500v2 control unit. The cards will not operate if the trunk daughter card is removed or change for one of another type.
- ETR6 Cards
  This card supports 6 Partner ETR phones (ETR 6/18/34D, 3910 and 3920). The card can only be used in IP500v2 systems running in Partner Version mode. A maximum of 3 ETR6 cards can be installed.

Partner Version Mode

IP500v2 systems run in Partner Version mode by fitting them with a Partner Feature Key. In this mode the operation of the system and the telephones supported closely emulate Partner ACS systems.

- IP Office Essential Edition - PARTNER® Version systems support the following equipment.

	Maximum	Variants Supported
IP500 Base Cards	4	• Digital Station 8 (Max 3) • Phone 2 • Phone 8 • ETR6 (Max 3) • Combination Card ATM4
IP500 Daughter Cards	4	• ATM4 Uni • PRI 1 Uni (Max 1)
Expansion Modules	1	• Digital Station 16 • Phone 16
Phones	40	• 1403, 1408 and 1416 Phones • ETR 6, ETR 6D. • ETR 18, ETR 18D. • ETR 34D (Max 4 total. Max 2 per base card) • Analog phones

- The Feature Key dongle provides voicemail services for the system.
- In addition to emulating Partner ACS operation, SIP trunks and mobile twinning can be licensed.
- IP Office Essential Edition - PARTNER® Version systems cannot be included in a Small Community Networks.

Manager

- **Auto Connect at Start Up**
  By default, when started Manager will automatically performed system discover and display a list of systems found. If only when system is found it will start login to the system. Manager will also do this when an existing configuration open in Manager is closed. Auto connect can be switched off through the Manager's preferences.
- **Partner Version Administration Mode**
  When a configuration from an IP500v2 running in Partner Version mode is loaded, Manager will switch to displaying Partner Administration menus.
- **Memory Card Commands**
  Manager can be used to perform a range of memory card commands. For IP500 and IP500v2 systems it can shutdown and restart memory card operation. For IP500v2 systems it can be used to copy, create and upgrade memory cards.
- **System Shutdown**
  Manager can be used to issue a shutdown command to an IP Office 6.0 system. The shutdown can be for a specified duration in minutes or permanent until the control unit power is switched off and on again.

Features No Longer Supported

In addition to the ending of Voicemail Lite support in IP Office 5, the following applications are no longer supported in IP Office 6.0

- Delta Server
- Customer Contact Center (CCC)
- Compact Business Center (CBC)
- Conferencing Center
- Feature Key Server

Licenses

For full details of licenses used by IP Office 6.0 refer to Licenses 830.

- **Feature Key Dongle Support**
  The Feature Key Server application is not supported for systems running IP Office 6.0, meaning that the use of parallel port and USB port Feature Key dongles attached to a server PC is not supported. Therefore IP406 V2 and IP412 systems being upgraded to IP Office 6.0 must also be upgraded to using a serial port feature key.

- **System Edition Licenses** 832
  systems can be upgraded from the unlicensed Essential Edition mode to Preferred Edition mode and then to Advanced Edition mode using **Preferred Edition** and **Advanced Edition** licenses. Each license enables a range of additional features.

  - **Preferred Edition**
    Enables support for Voicemail Pro with 4 voicemail ports and licensable for additional ports. Preferred Edition voicemail features include announcements, customizable call flows, campaigns and generic TTS email reading for licensed users.

  - **Advanced Edition**
    Enables support for Customer Call Reporter and advance Voicemail Pro features (voicemail database interactions, voicemail speak text actions (8 ports generic TTS), visual basic scripting and ContactStore.

- **Software Upgrade Licenses** 832
  Existing systems being upgraded to IP Office 6.0 require a Software Upgrade license. If the license is not present following an upgrade, phones will display an unlicensed message and can only be used for emergency calls.

- **User Profiles and Profile Licenses** 835
  For IP Office 6.0, each user has a profile setting; *Basic User*, *Office Worker*, *Teleworker*, *Mobile Worker* or *Power User*. A user's profile controls whether they can be enabled for other functions. Profiles other than *Basic User* require appropriate licenses.

	Basic User	Office Worker	Teleworker	Mobile Worker	Power User
one-X Portal for IP Office	Yes[1]	Yes	Yes	–	Yes
" Telecommuter options	Yes[1]	–	Yes	–	Yes
UMS Web Services	Yes[1]	Yes	Yes	–	Yes
Mobility Features	Yes[1]	–	–	Yes	Yes
TTS for Email Reading	–	–	–	Yes	Yes
IP Office SoftPhone	–	–	Yes	–	Yes

1. These features are supported for Basic User users on systems with the appropriate pre-IP Office 6.0 legacy licenses.

  - **Teleworker Profile License**
    These licenses set the number of users who can have their profile set as *Teleworker*. For user with this optional, additional settings are enabled in the system configuration for the following services: **one-X Portal for IP Office** and **UMS Web Services**.

  - **Mobile Worker Profile License**
    Thee licenses set the number of users who can have their profile set as **Mobile Worker**. For user with this optional, additional settings are enabled in the system configuration for the following services: **Mobility Features** and **TTS for Email Reading**.

  - **Office Worker Profile License**
    These licenses set the number of users who can have their profile set as **Office Worker**. For user with this optional, additional settings are enabled in the system configuration for the following services: **one-X Portal for IP Office** (no telecommuter features) and **UMS Web Services**. If no *Office Worker Profile* licenses are present, existing legacy *Phone Manager Pro* licenses can also be used to enable users for the Office Worker Profile.

  - **Power User Profile License**
    These licenses set the number of users who can have their profile set as **Power User**. For user with this optional, the same additional services as for Teleworker and Mobile Worker are enabled for the user in the system configuration plus the following service: **SoftPhone**.

  - Pre-IP Office licenses for individual features now controlled by profile licenses can still be used by users set to the Basic User profile.

  - Changes to the licensing for TTS and user profiles mean that on systems being upgraded to IP Office 6.0, users currently enabled for TTS email reading may find that function not longer enabled.

- Avaya IP Phone Licenses 835
  On IP500 and IP500v2 systems, Avaya 1600, 4600, 5600, 9600, IP DECT, DECT R4, T3 IP, Spectralink and VPN phones are licensed by Avaya IP Endpoint licenses. These licenses are consumed by each phone as it registers with the system. Existing VCM channel resources and VCM Channel licenses enable a number of unlicensed phone registrations.
  - Existing VPN Remote licenses are no longer used. Phones using VPNremote client software require an Avaya IP Phones license.
  - Licenses are used on a first come first served basis as phones register with the system. Phones that register but cannot get a license will only be able to make emergency calls. Through the system configuration, it is possible to pre-allocate available license capacity to selected extensions.
- Text to Speech Licensing
  The licenses for Voicemail Pro text to speech functions, TTS email reading and the call flow speak text actions, have now been separated.
  - TTS email reading is now licensed and enabled by a combination of the Preferred Edition system license and the Mobile Worker Profile or Power User Profile user licenses.
  - 8 ports of generic TTS usage for use of the Voicemail Pro Speak Text action are enabled by the Advanced Edition system license. Additional ports can be enabled using the existing legacy TTS licenses.

Telephone Support

- **Avaya IP Endpoint Licenses**
  On IP500 and IP500v2 systems, Avaya 1600, 4600, 5600, 9600, IP DECT, DECT R4, T3 IP, Spectralink and VPN phones are licensed by **Avaya IP Endpoint** licenses. These licenses are consumed by each phone as it registers with the system. Existing VCM channel resources and VCM **Channel** licenses enable a number of unlicensed phone registrations.

- **Additional Phones**
  The following additional phones are supported by IP Office 6.0

  - **1400 Series Phones**
    Using digital station ports, IP Office 6.0 supports the Avaya 1400 Series phones; 1403, 1408 and 1416. To the user they provide the same range of features are the equivalent 1600 Series phones. Each 1416 can support up to 3 DBM32 buttons modules.

  - **1600-I Series Phones**
    The 1600-I variants of the 1603, 1603SW, 1608 and 1616 phones are supported.

  - **9600 Series Phones**
    The following Avaya H323 IP phones are supported: 9620L, 9620C, 9630G, 9640, 9640G, 9650 and 9650C. All except the 9620s can support up to 3 SMB24 button modules.

    - These phones are supported by IP Office 6.0+ on IP500 and IP500v2 systems only.

    - The voice activated dialing and USB features are not supported.

  - **ETR Phones and 3900 Series Phones**
    Using the ETR ports provided by the ETR6 base card, the IP500v2 control unit can support the ETR 6D, ETR 18D and ETR 34D phones plus the 3910 and 3920 wireless phones.

  - **Avaya IP Office SoftPhone**
    IP Office 6.0 supports the IP Office SoftPhone application. This application is licensed by a **SoftPhone IP Endpoint** license.

Telephony Features

- **Mobile Callback** 733
  This feature is an enhancement to the existing mobile call control features. Mobile callback allows the user to call the system and then hang up. The system will then make a call to the user's CLI and when answered, provide the user with dial tone from the system to make calls.

- **SMDR Buffer Persistence** 898
  For IP Office 6.0, the operation of SMDR on IP500 and IP500v2 control units has been enhanced to store any buffered SMDR records during any controlled system power downs or reboots. The current buffer is also stored at approximately midnight and midday.

- **Embedded Voicemail Dial by Number Enhancement** 400
  The new auto attendant setting **Direct Dial-by-Number** can be used to select how keys set to the ***Dial By Number*** operate.

  - If selected, the key press for the action is included in any following digits dialed by the caller for system extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller can dial 201 for extension 201.

  - If not selected, the key press for the action is not included in any following digits dialed by the caller for system extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller must dial 2 and then 201 for extension 201.

- **Twinned Caller Number**
  Where supported by the ISDN provider, on ISDN trunks the system can pass through the caller ID to the twinned call destination.

- **ISDN Redirected Caller Number** 197
  For twinned calls routed out on ISDN trunks, the original number of the caller can be displayed on the twinned call destination. This function is only supported on IP500 and IP500v2 systems.

- **Telephone Features**
  For 1400, 1600 and 9600 Series phones, the following features have been added:

  - **En-bloc Dialing**
    Users can select en-bloc dialing mode through the phone's own menus. In en-bloc mode the user can compose and edit the number to dial while on-hook. This is similar to the method of dialing supported by mobile cell phones and wireless handsets. The number composed is dialed from the phone when the user goes off hook or presses an appearance button.

  - **Call Park Button Enhancement**
    The operation of the Call Park button has been enhanced to display details of parked calls. This allows the user to select whether to unpark a call or to cancel the action.

- **Phone Inactivity Timers - Auto Return and Auto Lock**
  The phones now support two inactivity timers. These can be adjusted by the user through the phone's menus. For 9600 Series phones these are in addition to the phone screen saver options.

  - **Auto Return**
    When this timer expires, the phone will exit any menu and return to it normal idle phone display.

  - **Auto Lock**
    When this timer expires, the phone will be automatically locked.

- **Name Display** *(1400 and 1600 Series only)*
  Through the phone's menus the user can select to have their name displayed rather than the extension number when the phone is idle.

- **Visual Voice**
  The operation of visual voice has been enhanced. After selecting a messages category (New, Old or Saved) playback no longer starts automatically. Instead the user can scroll through a listing of the messages and select which message to play or delete.

- **Small Community Networking**
  The Small Community network user limit has been increased from 500 to 1000.

DECT R4

The following features are support for DECT R4 systems upgraded to the new firmware supplied with IP Office 6.0

- **Compact Base Station**
  This is a 4 channel base station with internal aerials. Up to 5 compact base stations can be include in a system, in conjunction with existing 8 channel base stations.

- **Directory Integration**
  Directory integration can now be done without needing a AIWS unit on the LAN. An AIWS is still required for over the air firmware upgrades to the phones.

SIP Trunk Enhancements

- SIP Trunk Prefixes 235
  The SIP line settings now include fields for defining the prefixes to be added to the caller's number. This is similar to the prefix fields used by other supported external line types. On systems where a prefix is required to make external calls, adding the prefix to the number received on incoming calls allows the call records shown in call logs and other areas to be used for return calls.

- SIP URI Parameters for Hunt Groups 322 and Voicemail
  SIP URI information can now be configured the hunt groups and for the Voicemail Pro voicemail server.

- SIP Session Timers
  Conforming to RFC4028, the system uses SIP session timers to detect whether a call is no longer connected.

- Multiple SIP Accounts per Trunk
  Previously only two sets of ITSP account details could be configured for each SIP line to a particular ITSP IP address. IP Office 6.0 allows up to 30 sets of account details to be added to the line settings.

- SIP Trunk Out of Service Detection 234
  The system can regularly check if a SIP trunk is in service. Checking that SIP trunks are in service ensures that outgoing call routing is not delayed waiting for response on a SIP trunk that is not currently useable.

  For both UDP and TCP trunks, the OPTIONS message will be regularly sent. If no reply is received the trunk is taken out of service.

  For TCP trunks, if the TCP connection is disconnected the trunk will be taken out of service.

  For trunks using DNS, if the IP address is not resolved or the DNS resolution has expired, the trunk is taken out of service.

- SIP Trunk Registration Using DNS Domain Names
  If the system is a DHCP client in a network with a DNS server, the SIP trunks can be configured to use the ITSPs domain name rather than requiring the ITSPs IP address.

one-X Portal for IP Office

For IP Office 6.0, the one-X Portal for IP Office application has been enhanced as follows:

- Conference Display and Conference Functions
  The display of conference calls has been changed and can now be used to drop parties from a conference and to mute parties in the conference including muting all.

- Profiles
  The user can now add and configure profiles. Each profile contains a range of telephone settings. The user can then use the one-X Portal to select which of their profiles is active. Profiles contain settings for mobility (forwarding, mobile twinning and telecommuting), voicemail (on/off, ringback, outcalling on/off and playback destination) and call pickup.

- Instant Messaging
  one-X Portal for users can use the application to launch instant messaging sessions between each other.

- Language Support
  In addition to English and German, the one-X Portal for IP Office also supports French, Italian, Dutch, Brazilian Portuguese and Russian.

- Personal Directory Tabs
  In addition to the default Personal tab, users can now add additional tabs onto which they can add personal contacts.

- Call Pickup
  Users can now enable call pickup for their calls by other one-X Portal users. When enabled, the user's directory entry in other user's directories will indicate when they have unanswered calls waiting and allow those calls to be picked up.

- Telecommuter Mode
  The one-X Portal for IP Office can be used in telecommuter mode.

- Voicemail Features
  The following additional options are supported with one-X Portal and Voicemail Pro 6.

  - Browser Playback
    Using one-X Portal profiles *(see above)*, the user can select to playback messages through their phone or through their browser.

  - Voicemail Greeting
    Users can upload mailbox greeting files using one-X Portal and select which greeting is active.

Customer Call Reporter

Version 1.2 of the Customer Call Reporter is supported by IP Office 5 and IP Office 6.0

- **Windows 2008 Server Support**
  Support for installation on Windows Server 2008 systems. That includes both 32-bit and 64-bit systems.

- **Switch Configuration**
  Switch discovery and selection is no longer part of the IP Office Customer Call Reporter software installation. Instead the system switch details can be edited manually by the IP Office Customer Call Reporter administrator after software installation.

- **Maximum Database Size Configuration**
  The maximum MS-SQL database size can now be set by the IP Office Customer Call Reporter administrator. The IP Office Customer Call Reporter will then provide warnings and take housekeeping actions when the actual database size approaches the configured maximum size.

- **Supervisor Scheduled Housekeeping Tasks**
  Through the scheduler tab previous used for historical reports, supervisors with **Self Administer** rights can now also configure a number of house keeping tasks. Housekeeping tasks configured by one supervisor can be seen by other supervisors. Each task can be configured to run with a specified frequency or just once. The housekeeping tasks include:

  - **Reset Realtime Statistic**
    Reset the statistics shown in all supervisor, agent and wallboard views.

  - **Backup Database**
    The backup database is placed into the default MS-SQL backups folder with a date and time prefix to the file name.

  - **Re-Index Database**
    By default this task is already scheduled and occurs at 1:30am. Re-indexing the database allows reports to run faster, however during the actual re-indexing the response of IP Office Customer Call Reporter is slowed.

  - **Reset Web Services**
    By default this task is already scheduled and occurs at 11:00pm.

- **Wallboard Display Mode**
  In addition to creating supervisor accounts, the administrator can also add wallboard accounts. When a browser is logged in using a wallboard account it displays real-time data on queues and agents in a style intended for display on large screen monitors being used a wallboards.

  - Each IP Office Customer Call Reporter supervisor license instance enables one supervisor login and also one wallboard login.

  - The wallboard display mode requires the web browser being used to support Microsoft Silverlight.

  - The wallboard display can include combinations of queue and agents statistics from all IP Office Customer Call Reporter queues and agents. For each statistic displayed, alarm and warning settings can be applied and, where appropriate, inclusion of internal and or external calls can be selected.

  - In addition to IP Office Customer Call Reporter statistics, a number of other items can be selected for the wallboard display such as:

    - **Title** - A customizable text title can be added to the wallboard display.

    - **Logo** - A logo picture file can be uploaded for display in the wallboard.

    - **Agent League Table** - A dynamic table showing the top, the bottom or a combination of the top and bottom agents for a selected queue and statistic.

    - **Graph** - Display a graph of a selected queue statistic over time.

    - **Supervisor Messages** - IP Office Customer Call Reporter supervisors can configure scheduled messages which appear in the message bar of selected wallboards.

- **Report Changes**
  The following changes have been made to the historical report templates:

  - **Connection Loss Reporting**
    If the data analyzer component of IP Office Customer Call Reporter is unable to connect to the system, it will record details in the IP Office Customer Call Reporter database. Any reports run that cover the same period as a lose of connection will include details of the connectionless period in the report.

  - **Call Summary Report Template Changes**
    In the Call Summary Report template, the *Total Calls* column is now *Call Interactions* and does not include *No Answer* calls. An additional column, *Customer Calls*, which is the number of unique calls has been added.

  - **Call Details Report Template Change**
    In the Call Details Report template, the summary sections now include totals for *Customer Calls* and *Call Interactions*.

Voicemail Pro

The following is a summary of the new features in the Voicemail Pro 6.0 release. For details of previous releases refer to the Appendix. Voicemail Pro 6.0 is supported with systems running IP Office 6.0 That includes IP406 V2, IP412, IP500 and IP500v2 systems.

Centralized Voicemail

In addition to the support for system control fallback added in IP Office 5, the following additional options are now supported within a Small Community Network using IP Office 6.0 and Voicemail Pro 6.0:

- **Backup Voicemail Server**
  An additional Voicemail Pro server can be installed. The address of this server is entered in the configuration of the central system. During normal operation; messages, call flows and other settings on the backup voicemail server are synchronized with those on the central voicemail server. If the central voicemail server becomes unavailable, the central system will switch to using the backup voicemail server for voicemail services. When the central voicemail server is restored, the central system will switch back to using it for voicemail services and any new messages on the backup server are synchronized with it. The backup voicemail server operates using the existing voicemail licenses held by the central system for normal operation.

- **Distributed Voicemail Servers**
  Multiple Voicemail Pro servers can be installed within a Small Community Network. These are referred to as distributed voicemail servers. Within the configuration of the systems in the network (other than the central system and its fallback if any), you can specify that the system uses a particular distributed voicemail server for its voicemail services. This requires the system to have licenses for voicemail operation and the voicemail features it requires. Multiple distributed servers can be supported and several systems can share the same distributed server, each using their own license set. The distributed server is used for all services apart from message collection and message waiting indication, those services are still performed by the central voicemail server. Messages recording is done by the distributed servers with the messages then being forwarded to the central voicemail server.

Voicemail Operation Features

- **Alarm Action Enhancements**
  The alarms provided by the voicemail server using the Alarm Set action and the Alarms 174 queue panel have been enhanced to allow alarm repetition and to require an dialed response to clear an alarm in order to prevent it repeating.

  - **Alarm Duration and Retries**
    The ring duration for an unanswered alarm call can be adjusted. In addition, a number of retries and the interval between retries can be specified for an unanswered alarm.

  - **Alarm Clearing**
    Normally the alarm and any repeats are cleared once the alarm call is answered. For Voicemail Pro 6.0, a cancel code of up to 4 digits can be specified and must be dialed to stop the alarm from using any further retries.

  - **Alarms Administrator**
    A new type of client account has been added. The Basic account user can only edit alarm settings shown in the Alarms 174 queue panel.

- **Administration of Mailbox User Settings for Outcalling and Personal Distribution Lists**
  User mailbox settings such as outcalling settings and personal distribution lists can be accessed and edited through the Voicemail Pro client.

- **TTS Prompt Generation**
  On voicemail servers licensed for text to speech (TTS), the prompts used for call flow actions can be generated using TTS. The text entered in the action's Description field is used as the script for the recording.

- **Voicemail Configuration Backup and Restore**
  The Voicemail Pro client can be used to configure daily, weekly and monthly automatic backups or to run an immediate manual backup. Each backup type can be individual configured for the types of files and settings it should include including messages. The client can also be used to restore the files from a previous backup.

- **Get Mail Action - Advanced Personal Options**
  For systems running in Intuity mode, a Get Mail action can be used in call flows to provide the user with access to a range of mailbox control actions. These actions become part of the mailbox telephone user interface. The options that become available are:

  - Voicemail on/off.
  - Voicemail email mode.
  - DND on/off.
  - Follow Me.
  - Forwarding.
  - Edit Voicemail.
  - Edit Callback Number.
  - Edit Mobile Twinning.

- Personal Options Menu Action
  The Play Configuration Menu action has been replaced by the Personal Options Menu action. This action can operate in one of two modes. The legacy mode

- Generic Action - Set Interdigit Delay
  The delay that the voicemail server allows between the dialing of digits in numbers (by default 5 seconds) can be adjusted for a call flow using a Generic action.

- Recording Auto Deletion
  System wide automatic deletion delays can now be specified for new and old recordings. These are separate settings from those used for new and old messages. In addition the playback order for recordings (first in-first out or last in-first out) can also be specified.

- Voicemail Server Shut Down and Suspend Controls
  The Voicemail Pro client can be used to shut down or suspend voicemail server operation. In either mode, voicemail is treated as no longer available by the system. Suspend mode can be canceled using the Voicemail Pro client, after which normal voicemail server operation is resumed. Shut down mode can only be canceled by restarting the voicemail service or the server PC. The shut down and suspend processes are polite processes, allowing existing calls to be completed while stopping new calls. However if required the shut down process can be turned from a polite shut down to an immediate shut down.

Application Support

This sections summarizes the support for applications by IP Office 7.0. Some specific features of applications may have additional requirements. Those requirements will be details in the appropriate application installation manual.

Windows Operating System Support

The following table gives a summary of the operating systems on which the applications that are part of the IP Office 7.0 have been tested and are supported. While the applications may function of other operating systems, they have not been tested by Avaya and are not supported.

Application	Windows Clients						Windows Servers		
	XP Pro		Vista		Windows 7		2003	2008	
	32	64	32	64	32	64	32	32	64
Voicemail Pro server	✓	–	✓	–	✓	✓	✓	✓	✓
... plus UMS	–	–	–	–	–	–	✓	✓	✓
... plus campaigns	–	–	–	–	–	–	✓	✓	✓
Voicemail Pro client	✓	✓	✓	✓	✓	✓	✓	✓	✓
ContactStore	–	–	✓	–	–	–	✓	✓	–
one-X Portal for IP Office	–	–	–	–	–	–	✓	✓	✓
Customer Call Reporter	–	–	–	–	–	–	✓	✓	✓
SoftConsole	✓	–	✓	✓	✓	✓	–	–	–
Manager	✓	✓	✓	✓	✓	✓	✓	✓	✓
System Monitor	✓	✓	✓	✓	✓	✓	✓	✓	✓
System Status Application	✓	✓	✓	✓	✓	✓	✓	✓	✓
TAPI - 1st Party	✓	✓	✓	✓	✓	✓	✓	✓	✓
TAPI - 3rd Party	✓	✓	✓	✓	✓	✓	✓	✓	✓
Phone Manager Lite/Pro	✓	–	✓	–	✓	✓	–	–	–
Phone Manager PC SoftPhone	✓	–	✓	–	✓	✓	–	–	–

- Vista support is only on Business, Enterprise and Ultimate versions.
- Windows 7 support is only on Professional, Enterprise and Ultimate versions.

Virtual Server Support

For IP Office 6.0, all applications supported on Windows server operating systems are supported while running on the following virtual servers:

- VMWare Server.
- Microsoft Virtual Server 2005 R2.
- Microsoft Server Hyper-V.

Browser Application Support

The following applications are accessed using web browsers. The table below details the browsers tested by Avaya.

Application	Windows				Mac
	Internet Explorer	Firefox	Opera	Safari	Safari
Voicemail Pro UMS	✓ 7+	✓ 3+	✓ 2+	✓ 3.2+	✓ 3.2+
one-X Portal for IP Office	✓ 7+	✓ 3+	✓ 2+	✓ 3.2+	✓ 3.2+
Customer Call Reporter	✓ 7+	✓ 3+	✓ 2+	✓ 3.2+	✓ 3.2+
ContactStore	✓ 7+	–	–	–	–

Microsoft Outlook Support

Where applications interact with Microsoft Outlook, for IP Office 6.0 the version of Outlook supported are Outlook 2003 and Outlook 2007.

16.3 What was New in Release 5.0

This section outlines the major new features and changes in IP Office 5. For detail of other releases see Appendix: History 960.

- IP500 Control Unit 951
- Licenses 951
- IP Telephony 952
- Centralized Directories and Call Logs 954
- 1600 Series Phones 955
- Voicemail (General) 957
- Embedded Voicemail 957
- Small Community Networking 953
- Appearance Buttons 955
- SIP Telephony 953
- Voicemail Pro 958
- Other Features 956

General

- Supported Control Units
  Release 5.0 is supported on IP500, IP412 and IP406 V2 control units. It is not supported on Small Office Edition control units or IP401, IP403 and IP406 V1 control units.

IP500 Control Unit

- Extension Capacity Increased to 384
  The maximum supported extension capacity of IP500 control units has been increased from 272 to 384 extensions. The maximum number of supported users remains 500.
- Conference Capacity Increased to 128
  The conference capacity of IP500 control units has been increased to 128 conference channels with a maximum of 64 parties in any particular conference. Unlike the IP412 control unit, this capacity is not split into two separate banks of 64 party capacity. Note that the capacity is reduced by the number of embedded voicemail channels in use.
- Voicemail Channels Capacity Increased to 40
  The IP500 can now be licensed for up to 40 voicemail channels.
- IP500 4-Port Expansion Card
  This IP500 base card can be used to provide the IP500 control unit with an additional 4 ports for connection to external expansion modules.
  - Installing the card provides the following changes to the IP500 system capacity:
    - Support for 12 expansion ports. These can be used for the same external expansion units as already supported by the IP500's integral expansion ports.
  - The card is only supported in slot 4 of the control unit. That is the right hand slot when facing the control unit.
  - The card does not support the addition of an IP500 trunk daughter card.
  - The card is supplied with four 2 metre yellow interconnect cables. These cables or the 1 metre blue interconnect cables supplied with each external expansion unit can be used for connection to the IP500 4-Port Expansion Card. No other cables must be used. The 2 metre yellow interconnect cables must not be used for the connection of external expansion modules to the expansion ports on the rear of the control units.

Licenses

The following changes have been made to license operation:

- IP500 Professional Licenses No Longer Needed
  For IP Office 5+, the *IP500 Upgrade Standard to Professional* license is no longer required. Features previously restricted if this license was not present and valid can now be used. That include support for the Voicemail Pro and Conferencing Center applications, support for external expansion modules and support for the Conference Meet me short code.
- IP500 Voice Networking License Change
  For IP Office 5+, *IP500 Voice Networking (Additional channels)* licenses can be used without needing an *IP500 Voice Networking (Base 4 channels)* license to be present first.
- Advanced Small Community Networking License No Longer Needed
  For IP Office 5+, the *Advanced Small Community Networking* license is no longer needed to enable advanced Small Community Network features. That includes support for remote hot desking and for distributed hunt groups.

one-X Portal for IP Office

- one-X Portal for IP Office is an application that allows users to control their phone via a web browser. It can be used to make and answer calls and to perform call related actions such as hold, transfer and conference. It also allows the user to access their voicemail messages, call log, personal and system directories.
  - one-X Portal for IP Office runs on a server connected to the system via the LAN.
  - Accessed is licensed per user and enabled by the Enable one-X Portal Services 264 option (User | User 264).
  - The application uses an new TSPI interface provided by the system and configured within the system's security settings.

IP Telephony

- SIP Extension Support
  The system can now support SIP extensions. Each SIP extension requires an *IP End Points* license in order to register with the system.
- H323 Phone File Generation
  When using the system memory card as the source for H323 phone files, the system is able to provide auto-generate appropriate settings files in response to phone request when the actual file is not present. This feature is supported for phones requesting any of the following files: *46xxupgrade.scr*, *16xxupgrade.txt*, *46xxsettings.txt*, 16xx language files and *ext_16xxdata.txt* files.
- Phone Firmware Upload
  Within Manager the Embedded File Management option can be used to view the files on a system memory card. For IP Office 5, an option Upload Phone Files is provided. When selected, all necessary binary files for DS, H323 and IP DECT are transferred from Manager to the embedded memory card.
- 1600 Series Phone User Settings Backup/Restore
  These phones can be configured to restore and backup user settings from an HTTP server. For IP Office 5+, the system memory card can be used for this backup and restore operation.
- Codec Selection
  A number of changes have been made for compression codec selection on IP lines and extensions.
  - By default, for all IP lines and extensions the Compression Mode is set to *Automatic Select*. That uses the following order of preference for codec negotiation during call setup: G729, G723, G711 U-Law, G711 A-Law. For IP Office5, the Automatic Codec Preference 160 setting (System | Telephony | Telephony 160) allows selection of which codec should come first in the order of preference for lines and extension set to *Automatic Select*.
  - On SIP/SES lines and SIP extensions, the codecs used and the order of preference can be adjusted.
- Phone Quality of Service (QoS) Monitoring 150
  The System Status already displays H323 trunk QoS information. For IP Office 5 it can also display QoS information for phones. For 4600, 5600 and 1600 Series H323 phones, the system can collect VoIP QoS data from the phones. For other phones, including non-IP phones, it can collect QoS data for calls that use a VCM channel.
  - The collection of phone QoS data is enabled by the Enable RTCP Monitor 150 setting (System | LAN1 | VoIP 150).
  - The QoS data collected by the system is displayed by the System Status Application.
  - QoS alarm thresholds can be configured for round trip delay, jitter and packet loss. If during a call, any of the thresholds is exceeded, an alarm is sent to the System Status Application. The alarm can also be sent to any of the System Events 172 outputs configured in the system configuration (SNMP, Syslog and SMTP Email).

SIP Telephony

- SIP Extensions
  The system now supports SIP extension devices. That includes both SIP phones and SIP devices such as conference servers. SIP Extension use the *IP End-Points* license.

- T38 Fax Relay 749
  T38 Fax Relay is a set of protocols that allow the transport of Group 3 faxes over IP connections. IP Office 5 can act as a T38 Fax Relay gateway. This is only supported for IP500 systems fitted with an IP500 VCM card.

  - T38 fax is supported on SIP trunks and SIP extensions. Each T38 fax call uses a VCM channel.

  - Within a Small Community Network, an T38 fax call can be converted to a call across across an H323 SCN lines using Fax Transport Support protocol. This conversion uses 2 VCM channels.

  - In order use T38 Fax connection, the Equipment Classification of an analog extension connected to a fax machine can be set *Fax Machine*. Additionally, a new short code feature Dial Fax is available.

- IP Office 5.0+: If the wildcard * is used in the SIP trunk's Local URI, Contact and Display fields, that SIP trunk will accept any incoming SIP call. The incoming call routing is still performed by the system incoming call routes based on matching the values received with the call or the URI's incoming group setting. Outgoing calls using the trunk will need to use another URI set against the trunk.

Small Community Networks (SCN)

The following new features are supported for Small Community Networks.

- Support for 32 Systems 810
  The maximum supported capacity for an SCN has been increased from 16 systems to 32. The maximum number of users within the network remains 500.

  - This option is only supported if all systems in the network are running IP Office 5 or higher.

  - Networks including any system running pre-IP Office 5 software are still restricted to a maximum of 16 systems.

- Support for Meshed Layouts 811
  Within a Small Community Network of IP Office 5+ systems, the H323 SCN lines can be connected in a meshed layout.

- Small Community Network Fallback 826
  A number of fallback features are supported. These are intended to maintain a minimal level of functionality while possible issues with the system are resolved. One system SCN Line within the local systems configuration can be set a its fallback trunk. The other system to which that trunk connects will then provide fallback support. The following features can then be selected for fallback support:

  - Secondary DHCP Gatekeeper for IP Extensions
    Avaya 1600, 4600 and 5600 Series phones registered with the system are re-registered with the other system during fallback.

  - Advertised Hunt Groups
    Hunt groups hosted by the system are advertised from the other system during fallback.

  - Voicemail Pro Server Support
    If the local system is hosting the Voicemail Pro server for centralized voicemail, the other system act as the voicemail server host during fallback.

Centralized Directories and Call Logs

- Centralized System Directory 741
  The system directory consists of both configuration records and temporary imported records. The following changes have been added for IP Office 5:
  - The capacity for directory records has been increased as listed in the table below.
  - In addition to the existing LDAP directory import, a system can be configured to automatically import directory records from another system using HTTP.
  - LDAP directory records are now included in the directory on phones. Previously LDAP records were only included in Phone Manager and SoftConsole listings.
  - A system phone user, using a 1600 Series phone, can edit a system's configuration records

System	Number of Directory Records			Total Number of Directory Records
	Configuration	LDAP Import	HTTP Import	
IP500 / IP500v2	2500	5000	5000	5000
IP412	2500	2500	2500	2500
IP406 V2	2500	2500	2500	2500

Centralized Personal Directory 744
Previously, when using a T3 phone, the user could view up to 100 personal directory entries that were stored for them within the system configuration. For IP Office 5 the following changes have been made to this personal directory:

  - The user can now view and use their personal directory when using a 1600 Series phone or T3 phone.
  - In addition, when using a 1600 Series phone, the user is able to edit their personal directory entries.
  - If the user host desks to another T3 or 1600 Series phone in the Small Community Network, their personal directory is accessible across the Small Community Network.

- Centralized Call Log 745
  Previously, individual phones have maintained the call log shown to users, ie. it is a log of calls handled by the phone rather than calls to the user. If the user then hot desked to another phone, the call log did not move with them. For IP Office 5, the call log shown on 1600 Series phones is a user call log stored by the system.
  - The user call log is updated even when the user is using a phone that cannot display it. The log contains records for the 30 most recent calls (10 on IP412 and IP406 V2 systems).
  - It includes calls while the user is logged off or set to do not disturb.
  - It includes calls made and answered by the user using a twinned devices, one-X Mobile client, using mobile call control and Phone Manager Telecommuter mode.
  - When the user hot desks to another 1600 Series phone, their call log becomes viewable on that phone.
  - When the user has hot desked to another system in the Small Community Network, their call log is still updated. If they are using a 1600 Series phone they can also still view and their call log.
  - The system also maintains a missed call log for hunt groups (a missed call being one not answered by any of the group members). This group log will contain records for the 10 most recent missed calls. The system can be configured to include selected hunt group logs within a users view of their personal call log.
  - The user's call log is also used for the Redial function.

1600 Series Phones

- **1603SW Phone**
  The new variant 1603SW phone is supported. This phone is similar to the existing 1630 but provides a PC port for connection of the user's PC through the phone.

- **1600 Series Phone Display and Menu**
  The menus used to access features on 1600 Series phones have been changed and expanded to allow access to wider set of user features. This set is similar to to those accessible by T3 phone users.

- **Phone Locking**
  1600 Series phone users can select a **Phone Locked** option through the phone's menu. This prevents the being used to make outgoing external calls. It also prevents changes to user settings. The phone can still be used to answer incoming calls and to make internal calls. The lock can only be removed using the user's login code.

- **Line Appearances** 690
  IP Office 5 allows line appearance buttons to be placed before call appearances. The user still requires at least one call appearance button. This configuration is only supported for users using 1600 Series phones. When configured in this way, idle line preference will select the first free line appearance before the first free call appearance.

Appearance Buttons

- **Bridged Appearance / Call Covering Appearance Ring** 278
  For each user, IP Office 5 allows the selection of either normal ringing, a single abbreviated ring or no ringing when a call alerts on any of their call coverage appearance and bridged appearance buttons. The choice is selected through the Coverage Ring 278 setting (User | Telephony | Multi-line Options 278). This is in addition to the existing per button settings for immediate, delayed or no ring.

- **Bridged Appearance / Call Coverage Appearance Lamp** 160
  On 1400 Series and 1600 Series phones and BM32 button modules, the button lamp flash type for external calls can be selected. This is done using the Visually Differentiate External Call 160 setting (System | Telephony | Telephony 160). When selected, a slow flash (500ms on / 20ms off) is used for external calls. When off and for internal calls the normal flash is used (500ms on / 500ms off).

- **Line Appearances** 690
  IP Office 5 allows line appearance buttons to be placed before call appearances. The user still requires at least one call appearance button. This configuration is only supported for users using 1600 Series phones. When configured in this way, idle line preference will select the first free line appearance before the first free call appearance.

Other Features

- Coverage Group 748
  A user can be configured to have unanswered external calls presented to a group of other users rather than to voicemail. When a Coverage Group 276 (User | Telephony | Supervisor Settings 276) is selected, unanswered external calls continue ringing at the users extension. After the user's no answer time expires, the call also alert at the phone's of the Coverage Group members. If the user is logged off or on DND, the coverage group is used immediately for external calls.

- CDR Support 910
  Output of CDR records is not supported by IP Office 5+.

- Forward to Voicemail 281
  A user's forward unconditional setting can be set to voicemail. This is done through the system configuration or the menu on 1600 Series phones. This option is not applied to hunt group calls and is cleared if the user disabled forward unconditional using short codes or an application.

- Drop External Only Impromptu Conference Control 160
  This option (on the System | Telephony | Telephony 160 tab) can be selected to have the system automatically end any conference when the last internal user exits. This stops users using conference as a method to transfer external callers back off switch.

- CSV User Import/Export 61
  The CSV import and export of user details now supports the full name of the user.

- Button Label Printing 133
  If a version of DESI label printing software is installed on the same PC as Manager, the option **Tools | Print Button Labels** becomes available. Manager can then be used to select and export user button details to the DESI software for printing.

- Dial Tone Transfer 751
  A user who is not able to make external calls (for example, due to **Outgoing Call Bar**), can call a user who is able to make external calls and ask to be transferred to dial tone. Similarly, this feature can be used by a user unable to dial specific numbers to request dial tone from a user who is able to dial such numbers. The transfer destination must be an ARS form that provides secondary dial tone.

- Improved Internal Twinning 293
  When using T3 or 1600 Series phones that are internally twinned, access to user settings and information from either extension has been improved. That includes status indication for those features and related functions such as message waiting indication.

  - Personal directory.
  - Group membership.
  - DND Exceptions list.
  - Call Log.
  - Forwarding.
  - Voicemail access.
  - Redial.
  - Voicemail On.
  - Do Not Disturb.

  - On 1600 Series phones twinned status is indicated by a T in the display.

- **Phone Manager Screen Popping**
  For IP Office 5 and higher, screen popping is only supported with Outlook 2003/2007.

- When you attempt to save a configuration that is too large, you will be prompted and the save is canceled.

- During normal operation, additional configuration entries can be added to the configuration without using Manager (for example call log entries and directory entries made from phones). If, during the overnight backup to flash memory 49, the configuration if found to be too large, entries will be removed until the configuration is sufficiently small to be backed up. The entries removed are call log records, system directory records and then personal directory in that order. Note that those entries will still exist in the configuration running the system in its RAM memory, however if the system is restarted they will disappear as the configuration is reloaded from the Flash memory.

Voicemail (General)

- **System Default Mailbox Reception Numbers**
  Breakout numbers allow callers to select another destination while listening to a user's mailbox greeting. Each user mailbox can have up to 3 possible breakout numbers set, accessed by the caller dialing 0, 2 or 3 (*0, *2, *3 on Embedded Voicemail).
  - IP Office 5+ allows system default numbers to be set for each mailbox breakout. These system defaults are then applicable to all user mailboxes unless overridden by a user's own breakout number settings.
  - IP Office 5+ also allows the breakout numbers to be set through **User Rights**.
  - For Voicemail Pro 5.0 with IP Office 5, the additional breakouts 2 and 3 are now also supported in IP Office mailbox mode as well as Intuity emulation mailbox mode.
- **Forward Unconditional to Voicemail**
  Within the system configuration for a user's forwarding settings, for **Forward Unconditional** the option *To Voicemail* can be selected. This will override any forwarding number set and send calls immediately to voicemail when forward unconditional is enabled. This option is supported with all system voicemail types including Voicemail Pro.
- **Voicemail Lite**
  The Voicemail Lite application is not supported with IP Office 5.

Embedded Voicemail

The following new feature has been added to embedded voicemail operation:

- **Auto Attendant Dial by Name Action** 403
  Embedded voicemail auto attendant menus can now include Dial by Name as an menu option. Callers can dial the name, or part of the name, of the user they require. Then, using the recorded name prompts of mailbox users, the matching names are listed to the caller to either make a selection or to enter further characters to improve the match. When a match is selected, the user is called.
  - Users without a recorded name prompt or set as Ex Directory are not included.
  - Each auto attendant can be configured to match names using either *Last then First* or *First then Last* order.
  - The mailbox prompt menu now includes an option (*05) that lets mailbox users record their name prompt.
  - The existing Record Message short code feature can be used to record mailbox name prompts.
- **Prompt Upload** 119
  Using the embedded file management option, the Manager application can now automatically load the prompts onto the Embedded Voicemail Card including any new prompts required for IP Office 5 features.

Voicemail Pro 5.0

- **ContactStore 7.8**
  The ContactStore software has been updated. The new version uses a different database format and supports a wider range of search options. The method of interaction with the voicemail server and the system has not changed. However, ContactStore 7.8 has not been tested with pre-5.0 versions of Voicemail Pro and IP Office. For full details refer to the IP Office ContactStore Installation manual.

- **Updated Text To Speech TTS (ScanSoft) Prompts and Software**
  The TTS (ScanSoft) software include with Voicemail Pro 5.0 has been updated. The change allows:

  - **Additional Languages**
    Additional language support for Chinese (Mandarin), Danish, Finnish, French Canadian, Greek, Hungarian, Polish, Portuguese and Swedish. The set of TTS languages now matches the recorded prompt languages provided by Voicemail Pro with the exception of Hungarian.

  - **Vista Support**
    The updated TTS drivers are supported on Vista and on 64-bit versions of supported Windows operating systems.

- **IP500 Voicemail Pro Ports**
  For IP500 control units, the maximum number of licensable voicemail channels has increased to 40.

- **Small Community Network Fallback**
  Within a Small Community Network, voicemail server is associated with a central system via which it provides voicemail services to all the systems in the SCN. IP Office 5 provides a number of fallback settings, including allowing the voicemail server to work with fallback system if the central system is removed from the network for any reason. This feature is configured within the system configurations and requires the fallback system to have the appropriate licenses for the voicemail features required during fallback.

UMS Enhancements

The following additions have been made to UMS operation:

- **Hunt Group Mailbox Support**
  Hunt group mailboxes are now supported. A UMS Web Service option is available on the **Hunt Group | Voicemail** tab within the system configuration. The use of this option allows access to the hunt group mailbox messages using IMAP or a web browser. This consumes *UMS Web Service* licenses in the same way as enabling users for UMS.

- **Additional Web Browser Playback Options**
  During message playback through the UMS web browser interface, options for previous message, next message, first message and last message are now available to the user. In addition if playing back using an extension an option to callback the caller is available if the CLI is known. Also the forward message option now displays a list from which selection of the forward destination or destinations can be made.

- **UMS Exchange 2007**
  A user or group can be configured to have their voicemail messages forwarded to the inbox of an Exchange 2007 email account. Telephone, including Visual Voice, mailbox access is redirected to that email inbox as the store for voicemail messages. Alternatively the user can access their voicemail messages using Outlook 2007 or any other mechanisms supported by Exchange 2007. Voicemail messages in an Exchange 2007 inbox are not visible to UMS IMAP and UMS Web Voicemail, however Exchange 2007 provides its own methods for IMAP and web browsing of Exchange mailboxes.

Call Flow Actions

The following changes have been made to Voicemail Pro actions.

- **Whisper Action**
  Two new options are now provided by the Whisper action. The action can now be used requiring a caller recording. The transfer target is still able to accept or reject the call but will do this based on the displayed text and the prompts pre-recorded with the action when it was setup. Whisper calls can also be used with auto accept. When selected, after hearing the caller's recording and the action prompts the call is automatically connected to the transfer target.

- **Alarm Set**
  This action was previously restricted to setting up a single non-repeated alarm back to an internal caller's own extension. The action has now been enhanced to allow the setup of alarms to other extension and repeating alarms.

- **Clock Action**
  The clock action can now be used to say the time just once before moving to the next call flow action or to repeat the time until the caller presses a DTMF key or hangs up.

- Increment and Test Counter / Decrement and Test Counter
  These two new actions have been added to the list of Condition actions. They can be used to change the value of one of the 15 new *$COUNTER* call variables and then branch the call flow if the new value matches a specified target value.

- Transfer / Assisted Transfer
  These actions now include an option to change the caller's priority prior to the transfer.

- Generic Action
  The Specific tab settings of this action have been changed to allow the entry of generic commands by the selection from a list of commands and then completion of relevant parameters. The resulting text string for the resulting generic command can still be displayed and edited if required or if a generic command not included in the parameterized command list is being used.

Call and User Defined Variables

The following changes have been made to the call and user defined variables provided by the voicemail server.

- $COUNTER
  A set of $COUNTER call variables, $COUNTER1 to $COUNTER15 have been added. The value of these can be set, incremented and decremented using Generic actions and the specific Increment and Test Counter, Decrement and Test Counter actions.

- User Defined Variable Display
  The current values of all the user defined variables can be displayed and edited through using the Voicemail Pro client. This is done by selecting Server Queues and then User Variables in the left hand navigation pane.

Voicemail Pro Client Changes

- Minimum Message Length
  Through the voicemail server's general preferences, the minimum message length saved by the voicemail server can be seen and changed. The value can be set between 0 and 10 seconds.

- Navigation Changes
  A number of changes have been made to the items that can be selected in the left hand navigation pane of the Voicemail Pro client.

  - Users / Groups
    Selecting Users or Groups in the left hand navigation pane displays details for each mailbox. Voicemail Pro 5.0 provides the following additional options when using this display.

    - Disable Mailbox
      By right clicking on a listed mailbox and selecting Disable Mailbox, the use of the mailbox can be disabled. Attempts to connect to the mailbox will receive number unobtainable indication from the voicemail server.

    - Clear Mailbox
      By right-clicking on the listed mailbox and selecting Clear Mailbox, all existing messages and prompts in the mailbox are deleted.

  - Server Queues
    The option Server Queues in the navigation pane gives access to the following information.

    - Alarms
      When selected, outgoing alarms calls set using Alarm Set actions are listed in the Voicemail Pro client's right hand pane. The list can also be used to add, delete alarms and to edit alarm settings.

    - User Variables
      When selected, the current values of user defined variables are listed in the Voicemail Pro client's right hand pane. The list can be used to add, delete user defined variables and to edit the current value of those variables.

    - Outcalls
      When selected, outgoing calls being made or scheduled to be made by the voicemail server are listed in the Voicemail Pro client's right hand pane. The list can be used to delete calls.

  - Conditions Import/Export
    Existing conditions can now be exported to a file and then imported into the configuration of another Voicemail Pro 5.0+ system.

16.4 What was New in 4.2

This section summarizes the main changes in IP Office 4.2.

4.2 Q4 2008 Maintenance Release

- Handsfree Announced Transfer 784
  The IP Office now supports a number of methods by which the enquiry call part of a supervised transfer can be made as a call which is auto-answered call.

- HTTP-TFTP Relay Using Manager 144
  For IP phones making HTTP files requests for phone firmware files, HTTP-TFTP Relay is support using Manager as the TFTP server. This is done by setting the TFTP Server IP Address to the address of the Manager PC and the HTTP Server IP Address to the control unit IP address. This method is supported for up to 5 phones. This is in addition to using the Embedded Voicemail memory card to support up to 50 IP phones.

Manager Changes

- Embedded File Management 119
  For systems with a memory card installed, the contents of the card can be viewed through Manager. This mode is accessed through the File | Advanced | Embedded File Management 119 option. This view can also be used to add and remove files from the card. This may be useful when the memory card is being used to store music on hold files and or phone firmware files.

  - Embedded Voicemail message and mailbox greeting files in *.clp* format are protected and cannot be deleted or downloaded. Embedded Voicemail prompts files in *.c11* or *.c23* formats can be managed through this interface.

- Improved Backup File Control 110
  Previously if set to keep up backup copies of configuration, the number of backup kept was not controllable. The IP Office 4.2 Manager now allows the number of backups kept by Manager for each system to be limited. This is controlled by the Number of Backup files to keep control within File | Preferences | Security 110.

- Discovery of Unsupported Systems 55
  Previously the Manager discovery menu did not display systems with core software that it did not support. The discovery menu will now show those units but indicate that they are not supported.

- SMTP Server Settings 177
  The SMTP server settings have been moved from the System | System Events 172 tab to a separate System | SMTP 177 tab. This reflects the usage of SMTP sending for functions other than just system event notification. For IP Office 4.2 that includes Voicemail Email support for Embedded Voicemail.

- IP Office Discovery using DNS 108
  For customers who have configured their systems as entries on a DNS server, Manager can no be set to use DNS to locate the required system. The use of DNS is configured through Files | Preferences | Discovery 108. When selected the Unit/Discovery Address field on the Select IP Office 55 dialogue is replaced by a Enter Unit DNS Name or IP Address field.

Button Programming

- Set Night Service Group 657 / Set Out of Service Group 658
  Programmable buttons can be configured to change the Night Service Group and Out of Service Group targets of the hunt group associated with the button.

Voicemail

- Visual Voice through MESSAGES Button
  The System | Voicemail 157 option Messages button goes to Visual Voice allows the default spoken prompts to be replaced by the Visual Voice menus on phones able to support Visual Voice.

Voicemail Pro

- IP Office Unified Messaging Service (UMS)
  Two new methods for users to access their mailbox are now supported. The methods are mailbox access using email applications that support IMAP (Internet Message Access Protocol) and mailbox access via web browsers. The use of these requires the IP Office configuration to contain *UMS Web Services* licenses for the number of required users.

  - IMAP Service
    The Voicemail Pro now includes an IMAP server. Users can then access their voicemail messages using email clients such as Outlook and Lotus notes that support an IMAP Client. When connected, the IMAP client and Voicemail Pro will synchronize messages in the mailbox with message files in the IMAP client. Playback is through the sound facilities of the user PC.

  - Web Voicemail Service
    The Voicemail Pro web service allows users to access their mailbox using a web browser. This has been tested with the Internet Explorer 7, Firefox 2 and Opera 9.10 PC browsers. This method of access requires Voicemail Pro to be installed on a PC already running Microsoft's IIS web server. Users can select to have message playback via an IP Office extension or through their PC's sound facilities.

- Test Variable Action
  The existing Check Digits action has been replaced by a Test Variable action. This, in addition to offering the capabilities of the Check Digits action, provides significant enhancements. In addition to being able to match the user's DTMF input against a specified string offered by the Check Digits action, the Test Variable action allows the testing of the contents of any system variable against known user extensions, hunt-groups, mailboxes and the contents of another variable. This will allow callers to enter numbers via a menu action, that can be verified as matching an existing extension or hunt groups prior to attempting to carry out transfers to otherwise potentially non-existent numbers.

- Menu Action Invalid Input Handling
  The Menu action has been enhanced. It now includes a control for the number of retries for the caller to make a valid entry and an *Invalid Input* result for connection to following call flow actions. Also prompts can be selected for playback whenever an invalid entry or entry timeout occurs.

- License and Service Status Display
  When the Voicemail Pro client is connected to a Voicemail Pro server, the Help | About screen displays a list of the licenses being used by the Voicemail Pro server. This license details include the validation status and capacity of those licenses. The status of related services, for example the UMS IMAP server, are also listed.

- System Variable Length Increase
  Previously the length of values stored by system variables has been limited to 64 characters. That maximum length has been increased to 512 characters.

- Outcalling Configuration
  In conjunction with Phone Manager 4.2, Voicemail Pro 4.2 allows users to adjust their outcalling settings through using a visual menu within Phone Manager.

  - Using the Phone Manager interface, users can now apply a delay between each notification call in an escalation list.

Embedded Voicemail

- Cantonese Prompt Support
  Cantonese Chinese prompts are now supported. They will be used for the system or user Locale set to *Hong Kong (Cantonese)*.

- Voicemail Email Support
  The Voicemail Email options are now useable with Embedded Voicemail. This requires an accessible SMTP server to be configured in the System | SMTP 177 tab. Not supported on the Small Office Edition.

Mobility Features

The user Twinning options have been relabeled Mobility to reflect the addition of a number of features for Mobile Workers. In addition the Mobile Twinning license has been relabeled Mobility Features. The additional features are:

- Mobile Call Control 730
  IP500 systems with digital trunks (PRI, BRI or SIP) can support users configured for Mobile Call Control. When those users receive a twinned call from the IP Office, they can dial ** to place that call on hold and access IP Office dial tone. They can then dial as if an extension on the IP Office. The IP Officecan also be configured to allow incoming calls from the user's mobile twin device to receive IP Office dial tone (also know as DISA (Direct Inward System Access)).

- one-X Mobile Client Support 734
  IP Office 4.2 supports devices using the Avaya one-X Mobile Client software (Windows Mobile V5 and V6 and Symbian Single-Mode only) as the destinations for mobile twinning. An IP Office specific set of one-X Mobile functions are supported using DID's routed to special IP Office short codes. This allows one-X Mobile Client users to invoke various IP Office functions. For example picking up a call ringing at their primary extension or on calls received from the IP Office, transferring the call to their voicemail mailbox.

- Logged Off Mobile Twinning 293
Previously mobile twinning was not used when the user logged out the IP Office. IP Office 4.2 allows selection of whether calls should still alert the user's twinned device when the user is logged out. This selection is per individual user. When used, the user's Mobile Dial Delay setting is ignored when logged out.

- Longest Idle Status for Twinned Users
The longest idle information is now updated for calls to the internal or mobile twin of a hunt group member. The member is seen as busy to hunt group calls if either the master or the twin is on a call. In addition, for mobile twinning:

  - If mobile twinning of hunt group calls is selected and the master is idle, the longest waiting state will reflect that of the mobile twin.

  - The above also applies if the master is logged out and logged out twinning is selected.

- User BLF Indication Changes
Changes have been made to the BLF status applied to users. For most user, logged in or associated with a specific extension, there will be no visible change.

  - Users set to DND are now indicated as busy on BLF indicators. This applies to BLF indication only and not to how calls are routing which still follow the previous operation for DND on.

  - For users who are logged out there are a number of changes:

    - The status shown for a logged out user without Logged Off Mobile Twinning depends on whether they have Forward Unconditional enabled. If they have Forward Unconditional enabled, the user is shown as idle. If they do not have Forward Unconditional enabled they will show as if on DND.

    - The status shown for a logged out user with Logged Off Mobile Twinning will be as follows:

      - If there are any calls alerting or in progress through the IP Office to the twin the user status is shown as alerting or in-use as appropriate. This includes the user showing as busy/in-use if they have such a call on hold and they have Busy on Held enabled.

      - If the user enables DND through Mobile Call Control or one-X Mobile client their status will show as DND.

      - Calls from the IP Office direct to the users twinned destination rather than directed by twinning from their primary extension will not change the user's status.

- License Change
The Mobile Twinning license is now called the Mobility Features license, reflecting the fact that it can be used for Mobile Call Control and one-X Mobile Client support in addition to mobile twinning. The mode of license operation has also changed. Prior to IP Office 4.2, the license was only consumed by users who had Mobile Twinning enabled. For IP Office4.2 it is consumed by a user when they are configured for any of the mobility features, including mobile twinning even if they have turned mobile twinning off.

Music on Hold

The following changes have been made to music on hold (MoH) support.

- 3 Additional Internal Sources
Previously the IP Office has supported a single music on hold source which could be either an internal source (holdmusic.wav file), an external source connection or a default tone (IP Office 4.0+). Except for Small Office Edition, IP Office 4.2 systems can be configured with up to 3 additional internal hold music sources. These are loaded and stored in the same way as the standard *holdmusic.wav* file plus the IP Office will attempt to load new sources specified in a configuration merge.

- IP500 Increased Internal Hold Music Length
The standard length of the holdmusic.wav file is 30 seconds. For the IP500 control unit, the length for all internal wav files for music on hold can be up to 90 seconds.

- Incoming Call Route Hold Music Selection
On systems with multiple music on hold sources, the source to associate with external calls can be specified by the Incoming Call Route 344 applied to the call.

- Hunt Group Hold Music Selection
On systems with multiple music on hold sources, the source to associate with calls received by a hunt group can be specified within the hunt group 306 configuration.

  - Calls overflowing from a hunt group will use the hold music source setting of the original hunt group and ignore the setting of the overflow group.

  - Calls going to night service or out of service fallback group use the hold music source setting of the original hunt group and then, if different, the setting of the fallback group. The setting of further fallback groups from the first are ignored.

- Small Community Network Support
The 4 possible music on hold sources are numbered 1 to 4, with 1 being the IP Office System Source. Where the source number to associate with a call is changed by an Incoming Call Route or Hunt Group, that number is retained when the call is rerouted across the SCN. If the receiving switch has a matching numbered alternate source that source is used, otherwise it default to the default system source.

- **Merging Music on Hold Changes**
  Where possible, when an internal source is specified or details of an existing internal source are changed, the configuration changes can be merged and IP Office will attempt to download any newly specified file at that time. Once downloaded, any existing file is replaced even if callers are listening to that source.

- **System Status Reporting**
  The status of the up to 4 internal sources is reported in SSA (*Waiting to Load*, *Loading*, *Loaded* and *Failed to Load*). An alarm is also generated if the IP Office is unable to load a specified source.

Phone Manager

The following new features are supported for Phone Manager 4.2 users in conjunction the IP Office 4.2.

- **Change Password**
  Users can now change their Password 264 (User | User | Password) 264 using Phone Manager. This is the password used as part of the log in to IP Office for the Phone Manager, SoftConsole and IP Office TAPI applications.

- **Outcalling Configuration**
  Outcalling was added as part of Voicemail Pro 4.1. It allows users of Intuity emulation mode mailboxes to configure a escalating series of telephone numbers at which the Voicemail Pro would attempt to alert them about new messages. For IP Office 4.2, configuration of Outcalling can be done through Phone Manager Pro. This allows the user to view their current outcalling settings and to make changes when necessary.

  - Using the Phone Manager interface, users can now apply a delay between each notification call in an escalation list.

- **Forward Messages**
  When accessing a voicemail mailbox using Phone Manager Pro, the user can now right-click on a message and select to forward it to another mailbox.

- **Version Display**
  The **About Phone Manager** menu (**Help | About Phone Manager**) now includes version information for the IP Office phone system and the Voicemail Pro server. This is information that may aid in the resolution of support issues.

- **Phone Manager Pro PC Softphone**
  The software element used for the PC softphone updated with the following changes:

  - Phone Manager Pro PC Softphone 4.2+ is supported on Vista operating systems (excluding the Home editions).

  - The VoIP Settings tab has been removed from the Phone Manager preferences. All codec and similar options are controlled through the IP Office configuration.

System Status Application (SSA)

The following new features have been added to the System Status Application as part of IP Office 4.2+:

- **Feature Key and License Information**
  All licenses and their current status is listed on the SSA's *Resources | Licenses* screen. It includes details of how often the license limit has been reached and the last time that occurred. The type and serial number of the Feature key dongle is also shown.

- **Control Unit Audit Trail**
  The Audit Trail information held by the IP Office and previously visible through Manager is now also shown in SSA.

- **IP Routes**
  Details about IP routes are now listed.

- **Voicemail Summary and Details Screens**
  A voicemail status screen is now available. It details the voicemail server type and other information such as the number of ports and port in use. In addition a mailbox status screen is available detailing the basic voicemail settings of each mailbox and the number of new, read and saved messages.

- **Directory**
  The SSA can now display a directory of all the extensions, users and hunt groups known to the IP Office. This includes users and hunt groups known through Small Community Network connections. In addition any conflicting Small Community Network numbers are indicated.

- **IP Phone Control**
  SSA can be used to reboot specific IP phones. This allows control over which phones are upgrading.

1600 Series IP Phones

The IP Office now supports the Avaya 1603, 1608 and 1616 phones.

  - **1603**
    This is the basic model. It provides only 3 programmable feature keys. Supports power from a power supply unit or PoE, however to use a 1603 with PoE requires an additional PoE power adaptor.

* **1608**
  Provides 8 programmable feature keys with twin LED's on each button. Supports power from a power supply unit or PoE.
* **1616**
  Provides 16 programmable feature keys with twin LED's on each button. Supports power from a power supply unit or PoE.
* **BM32**
  Usable with the 1616 only, the BM32 provides an additional 32 programmable feature keys with twin LED's. Up to 1 BM32 can be daisy-chained to the same 1616 phone. Each BM32 draws power from its associated 1616 phone and requires the phone to be using a dedicated power supply rather than PoE. A maximum of 16 BM32's are supported on the whole system. Note that appearance buttons are not supported on BM32's.

VoIP Phone Support Enhancements

In parallel with the addition of 1600 Series IP phones, a number of improvements have been made to the IP Office's support for IP phones.

* **Avaya IP Phone HTTP Server Support**
  For Avaya IP phones using IP Office DHCP, the address of the HTTP server from which those phones should download their software and settings files can now be specified in the IP Office configuration. 4600 Series and 5600 Series phones attempt to load files via HTTPS and then HTTP before falling back to TFTP. 1600 Series IP phones only support HTTPS or HTTP.
  * For IP Office 4.2, using the Embedded Voicemail memory card is also supported for HTTP file requests for up to 50 IP phones. This is done by setting the TFTP Server **IP Address** and HTTP Server **IP Address** to match the control units IP address. This is supported for up to 50 IP phones. This feature is called HTTP-TFTP Relay. HTTP-TFTP Relay is not supported using Manager or other TFTP servers.
* **Secondary Site Specific Options Number**
  A Site Specific Option Number (SSON) is used by Avaya IP phones when requesting phone specific settings from a DHCP server. When the IP Office is acting as the DHCP server, the matching number must be set in the IP Office configuration. IP Office 4.2 now provides two fields for settings SSON numbers in order to support Avaya 4600 and 5600 Series IP Phones (which use a default SSON of 176) and Avaya 1600 Series phones (which use a default SSON of 242).
* **IP Phone Restart using System Status Application**
  Individual Avaya IP phones or groups of phones can be selected and then restarted remotely using the System Status Application. This allows individual phones or groups of phones (up to 20 at a time) to be restarted in order to upgrade their firmware.
* **IP500 DHCP Enhancements**
  The scope of DHCP support on IP500 has been enhanced in a number of areas.
  * **Full Avaya IP Phone Support**
    Previous only a maximum of 5 IP phones have been supported if using the IP Office for DHCP and TFTP functions. An external DHCP server is required to support more than 5 Avaya IP Phones. For IP Office 4.2+, the IP500 supports up to 272 Avaya IP phones, the maximum extension capacity of the IP500 control unit.
  * **Multiple DHCP IP Address Pools**
    On each IP Office LAN interface, up to 8 DHCP address ranges (called 'pools') can be specified. These pools do not have to be on the same subnet as the IP Office itself. This allows devices being supported by IP Office DHCP to be given addresses on a different subnet than the IP Office.
  * **DHCP for Avaya IP Phones Only**
    The DHCP pools provided by the IP Office can be restricted for use by Avaya IP phones only. The IP Office will then not respond to DHCP request from other devices.
* **IP500 VCM Controls**
  For IP Office 4.2+, the VCM controls for echo and comfort noise supported in the IP Office configuration (System | VCM 181) are now also applied to IP500 VCM cards.

Trunk Support Changes

* **IP500 BRI Trunks So/To Control** 193 *(IP500 BRI Daughter cards only)*
  IP500 BRI daughter cards can be configured for So (S-Bus) operation for connection to ISDN terminal devices. Note that this requires the addition of terminating resistors at both the IP Office and remote end, and the use of a suitable ISDN cross-over cable. For full details refer to the IP Office Installation manual.
* **IP500 Analog Trunk Impedance Matching** 189 *(IP500 Analog Trunk Daughter cards only)*
  For systems set to the *United States* or *French Canadian* locale, the number of trunk impedance settings supported has been increased. In addition, Manager can now be used to perform line testing that will attempt to detect and display the recommended impedance setting. A In addition, options for improved signalling of long lines (**Quiet Line**) and mains hum filtering (**Mains Hum Filter**) have been added.

- **Increased ETSI CHI Trunk Support**
  Lines that support ETSI CHI (Channel Allocation ID) allow the IP Office to indicate to the central office exchange which trunk B-channel should be used for a call. IP Office 4.2 extends ETSI CHI support to all trunks that support ETSI and extends the number of features that can be used on ETSI CHI trunks.
  - E1 PRI and US PRI trunks can now select ETSI CHI operation.
  - The default Channel Allocation order to use for outgoing calls can be specified.
  - Previously only Line Appearance was configurable for each channel. For IP Office 4.2, the Incoming Group, Outgoing Group, Admin, Direction and Bearer can be set for each individual channel.
- **Network Interconnection Control (Trunk Cross-Connect Barring)** 782
  For the system, Restrict Network Interconnect 160 (System | Telephony | Telephony 160) can be enabled When enabled, each trunk is provided with a **Network Type** option that can be configured as either *Public* or *Private*. The IP Office will not allow calls on a Public trunk to be connected to a Private trunk and vice versa, returning number unobtainable indication instead. This features is designed for markets that restrict certain trunk to trunk connections between private and public networks.
  - Due to the nature of this feature, its use is not recommended on systems also using any of the following other system features: Small Community Networks 810, VPNremote, Phone Manager telecommuter mode.
- **SIP Trunk Enhancements** 233
  The following changes have been made for SIP trunks offered by the IP Office:
  - **SIP Trunk ARS Routing Enhancement** 233
    Previously, when used within an ARS form, alternate call routes were only used when no further channels were available on the SIP trunk. The IP Office now supports a **Call Initiation Timeout** on SIP lines which determines how long the IP Office should wait for a response to a SIP packet sent to initiate a new call before failing to an alternate route. The default timeout is 4 seconds and can be configured between 1 and 99 seconds.
  - **Use Offerer's Codec** 233
    When negotiating the codec to be used for a call, the call answerers preferences are normally used. For some SIP providers this needs to be changed and priority given to the offerers preferences. The **Use Offerer's Codec** option on the SIP line can be used to enable this option.
- **IP Trunk Enhancement** 223
  The timeout used to determine when an IP line waiting for a response should use an alternate ARS route was previously set by the *H323SetupTimerNoLCR* option in the NoUser user **Source Numbers**. This option is now set for each IP line through a **Call Initiation Timeout** option on the IP lines VoIP Settings tab. The default timeout is 4 seconds and can be configured between 1 and 99 seconds.

Hunt Group Operation

- **Call Priority Enhancements**
  Call priority is used by hunt group queues, with calls in the queue being sorted by priority and then longest waiting time. Previously the priority setting applied to a call was set by the **Incoming Call Route** that routed the call onto the IP Office or for all other calls was *1-Low*. IP Office 4.2 allows call priority to be changed in a number of ways.
  - **Call Priority Promotion Time** 160
    A system wide call promotion time can be defined (System | Telephony | Telephony | Call Priority Promotion Time 160). By default this feature is disabled. When enabled, calls queued for a hunt group have their priority increased by 1 each time they have been queued longer than the promotion time value until they have reached the maximum priority of 3.
  - **Call Priority Short Code Character** 427
    The character p can be used in the **Telephone Number** field of *Dial Extn* short codes to change the priority of a call. It is added as a suffix to the Telephone Number using the format *p(x)* where *x* is *1*, *2* or *3* for low, medium or high priority.
- **Hunt Group Music on Hold Selection**
  With IP Office 4.2 now supporting up to 4 music on hold sources, the source to use for calls presented to a hunt group can now be specified to override the system or incoming call route setting. See Music on Hold 962 above.
- **Longest Idle Status for Twinned Users**
  The longest idle information is now updated for calls to the internal or mobile twin of a hunt group member. The member is seen as busy to hunt group calls if either the master or the twin is on a call. In addition, for mobile twinning:
  - If mobile twinning of hunt group calls is selected and the master is idle, the longest waiting state will reflect that of the mobile twin.
  - The above also applies if the master is logged out and logged out twinning is selected.
- **Hunt Group Queued Call Presentation** 316
  For hunt groups where multiple calls are waiting to be answered, two methods of setting which calls the hunt group member answers when they go off-hook. This is controlled by the **Queue Type** option on the **Hunt Group | Queuing** tab.

- Assign Call On Agent Answer
  In this mode the call answered by the hunt group member will always be the longest waiting call of the highest priority. At the moment of answering that may not necessarily be the same call as was shown by the call details at the start of ringing. This is the default mode for IP Office 4.0+.

- Assign Call on Agent Alert
  In this mode, once a call has been presented to a hunt group member, that is the call they will answer if they go off hook. This is similar to the method used for IP Office 3.2 and earlier. This mode should be used when calls are being presented to applications which use the call details such as a fax server, CTI or TAPI.

- Overflow Mode Selection
  Pre and post IP Office 4.0 system used different overflow methods. For pre-IP Office 4.0 system, whether a call should overflow was determined on a call by call basis. For IP Office 4.0+ system, once one call had met the criteria to overflow, all currently queued calls overflowed. For IP Office 4.2 systems the mode of overflow operation can be selected through the hunt group **Overflow Mode** setting.

Call Presentation/User Information Display

- Called Number Display overrides Dialed Number Display
  For IP Office 4.0+, the digits displayed to users after making an outgoing call was changed to show the digits dialled to line by the IP Office rather than the digits dialed by the user. IP Office 4.2 returns to displaying the digits dialed by the user. When making a call via a directory feature the digits dialed from the directory are shown. The user dialed digits are retained with the call when parked, held, transferred and retrieved. Directory matching is still performed on the number dialed.

- User BLF Change
  Changes have been made to the BLF status applied to users. For most user, logged in or associated with a specific extension, there will be no visible change. However for users who have logged out there are a number of changes.

  - The status shown for a logged out user without mobile twinning will depend on whether they have **Forward Unconditional** enabled. If they have **Forward Unconditional** enabled the user is shown as idle. If they do not have **Forward Unconditional** enabled they will show as if on DND.

  - The status shown for a logged out user with mobile twinning will be as follows:

    - If there are any calls alerting or in progress through the IP Office to the twinned destination, the user status is shown as alerting or in-use as appropriate. This includes the user showing as busy/in-use if they have such a call on hold and they have **Busy on Held** enabled.

    - If the user enables DND through Mobile Call Control or one-X Mobile client, their status will show as DND.

    - Calls from the IP Office direct to the user's twinned destination number rather than redirected by twinning will not change the user's status.

- Longest Idle Status for Twinned Users
  The longest idle information is now updated for calls to the internal or mobile twin of a hunt group member. The member is seen as busy to hunt group calls if either the master or the twin is on a call. In addition, for mobile twinning:

  - If mobile twinning of hunt group calls is selected and the master is idle, the longest waiting state will reflect that of the mobile twin.

  - The above also applies if the master is logged out and logged out twinning is selected.

SMDR

- SMDR via IP Direct from the IP Office 898
  The control unit can be configured to send its SMDR call log records via IP to a designated address and port. The controls for this are on the System | CDR/SMDR 179 form.

- Increased SMDR/CDR Buffering
  For systems other than the Small Office Edition, the maximum of records that the system can buffer if the remote connect is lost has been increased from 1500 to 3000.

T3 Phone Enhancements

The following changes have been made to the range of features supported by T3 phones (all types) supported on IP Office.

- Line Appearances 695
  The use of line appearance buttons is now supported with T3 phones. Other appearance button types are not supported. Unlike other phones that support line appearances, the T3 phones do not need to have call appearance buttons programmed.

- Self-Administered Button Programming 512
  T3 phone users can be given access to a menu to program their own phone buttons (**Menu -> Settings -> Button programming**). This feature is not available by default, the user must be configured for this feature.

- Voicemail Settings Menu
  On systems with a voicemail server installed, T3 phone users can now access a menu of voicemail settings via *Menu -> Settings -> Voicemail Settings*. The options available depend on whether the voicemail server supports Visual Voice (Voicemail Pro and Embedded Voicemail).
  - On systems that do not support Visual Voice, the menu provides options for turning the user's voicemail on/off and for listening to their messages via traditional audio prompts.
  - On system that support Visual Voice, then menu provides options for turning the user's voicemail on/off, listening to messages through menu of visual options (Listen) and the other Visual Voice menu options (Message, Greeting, PIN, Email and hunt group mailboxes).
- Visual Voice Support
  In addition to access to Visual Voice through the menu above, buttons set to access Visual Voice go direct to the Visual Voice Listen function.
- Display of "Follow Me Here"
  When another user sets a follow me to the T3 user's phone, Follow me here and the user using the follow me are displayed on the T3 phone's status display. The item can be selected and used to cancel the follow me.
- Hunt Group Membership Display
  For selected hunt groups of which the user is a member, the user can be configured to display and change the status of their membership. The hunt groups are selected through the Manager (User | Hunt Group Memberships | Change Hunt Group Membership Status 292). Previously that list of hunt groups was accessed on the T3 phone using *Menu -> Settings -> Display/Audio ->Group Membership*. For IP Office 4.2 that menu option has been removed, instead the selected hunt groups are displayed on the phone's status display and the status can be changed through that display.
- Change Hunt Group Status
  For selected hunt groups on the system the user can be configured to be able to display and change the in service, out of service and night service status of the group. The hunt groups are selected through the Manager (User | Hunt Group Memberships | Change Hunt Group Membership Status 292). The user can then change the status from their phone using the *Menu -> Group state* option. For IP Office 4.2 that option now includes options to set all the displayed groups in service, out of service or into night service. Previous this had to be done group by group for each displayed group.
- System Name and Version Display
  Following a restart of the system or the T3 phone, the phone will briefly display the system software level and system name. This information can also be accessed using *Menu -> Settings -> System Info*.

Other Features

- Logged Off Users
  The following changes have been made in regard to logging out:
  - For IP Office 4.0 and 4.1, users without a configured Login Code are not allowed to log out. For 4.2+, these users are allowed to log out if they are currently associated with (logged in at) an extension that has a different Base Extension Number from their own Extension Number, so long as they are not set to Forced Login. This allows system maintainers to change and merge changed Base Extension Numbers. Affected users can then log out and will automatically be logged back in at the appropriate extension if available.
  - Mobile Twinning Users 293
    Mobile twinning users can be configured to still receive calls for their primary extension on their twinned device even when logged out from their primary extension. This is done using the Twin When Logged Off 293 (User | Mobility) 293 option.
- Static NAT Options in IP Office Firewalls 366
  Static NAT entries can be added to IP Office Firewall Profiles in order to translate between external and internal IP addresses. When static NAT entries are added to a Firewall Profile, only traffic that matches one of the static NAT entries (plus all the other firewall settings) is allowed. Each Firewall Profile can contain up to 64 static NAT pairs.
- Enhanced Outgoing Call Barring
  A number of new features have been added around the existing user setting Outgoing Call Bar 276 (User | Telephony | Supervisor Settings 276). When enabled, this setting stops the user from making any outgoing external calls other than those using Dial Emergency features. Note that the features below have also been added to the IP Office 4.1 2008 Q2 maintenance release.
  - Outgoing Call Bar Short Code Controls
    The short code features Outgoing Call Bar On 483 and Outgoing Call Bar Off 483 can be used by a user to change their outgoing call bar status. In addition the short code Change Login Code 451 has been added.
- User Password Controls
  Each user has a number of passwords.
  - Login Code Controls
    The user log in code is used to log in at an extension and to switch off outgoing call bar. A Change Login Code 451 short code feature has been added.

- **Voicemail Code**
  The voicemail code is used for access to the user's voicemail mailbox when accessed from destinations not configured as trusted sources. This code can be changed through the mailbox prompts interface or through Visual Voice. For IP Office 4.2 it can also be changed through the UMS Web Voicemail (IMAP) interface.

- **User Password**
  The user password is used by IP Office application such as Phone Manager, SoftConsole and TAPI. Phone Manager 4.2 now provides a Change Password option that allows the user to change their password. In addition to changing their password, it will update the Phone Manager profile if **Remember Password** is enabled and will attempt where possible to update the TAPI settings if the IP Office TAPI driver is installed on the same PC (Phone Manager will display a warning if the TAPI configuration needs to be manually updated to the new password).

- **Phone Time and Date Display**
  For 4.2, the following phones now display the current time and date at the bottom right of their display when idle: 2410, 2420, 5410 and 5420.

Platform Restrictions

Where some of the features listed about are not supported on all systems that can run IP Office 4.2, the table and notes below detail the restrictions.

Restricted Feature	Small Office Edition	IP406 V2	IP412	IP500
Mobile Call Control	✗	✗	✗	✓
one-X Mobile Support	✗	✗	✗	✓
Switchable To/So BRI	✗	✗	✗	✓
VoIP Enhancements	✗	✗	✗	✓
Embedded Voicemail Email	✗	✓	✗	✓
Embedded File Management	✗	✓	✗	✓
T3 Phone Enhancements	✗	✓	✓	✓
Multiple Music on Hold Sources	✗	✓	✓	✓

16.5 What was New in 4.1

This section summarizes the main changes in IP Office 4.1.

General Manager Changes

- Optional Legal / Informational Message Display 16
  When starting Manager, a message dialogue can be displayed which features Continue and Cancel (closing Manager) options. This can be used to display legal warning text or information text before Manager is used.

- Application Idle Timeout 110
  A timeout can be enabled. After 5 minutes of no mouse or keyboard activity, Manager will request reentry of the service user password used to load the current configuration or security settings.

- Card and Port Indication 188
  For line types provided by the installation of physical trunk cards or modules, within the line settings the card slot or module location is now indicated and the port number on that device.

- System Events 172
  The System Alarms tab has been renamed Systems Events.

- Merging Extension Changes 252
  For extension settings, the Base Extension and Disable Speakerphone fields are now mergeable.

Security Enhancements

- Service User Password Controls 85
  Various controls have been added within the IP Office security settings for service user passwords. These control can be used to enforce the required level of password complexity. Service user accounts can be disabled after too many incorrect password entries, after a specified period of not being used or after a specific expiry date. Service users can also be prompted to change their password when they next log in after a specified period.

- Security Reset 119
  Service users with sufficient rights can reset the IP Office security settings to their defaults using Manager. The command File | Advanced | Erase Security Settings (Default) 119 in available in configuration mode and File | Reset Security Settings 134 in security mode.

- Transport Layer Security (TLS) Secure Connection Support 68
  Secured communication is supported between the system and Manager using TLS for both authentication and encryption. Each of the IP Office services (configuration access, security access and system status access) can be configured for secure and/or unsecure connect. Secure connect can include the exchange of security certificates between the system and the Manager PC.

Avaya SIP for Branch Support

The Avaya SIP for Branch is a service delivered by interlinking Avaya switches through an Avaya SES (SIP Enablement Service) server. IP Office 4.1+ supports connect to the SES server through the following new features:

- SES Lines 245
  This type of line is used for connection from the system to the SES server. It is a variant of the SIP trunk type and requires the IP Office to have SIP Trunk Channel licenses available.

- Branch Prefix 143
  Each system within a SIP for Branch network requires a unique branch prefix. The Branch Prefix field on the System | System 143 tab is used to set that prefix. Calls to extensions on other systems within the network require the dialing of the branch prefix followed by the extension number.

- Local Number Length 143
  Extension numbers on systems within a SIP for Branch network should all be the same length. The Local Number Length field on the System | System 143 tab can be used to set the length of user, extension and hunt group extension numbers. Attempting to enter an extension number of a different length will cause a warning with Manager. Though intended for systems within a SIP for Branch network this field can be used in any system configuration.

General IP Office Features

- Time Profile Calendar Dates 360
  IP Office Time Profiles have been enhanced. In addition to the existing weekly time patterns, specific times on particular calendar dates can now also be added. Dates in the current and next year can be specified, including multiple dates.

- Incoming Call Route Multiple Time Profile Support 349
  Incoming call routes have now been amended to allow the use of multiple time profiles, with separate destinations and fallback destinations for use when a particular time profile is in effect.

- IP406 V2 LAN2 Support 148
  RJ45 Ethernet LAN port 8 on the front of the IP406 V2 control unit can now be specified as being the LAN2 port for the system. This enables the System | LAN2 155 tab and associated settings within the IP Office configuration. This mode is controlled by a Use Port 8 as LAN2 option on the System | LAN1 | LAN Settings 149 tab.

- System Events Syslog Support 172
  In addition to using SNMP and SMTP email, the IP Office can now send system events to up to two Syslog server destinations. The Syslog output can include IP Office Audit Trail events (not supported for SNMP and SMTP email).

- Phone Manager Pro Telecommuter Mode 291
  This additional mode has been added to the available Phone Manager Type options. This mode is supported by Phone Manager Pro 4.1+. Users of this mode can start Phone Manager as either a remote or local application. In remote mode the user uses a data connection for Phone Manager and specifies a telephone number available to them to make/receive calls. The IP Office then makes calls to that number when the user makes or answer calls using their Phone Manager application. In local mode the application function as a normal Phone Manager Pro with the user's associated extension. For full details refer to the Phone Manager User Guide and Phone Manager Installation Manual.

Hunt Group Operation

- Queuing Alert Extension 316
  For each hunt group, when a number of queued calls threshold is reached, an analog extension can be alerted with ringing. This is intended for an extension connected to a loud ringer device or similar (that is calls are not answered at that extension). This feature is controlled through the Hunt Group | Queuing 316 tab settings Calls in Queue Threshold and Pots Extension to Notify.

- Automatic Recording Mailbox for Account Codes 376
  By default automatic recordings for account codes are routed to the mailbox of the user making the call. Previously this could not be changed except through customized call flows on the Voicemail Pro. An alternate mailbox destination can now be specified through the Account | Voice Recording 376 tab.

Telephony

- Group Listen 630
  Using group listen allows callers to be heard through the phone's handsfree speaker but to only hear the phone's handset microphone. This feature can be enabled/disabled using either short codes and or a programmable button. Note: Group listen is not supported on IP phones.

- Disable Speakerphone 252
  The handsfree speaker enabled by the SPEAKER key on Avaya DS and IP phones can be disabled through the IP Office extension configuration settings. This allows handsfree operation to be disabled where such operation is not desirable.

Button Programming

- IP Office Date, Time and Version
  The Self Administer 651 button function can be used to set the system date and time. It can also be used to view the control unit type and software version. The user must be configured as a System Phone 274 (User | Telephony | Call Settings 274) and a value of *2* entered as the button's action data.

- Headset Force Feed 637
  On Avaya phones with fixed HEADSET buttons, a programmable button can be assigned to put the phone into headset force feed mode. In this mode, when headset mode is selected but the phone is idle, an incoming external call will cause a single tone and then be automatically connected.

- Group Listen 630
  The group listen feature (see Telephony above) can be assigned to a programmable button. That button can be used to switch group listen on/off and indicates when group listen is on.

- Enhanced Conference Meet Me
  For IP Office 4.1+ this button has been enhanced. Buttons associated with a particular conference ID will indicate when the conference is active. Callers connected on other appearance buttons can be transferred into the conference by pressing TRANSFER and then the Conference Meet Me button. This allows the user to place callers into the conference specified by the button without being part of the conference call themselves. This option is only support on Avaya phones with a fixed TRANSFER button (excluding T3 and T3 IP phones).

Short Codes

- Group Listen On/Off 477
  The group listen feature (see Telephony above) can be switched on/off through the use of the Group Listen On and Group Listen Off short code features.

- Default Embedded Voicemail Auto Attendant Short Codes
  Previously 4 system short codes were automatically added for each auto attendant added to the configuration. With the increase in the number of supported auto attendants to 40, the method of short code usage has changed to allow just 4 system short codes for all auto attendants by using the auto attendant number rather than name.

IP500

- IP Office 500 PRI Trunk Card (PRI-U)
  This card can be added as a trunk daughter card to any IP500 base card except the Legacy Card Carrier base card. The card is available in single and dual port PRI variants. The IP500 PRI-U card supports E1, T1 and E1-R2 PRI modes. To select the mode required, right-click on the line in the group or navigation pane and select Change Universal PRI Card Line Type. The systems supports 8 B-channels for each IP500 PRI-U port fitted, using in-service channels from port 9 of slot 1 upwards. Additional B-channels up to the capacity of ports installed and PRI mode selected require IP500 Universal PRI (Additional Channels) licenses added to the configuration. D-channels are not affected by licensing.

  - With the introduction of the IP500 PRI-U trunk daughter card, the design of stand off pillars supplied with IP500 trunk daughter cards has been changed. New cards will be supplied with 2 pre-fitted metal stand off pillar and 3 loose plastic pillars. Screws and washers are provided for the metal pillars for the final installation onto the IP500 base card. This changes is required for IP500 PRI-U cards but has been applied to all trunk daughter card types. This does not affect existing trunk daughter cards supplied with 5 plastic stand off pillars.

- H.323 Trunk Support in Standard Edition Mode
  H.323 trunks (IP trunks and QSIG trunks) on IP500 systems require the addition of IP500 Voice Networking licenses. At launch those trunk and license were only supported on IP500 system running in Professional Edition mode. IP Office 4.1 allows licensed H.323 trunks to be used in Standard Edition mode also.

- Small Community Networking in Standard Edition Mode
  The above change allows Small Community Networking to be used with IP500 systems running in Standard Edition mode. That includes Centralized Voicemail and, subject to the appropriate additional licenses, advanced Small Community Network features.

Embedded Voicemail

- 40 Auto Attendants 400
  Previously on 4 auto attendants were supported with the IP Office configuration. IP Office 4.1 allows up to 40 Auto Attendants to be created and to be chained together to support a flexible multi-tiered operation. Each auto-attendant is numbered and that number can be used in short codes for accessing the auto-attendant.

- Named Greeting Files - LVMGreeting 402
  For IP Office 4.1+ the utility required to create named greeting files (*LVMGreeting.exe*) is now provided with Manager. Those files can then be place on the embedded voicemail memory card and selected in the Recording Name field. The same recording can be shared between multiple auto-attendants. The tools is accessed from Manager via the menu command Advanced | LVM Greeting Utility 124. For full details refer to the IP Office Embedded Voicemail Installation manual.

Voicemail Pro

In conjunction with IP Office 4.1, Voicemail Pro 4.1 supports the following new features:

- **Automatic Recording Mailbox for Hunt Groups** 319
  By default automatic recordings for hunt groups are routed to the hunt group mailbox. Previously this could not be changed except through customized call flows on the Voicemail Pro. An alternate mailbox destination can now be specified through the Hunt Group | Voice Recording 319 tab.

- **Automatic Recording Mailbox for Account Codes** 376
  By default automatic recordings for account codes are routed to the mailbox of the user making the call. Previously this could not be changed except through customized call flows on the Voicemail Pro. An alternate mailbox destination can now be specified through the Account | Voice Recording 376 tab.

- **Call Data Tagging on Transfer Actions**
  The Transfer action now supports fields for setting the transfer source and description to display on phones receiving the transfer. The ability to associate call data for MS-CRM via Assisted Transfer actions is now also supported on Transfer actions.

- **Call Transfer Announcements**
  The Transfer and Assisted Transfer actions can be configure to announce the transfer to the caller. The announcement uses the recorded name of the mailbox associated with the transfer if available or the number if otherwise.

- **LIFO/FIFO Mailbox Operation**
  The default message playback order of First In-First Out (FIFO) can now be changed to Last In-First Out (LIFO). This is separately adjustable for new, old and saved messages. These are set through the System Preferences | Housekeeping tab (Administration | Preferences | General).

- **Time in Queue and Time on System Variables**
  Two new variables can be used in Queued and Still Queued call flows. They are $TimeQueued for the time in the queue and $TimeSystem for the time the call has been on the system.

- **Castelle Fax Server Support**
  The Voicemail Pro can be configured to recognize faxes of this type left in user's email mailboxes and include announcement of there presence in the user's mailbox prompts.

- **Hunt Group/Account Code Call Recording Destination**
  Previously the destinations for automatic call recording triggered by hunt groups or account codes could not be changed except through a custom Voicemail Pro call flow. The IP Office 4.1 configuration now allows the required destination for the call recording to be specified.

- **$DDI System variable for DDI Numbers**
  This variable is available on DDI calls passed from the IP Office to the Voicemail Pro.

- **Variable Routing (replaces the CLI Routing Action)**
  The existing CLI Routing action has been replaced by the **Variable Routing** action. This action allows the call routing to be based on matching specified values to system variables such as $CLI and $DDI. The numbers to which matching is performed can include wildcards such as ? for a single digits and * for any digits.

Key and Lamp Operation

- Abbreviated Ring Control 274
  For users with multiple call appearance buttons, it can be selected whether additional calls once a call is connected, are presented with a short single abbreviated ring or with normal ringing.

- Twinned Bridge/Coverage/Line Appearances 293
  For users with a twinned secondary phone, only calls on call appearances at their primary phone also alert at the secondary phone. For IP Office 4.1 it is possible to configure that calls alerting on bridged, coverage and line appearance buttons on the primary should also alert on the secondary.

Licenses

- **IP500 Voice Networking**
  This type of license is now supported on IP500 systems running in Standard Edition mode. This allows those systems to use IP and QSIG trunks and to be included in a Small Community Network.

- **IP500 PRI Universal (Additional Channels)**
  The systems supports 8 B-channels for each IP500 PRI-U port fitted. Additional B-channels up to the capacity of ports installed and PRI mode selected require IP500 Universal PRI (Additional Channels) licenses added to the configuration. D-channels are not affected by licensing.

- **VPN IP Extensions Licenses**
  IP Office 4.1+ supports 4610, 4621, 5610 and 5621 phones running the VPNremote firmware. These work in conjunction with the Extension | VoIP | VPN Phone Allow setting. The phones are licensed against VPN IP Extensions licenses added to the IP Office configuration.

Windows Operating System Support

- **Windows Vista**
  With IP Office 4.1, Vista support is expanded to all IP Office 4.1 applications also supported on Windows XP Pro. Note that the Vista support refers to Vista Business Ultimate and Vista Enterprise editions; the Vista Home Basic and Vista Home Premium editions are not supported.

System Status Application

The following changes have been made to the System Status Application (SSA) including in the IP Office 4.1 suite.

- **Device Version Numbers**
  IP500 trunk, extension and VCM cards have electronic version numbers. SSA is now able to display those version numbers.

- **Digital Trunk Clock Source Change Alarm**
  SSA can report when the clock source being used by the system changes from one digital trunk to another.

16.6 What was New in 4.0 May 2007

This section summarizes the changes made for the May 2007 maintenance release of the IP Office 4.0 suite of applications. This is not an exhaustive list. For full details refer to the appropriate IP Office Technical Bulletin.

- Call Pickup Line Short Code 445
  This short code feature allows the answering of calls ringing, parked or held on a line by use of the line's Line Appearance ID. This is intended for use with phones where Line Appearance buttons cannot be programmed. Not supported on T3 phones.
- Call Pickup User Short Code 446
  This short code feature allows the answering of calls ringing, parked or held against a user by use of the user's Extension Number setting. If the targeted user has multiple calls, preference is given to ringing, parked and held calls in that order. Not supported on T3 phones.
- Embedded Voicemail 400
  The following new features appears for Embedded Voicemail:
  - Enable Local Recording Control 402
    The use of short code to record auto attendant greetings can be disabled if required. The short codes are still active and can be used to hear the current prompts, however the option to record a new prompt will not operate.
  - Embedded Voicemail Shutdown Short Code 498
    The Embedded Voicemail memory card is not a hot-swappable device, and so the system must be switched off in order to remove/replace the memory card. Use of this short code feature allows safe removal of the memory card whilst the system is running.
- Known IP Office System Discovery 59
  Manager can be configured to record details of systems it discovers and to then allow later rediscovery of those known systems. This feature is only available when Manager has been configured with a .csv file location to which to record known system details. When this has been done, a Known Units button is available within the Manager discovery screen.
- Custom Locale 143
  For some systems, the locales fully supported by IP Office may not match the local requirements. In these cases, the option Customize can be selected in the Locale field of the System | System tab. Additional options are then made visible to select the tone usage, CLI type and busy tone detection settings.
- System Time Setting Through Phones 727
  For systems without access to an appropriate time server or where an immediate time and date change is required, a manual method for setting the system date and time has been added.
- Internal Twinning for North America 293
  The Internal Twinning function was added in the IP Office 3.1 release but not supported for systems with a North American locale. This feature is now supported in North American locales.
- IP DECT 227
  The Avaya IP DECT solution is now supported in North American locales. For Manager this means that an IP DECT line and IP DECT extensions can now be created on systems with a North American locale.
- 3641 and 3645 Wireless IP Phones
  These phones are now supported. They can be used with 802.11a, 802.11b and 802.11g wireless networks.
- T3 Direct Media Support
  T3 IP phones have previously not supported Direct Media mode connection. For T3 IP phones with T246 or higher firmware that restriction no longer applies and the specific restrictions on the number of T3 IP phones can be removed.

16.7 What was New in 4.0

This section summarizes the main changes in the IP Office 4.0 General Availability release.

Manager Changes

- Validation Control
  Control of when Manager applies automatic validation to configuration files is now available through the File | Preferences menu.

- Busy on Held Validation
  The Tools menu now contains a Busy on Held Validation option that checks all users. Previously this check was only performed when a user's settings were edited.

- Line Renumber
  The Tools menu now contains a Line Renumber option for renumbering all Line Appearance ID's upwards from a selected base number.

- BOOTP Entries
  The maximum number of BOOTP entries has been increased from 20 to 50.

Hardware Support Changes

- IP Office 500 System Unit (IP500)
  This control unit has no integral extension or trunk ports. On its front the unit has 4 card slots into which IP500 base cards can be inserted. These card provide various combinations of digital station port, analog extension ports, voice compression channels and trunk ports. On its rear the unit has slots for embedded voicemail, a feature key dongle slot, audio port, door relay switch port and ethernet LAN/WAN ports plus 8 external expansion module ports.

  - 4406, 4412 and 4424 are only supported on external expansion modules. They are not supported directly on the IP500 system unit.

- IP Office Standard Edition
  By default the IP500 control unit runs a subset of full IP Office functionality called IP Office Standard Edition. In this mode the IP Office is restricted as follows. These restriction can be removed by adding an IP500 Upgrade Standard to Professional license to the configuration.

  - The IP500 is restricted to a maximum of 32 users using ports on base cards in the control unit.

  - In Standard Edition mode the IP500 does not support any external expansion modules.

  - The applications Embedded Voicemail, Phone Manager Lite/Pro, SoftConsole, TAPI, Delta Server and CBC, Manager, SSA and Monitor are supported.

    - Advanced applications such as Voicemail Lite/Pro, CCC, Conferencing Center, MS-CRM, etc are not supported.

  - IP trunks (H323, QSIG, SCN) not supported. IP DECT and SIP trunks are supported. Enabling IP trunks requires an IP500 Upgrade Standard to Professional license and IP500 Voice Networking licenses. Note: This restriction was removed in later IP Office 4.0 maintenance releases.

  - Meet-me conferencing is not supported.

- Hardware Support
  The following hardware is not supported with IP Office 4.0.

  - The WAN3 external expansion module is not supported (the WAN3 10/100 is still supported).

  - All Network Alchemy external expansion modules are no longer supported.

  - The IP403 and IP406 V1 control units are not supported.

- Phone Support
  The following phones are not supported by IP Office 4.0. They may function but have not been tested with 4.0 and any faults reported with 4.0 will not be fixed.

  - The 20DT Analog DECT phone used with IP Office Analog DECT and Compact DECT is not supported. It may be used with Avaya IP DECT but only as a generic GAP compatible DECT device.

  - The 4606, 4612 and 4624 phones are no longer supported.

  - The Transtalk 9040 is no longer supported.

System Status Monitoring

- **System Status Application (SSA)**
  This application provides information about the equipment and resources in IP Office 4.0 and higher systems. This information includes indication of alarms and details of current calls in progress. Use of SSA requires a service user name and password configured for System Status in the IP Office's security settings.

- **Call Status Application**
  This application is not supported by IP Office 4.0. It has been replaced by the IP Office System Status Application above. Call Status is still included in the IP Office Admin suite for use with pre-4.0 systems.

- **Monitor**
  The SysMonitor application has been enhanced but is no longer fully backwards compatible with pre-IP Office 4.0 systems. Therefore two versions of Monitor are included in the IP Office Admin suite; version 6.0 for use with IP Office 4.0 systems and version 5.2 for use with pre-IP Office 4.0 systems.

SIP Trunks

- IP Office 4.0+ supports SIP calls through the implementation of SIP trunks. Through normal short code routing of outgoing group ID's any user can make outgoing calls using SIP services, ie. users do not require SIP phones to make and receive SIP calls. Incoming call routing can be used to route incoming calls on SIP trunks. SIP trunks are a licensed feature.

Hot Desking (Logging In/Out)

- **Agent Status on No Answer**
  The IP Office can change the status of call center agents who do not answer a hunt group call presented to them. This can include logging the agent off the system. The change of status can be set per user and the use of this option can be set per hunt group.

- **Remote Hot Desking**
  Users can now hot desk between systems in a Small Community Network. This requires a Advanced Small Community Networking license in the system where a user logs in remotely.

- **Logging Out**
  User who do not have a log in code set cannot log out.

- **NoUser User**
  By default the NoUser user's first programmable button is set to the **Login** function.

Voicemail

- Voicemail Channel Reservation
  IP Office 4.0 allows the licensed voicemail channels between Voicemail Pro and the IP Office to be reserved for particular functions or to be left unreserved for any function.

- Visual Voice
  Users with Avaya multi-line display phones can use a display menu driven interface for accessing and controlling the playback of messages in voicemail mailboxes. This is supported with Voicemail Pro, in Intuity emulation and IP Office modes, and Embedded Voicemail.

- Voice Recording
  A number of improvements have been made to call recording operation in conjunction with Voicemail Pro. In the descriptions below 'party' can mean user, hunt group or incoming call route involved in a call.

  - Calls including IP end points, including those using Direct Media, can now be recorded.
  - Voicemail Pro automatic call recording can be triggered by Incoming Call Routes.
  - Where recording is triggered by several parties within the same call, separate recordings are produced for each party.
    - For example if both automatic hunt group recording and automatic user recording are applicable to the same call, separate recordings are produced for both the hunt group and the user.
    - If a call is to be recorded multiple times to the same mailbox only a single recording is made; resolved in the order of: incoming call route, account Code, hunt group and user settings.
  - Recording only continues while the party triggering the recording is part of the call, for example:
    - Recording triggered by a user stops when that call is transferred to another user.
    - Recording triggered by a hunt group continues if the call is transferred to another member of the same group. Recording stops if the call is transferred to a user outside the hunt group.
    - Recording triggered by an incoming call route continues for the duration of the call on the system.
  - Parking and holding a call pauses recording. Recording is restarted in the same file when the calls is unparked or taken of hold.

- User Announcements
  With Voicemail Pro 4.0 and higher, announcements are supported for calls waiting to be answered by an individual user. User start points in Voicemail Pro now include Queued and Still Queued options.

- Embedded Voicemail
  - Embedded voicemail is supported on the IP500 control unit using the same options as the IP406 V2 control unit.
  - Hunt group announcements are supported using embedded voicemail.
  - The auto-attendant menu includes a Fax option for rerouting fax calls.
  - Support for Visual Voice.
  - Support for Fast Forward (#), Rewind (*), Skip message (9) and Call Sender (**) when listening to messages.
  - Support for 3 voicemail reception destinations using *0, *2 and *3.

Hunt Groups

- **Agent Status on No Answer**
  The IP Office can change the status of call center agents when a hunt group call is presented but not answered. The agent can be put into busy wrap-up, busy not available or logged out. The change of status can be set per user and the use of this option can be set per hunt group. This feature is not applied if the call is answered elsewhere before the No Answer Time expires.

- **Fallback**
  Night service fallback using a time profile is no longer applied to a hunt group already set to out of service. Short codes and buttons can be used to set a hunt group out of service, overriding the night service time profile.

- **Voicemail Answer Time**
  A separate value has been added to hunt group settings to control when hunt group calls go to voicemail if unanswered. The default value is 45 seconds.

- **Queuing**

  - Previously the definition of queued calls did not include calls ringing against hunt group members. The definition now includes ringing calls and calls waiting to be present for ringing.

  - Control and usage of announcements has been separated from queuing (see below).

  - The queue limit can be set to include queued and ringing calls or just queued calls.

- **Announcements**

  - Hunt group announcements have been separated from hunt group queuing and can be used even when queuing is off.

  - Hunt group announcements are now supported by Embedded Voicemail in addition to Voicemail Pro and Voicemail Lite.

  - The times for the first announcement, second announcement and between repeated announcements are configurable.

- **Advertised Hunt Groups**
  A hunt group can be set to be 'advertised'. This requires an Advanced Small Community Networking license. Hunt groups that are advertised can be dialed by users on other systems within the Small Community Network (SCN) without the need for short codes. This feature requires entry of a Advanced Small Community Networking license on each system.

- **SCN Distributed Hunt Groups**
  Hunt groups in a Small Community Network can include members located on other systems within the network. This feature requires entry of a Advanced Small Community Networking license on each system. Distributed hunt groups are automatically advertised to other systems within the SCN.

- **Idle Status**
  For longest waiting hunt groups, the type of calls that change a hunt group member's idle status can be selected.

- **Call Presentation**
  When additional calls are waiting to be presented, additional hunt group members are alerted using the hunt group type. However when any member answers a call it will be the first waiting call that is answered.

- **Set Hunt Group Night Service and Set Hunt Group Out of Service Short Codes**
  Previously the Set Hunt Group Night Service 493 and Set Hunt Group Out of Service 493 short code features toggled. That behaviour is not supported in 4.0 and higher.

- **Voicemail Mailbox Operation**
  For IP Office 3.2 and earlier, when voicemail was invoked, the mailbox of whichever hunt group was currently handling the call was used, for example the mailbox of the overflow or night service hunt group might be used if the call had gone from the original hunt group to an overflow or night server hunt group. For IP Office 4.0 and higher, the mailbox of the originally targetted hunt group is used even if the call has overflowed or gone to a night server hunt group.

Alternate Route Selection (ARS)

- **Least Cost Routes (LCR)**
  LCR has been replaced by ARS. On systems being upgraded to IP Office 4.0, any LCR entries will be automatically converted as far as possible to ARS forms. However due to the difference in method of operation the ARS forms will need to be checked.

- **Secondary Dial Tone**
  Where secondary dial tone is required it is provided through a check box option in ARS. This simplifies the configuration of secondary dial tone.

- **Outgoing Call Routing**
  LCR forms were never explicitly applied to particular calls. Instead any number to be dialed externally, was compared to the LCR short codes for a possible match. In IP Office 4.0, dialing short codes are explicitly routed either to a outgoing line group or to an ARS form.

ISDN Features

The following ISDN features are now supported by IP Office 4.0+. Note that availability of these feature is dependant on their also being supported and available from the ISDN service provider for which there may be charges.

- **Malicious Call Identification - MCID**
  Short codes and button programming features have been added so that users can be configured to trigger this activity at the ISDN exchange when required.

- **Advice of Charge - AOC**
  Advice of charge during a call (AOC-D) and at the end of a call (AOC-E) is supported for outgoing ISDN calls other than QSIG. The call cost is displayable on T3 phones and included in the IP Office Delta Server output. The IP Office allows configuration of call cost currency and a call cost mark-up for each user.

- **Call Completion to Busy Subscriber - CCBS**
  CCBS can be used where provided by the ISDN service provider. It allows a callback to be set on external ISDN calls that return busy. It can also be used by incoming ISDN calls to a busy user.

- **Partial Rerouting - PR**
  When forwarding a call on an ISDN channel to an external number using another ISDN channel, partial rerouting informs the ISDN exchange to perform the forward, thus freeing the channels to the IP Office. Not supported on QSIG.

Advanced Small Community Networking (Advanced SCN)

The following new features are supported for IP Office 4.0+ Small Community Networks. Note that these feature require entry of an Advanced Networking License into systems in the network.

- **Network Hot Desking**
  Hot desking is supported between systems within the Small Community Network.

- **Distributed Hunt Groups**
  Hunt groups can now include members who are located on different systems within the Small Community Network. This feature requires entry of a Advanced Small Community Networking license on each system.

- **Break Out**
  This feature is provided primarily to support network hot desking but can be used for other purposes. It allows the dialing on one system in the network to be done as if dialed locally on an other system.

Key and Lamp Operation

The following key and lamp operation features were added in IP Office 4.0:

- **Delayed Ring Preference**
  This user telephony setting works in conjunction with the user's Ringing Line Preference setting. It sets whether ringing line preference should use or ignore the ring delay applied to the user's appearance buttons.

- **Answer Pre-Select**
  Normally when a user has multiple alerting calls, only the details of the call on current selected button are shown. Pressing any of the alerting buttons will answer the call on that button, going off-hook will answer the current selected button. Enabling the user telephony setting Answer Pre-Select allows the user to press any alerting button to make it the current selected button and displaying its call details without answering that call. To answer a call when the user has Answer Pre-Select enabled, the user must press the alerting button to display the call details and then either press the button again or go off-hook.

- **Reserve Last CA**
  Phones with appearance buttons require a free call appearance button to perform actions such as call transfers. However it is possible in some scenarios for all available call appearances to be occupied or alerting. The user telephony setting Reserve Last CA can be used to restrict the user's last call appearance button for outgoing calls only.

Licences

The following new licenses are used by IP Office 4.0:

- **IP500 Upgrade Standard to Professional**
  This license is required for an IP500 system to run in IP Office Professional Edition mode rather than IP Office Standard Edition mode. It is a pre-requisite for the IP500 Voice Networking licenses and any licensed features not supported in Standard Edition mode.

- **IP500 Voice Networking (Base 4 channels)**
  For IP500 systems running in Professional Edition mode, this licences enables support for H323 IP trunks between systems and QSIG or Small Community Networking over those trunks.

  - **IP500 Voice Networking (Additional channels)**
    Allows an additional 4 H323 voice networking trunks.

- **IP500 VCM Channels**
  Used with the IP500 VCM 32 and IP500 VCM 64 base cards. Each card supports 4 channels by default, with additional channels enabled by the addition of licenses.

- **SIP Trunk Channels**
  This license is used to configure the maximum number of simultaneous SIP trunk calls supported. Multiple licenses can be added for the cumulative number of SIP trunks required.

- **Advanced Small Community Networking**
  This license is used to enable support for hosting hot deskers from remote systems, creation of distributed hunt groups and the viewing of advertised hunt groups.

Other Features

- **Private Call**
  Users can set a status of private call using short codes or a programmed button. Private calls cannot be recorded, intruded on, bridged into or monitored.

- **RTP Relay**
  RTP relay allows much more efficient use of the voice compression channels available in a system:

  - Calls between IP endpoints using the same audio codecs that are routed via the IP Office (for example when not using direct media path) no longer use voice compression channels.

  - Call setup and progress tones no longer require a voice compression channel. The exceptions are short code confirmation tones, ARS camp on tone, account code entry tone and G723 calls (except Call Waiting).

  - Page calls to IP devices use G729a only and therefore only 1 channel regardless of the number of IP devices.

  - For T3 IP devices to benefit from RTP relay they must be configured to 20ms packet size.

- **Password Lockout**
  Any phone features that require a validated entry (for example password or account code entry) will automatically fail if they have been preceded by 4 failed attempted in the previous 90 seconds.

- **Firewall IP Office Service Controls**
  The IP Office firewall standard settings now include controls to drop or allow connects to IP Office 3.2 configuration settings, security settings and system status.

- **User Announcements**
  User announcements can be configured for use with Voicemail Pro. These announcements are used for external calls waiting to be answered.

- **Feature Key Dongle Serial Number**
  The IP Office configuration settings now displays the serial number of the last Feature Key dongle with which the system validated its licenses and whether the dongle is local (ie. serial or Smart Card) or remote (ie. USB or parallel).

- **Line ID Numbers**
  For defaulted systems, all lines supported line ID numbers are numbers from 701 upwards by default. The Line Renumber tool is now available again within Manager to renumber all lines starting from a user selected starting number.

- **Ending Conferences**

  - For pre-4.0 systems, if a conference has two parties, and one party leaves, the conference call is ended. This may affect conferences that are just beginning but currently only contain the first two parties to join.

  - For IP Office 4.0, a conference remains active until the last extension or trunk with reliable disconnect leaves. Connects to voicemail or a trunk without reliable disconnect (for example an analog loop-start trunk) will not hold a conference open.

- **Disconnect Tone**
  For digital and IP phones, when the IP Office detects that the far end of a call has disconnected it can either make the near end go idle or play disconnect tone. By default this behaviour depends on the system locale. The Disconnect Tone 163 (System | Telephony | Tones & Music 163) field can be used to override the locale default and force either disconnect tone use or go idle.

Chapter 17.
Ports

17. Ports

As mentioned, a number of different ports are used for access to systems. The following table lists some of the ports on which the control unit listens for different types of access.◀ Indicates a listening port on the control unit. ▶ indicates a port to which the system sends, for example to a PC running an application.

* Indicates that the port and or protocol can be changed.

Port		Protocol		Function
25*	▶	SMTP	TCP	Email system alarms from the system to SMTP server. For IP Office 4.2 also used for Voicemail Email on Embedded Voicemail.
37	▶	Time	UDP	Time requests from the system to a Time Server (RFC868).
53	◀	DNS	UDP	Domain Name Service responses.
67	◀	BOOTP/DHCP	UDP	DHCP server operation.
68	▶	BOOTP/DHCP	UDP	DHCP client operation.
69	◀	TFTP	UDP	File requests to the system.
69	▶	TFTP	UDP	File requests by the system.
80	◀	HTTP	TCP	HTTP file requests to the system.
161*	◀	SNMP	UDP	From SNMP applications.
162*	▶	SNMP Trap	UDP	To addresses set in the system configuration.
500	◀	IKE	UDP	Key exchange for IPSec protocol.
389*	▶	LDAP	TCP	Lightweight Directory Access Protocol.
514	▶		UDP	Syslog client.
520	▶	RIP	UDP	To and from the system to other RIP devices. For RIP1 and RIP2 (RIP1 compatible) the destination address is a subnet broadcast, eg. 192.168.42.255. For RIP2 Multicast the destination address is 224.0.0.9.
520	◀	RIP	UDP	
833	◀	Network Relay	UDP	Network relay.
1701	◀	L2TP	UDP	Layer 2 tunneling protocol.
1718	◀	H.323	UDP	H.323 Discovery
1719	◀	H.323 RAS	UDP	H.323 Status. VoIP device registering with the system.
1720	▶	H.323/H.245	UDP	H.323 Signalling. Data to a registered VoIP device.
2127	▶	(UDP)	UDP	PC Wallboard to CCC Wallboard Server.
3478	▶	SIP	UDP	Port used for STUN requests from the system to the SIP provider.
5005	◀	RTCP Monitor	UDP	RTCP Monitoring information from Avaya H323 phones. *(IP Office 5+)*
5060	◀▶	SIP	UDP/ TCP*	SIP Line Signalling/SIP End points.
8080	▶	HTTP	TCP	one-X Portal for IP Office
8089	▶	Enconf	UDP	From the system to the Conferencing Center Server Service. User access to the conference center is direct via HTTP sessions.
8888	▶	HTTP	TCP	Browser access to the system ContactStore (VRL) application.
49152-53247 *	◀▶	RTP/RTCP	UDP	Dynamically allocated ports used during VoIP calls for RTP and RTCP traffic. The port range can be adjusted through the System | LAN1 | VoIP[150] tab.
50791	▶	IPO Voicemail	UDP	To voicemail server address.
50793	◀	IPO Solo Voicemail	UDP	From system TAPI PC with Wave drive user support.
50794	◀	IPO Monitor	UDP	From the system Monitor application.
50795	◀	IPO Voice Networking	UDP	Small Community Network signalling (AVRIP) and BLF updates. Each system does a broadcast every 30 seconds. BLF updates are sent required up a maximum of every 0.5 seconds.
50796	◀	IPO PCPartner	UDP	From a system application (for example Phone Manager or SoftConsole). Used to initiate a session between the system and the application.
50797	◀	IPO TAPI	UDP	From a system TAPI user PC.
50798	▶	(UDP)	UDP	*BT Fusion variant. No longer used.*
50799	▶	IPO BLF	UDP	Broadcast to the system LAN and the first 10 IP addresses registered from other subnets.
50800	▶	IPO License Dongle	UDP	To the License Server IP Address set in the system configuration.
50801	◀	EConf	UDP	Conferencing Center Service to system.
50802	◀	Discovery	TCP	System discovery from Manager.

Port		Protocol		Function
50804 *	◄	Service Access Protocol	TCP	System configuration settings access.
50805 *	◄		TCP	" TLS Secure.
50808 *	◄		TCP	System status access. Used by the System Status Application (SSA) and Customer Call Reporter Application (CCR).
50812 *	◄		TCP	System security settings access.
50813 *	◄		TCP	" TLS Secure.
50814 *	◄		TCP	System Enhanced TSPI access. *(IP Office 5+)*

- CDR/SMDR from the IP Office is sent to the port number and IP address defined during configuration and using either TCP or UDP as selected.

Chapter 18.
Single Server Support

18. Single Server Support

The following scenarios are supported for combining system server applications onto a single server PC.

In all cases, the individual requirements of each application as if installed on a separate server are still applicable. Also, depending on the application combination, additional restrictions and requirements may be applied as detailed below.

<table>
<tr><th></th><th>Voicemail Pro</th><th>Customer Call Reporter</th><th>one-X Portal for IP Office</th><th>Minimum IP Office</th><th>Minimum PC Specification</th></tr>
<tr><td>1.</td><td>16 Ports</td><td>150 Agents</td><td>–</td><td>Release 5.0</td><td>As per each application.</td></tr>
<tr><td>2.</td><td>8 Ports
(4 TTS)</td><td>–</td><td>50 Simultaneous users.</td><td>Release 6.0</td><td>2GHz Dual Core, 4GB RAM, Windows 2008 Server (32 or 64-bit).</td></tr>
<tr><td>3.</td><td>8 Ports
(4 TTS)</td><td>30 Agents</td><td>50 Simultaneous users.</td><td>Release 6.0</td><td rowspan="2">2GHz Quad Core, 6GB RAM, Windows 2008 64-bit.
CCR run in Windows 2003 on a virtual server.</td></tr>
<tr><td>4.</td><td>16 ports
(8 TTS)</td><td>50 Agents</td><td>150 Simultaneous users.</td><td>Release 6.0</td></tr>
</table>

- Voicemail Pro includes UMS, VB Scripting and 3rd party database operation. It also includes the installation of ContactStore if required.
- Both ContactStore and one-X Portal for IP Office use Tomcat servers as part of the application. For scenarios with both installed, the redirect port setting of the ContactStore's Tomcat server should be configured to a port other than 8080.
- The supported virtual servers are:
  - VMWare Server.
  - Microsoft Virtual Server 2005 R2.
  - Microsoft Server Hyper-V.
- When used in a virtual server configuration, Customer Call Reporter and one-X Portal each require a 2GB RAM virtual machine. Voicemail Pro and ContactStore each require a 1GB RAM virtual machine.

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