IP Office 7.0 and BCM 6.0 – SIP Interoperability Configuration Notes

Abstract: This document provides information on how to configure a network solution with IP Office 7.0 and BCM 6.0 using SIP trunks.

Document Publication History

Issue	Date	Status	Author(s)	Change Summary
1.0	April 19, 2011	Final	R. Appleton, A.Celer	Initial version of the document

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1.0 Overview

1.1 Capabilities

This document provides a description of the solution where a network of BCMs is connected to a network of IP Offices. In this configuration single BCM is connected to a single IP Office using SIP trunks.

- Basic Call:

  Basic Call Completion
  Handling of busy called party
  DTMF and ring-back tone
  Hold and Retrieve on both ends
  Call Waiting presentation
  Called Number display
  Calling number and name display
  Abandoned call

- Call Redirection (at node level - **no network optimization**):

  Call Forward
  Call Transfer (blind and consultative)
  Call Redirection, including call redirect to voicemail

- Conferencing

  3-party and multi party conferencing

- FAX
- Tandem call
- Privacy-Name and Number Blocking

1.2 Exclusions

The following was not tested since it is not supported:

- BCM users using IP Office voicemail system
- IP Office users using BCM voicemail system

1.3 Known Issues

1.3.1 CFAC to remote endpoint

Test Procedure

1. Called party has CFAC to remote set on the other switch(originating switch)
2. Caller calls called party on remote switch
3. Verify ringing on the forwarded destination
4. Verify displays on both sets
5. Answer the call at the forwarded destination and verify 2-way audio
6. Hang up the call and verify successful call termination

25 - 30 seconds later, set C drops the call
Set A still shows the call as live (timer is still going and icon is lit)

1.3.2 Blind Transfer to Originating switch

The target of the of the transfer will display the external line that the call is on and not the set on the other end and the transferee will display the set that did the transfer and not the target, even after the transfer is complete.

```
BCM                                     IP Office
Set A           Set B                   Set C
|------------------------Originate------------------------→|
|←----------------------Accept-------------------------→|
|←----Transfer-----→|                                      |
|                   |←----------Call establish-----→|
```

Display of Set B is the line number that the call is on and the display of Set C is of the originator, Set A.

1.3.3 Blind Transfer to remote switch

Display appears to not function correctly. If Set A on BCM calls Set B on IP Office and then IP Office transfers to Set C on BCM, then hangs up. When Set C answers call, both Set A and Set C have their display appearing as though they are both connected to Set B who has since exited the call.

BCM IP Office

```
Set A            Set B                          Set C
|-----------------------Originate----------------------->|
|<---------------------Accept------------------------->|
|                        |<-------------Transfer------>|
|<---Call establish ->|
```

Display of both Set A and Set B is that it is connected with Set C

Same display problem appears if both Set A and Set B are on IP Office and Set C is on BCM.

1.3.4 Consult Transfer To Originating switch

Note: same display note as in test case 1.3.2

1.3.5 Consult Transfer To Remote switch

Note: Same display note as in test case 1.3.3

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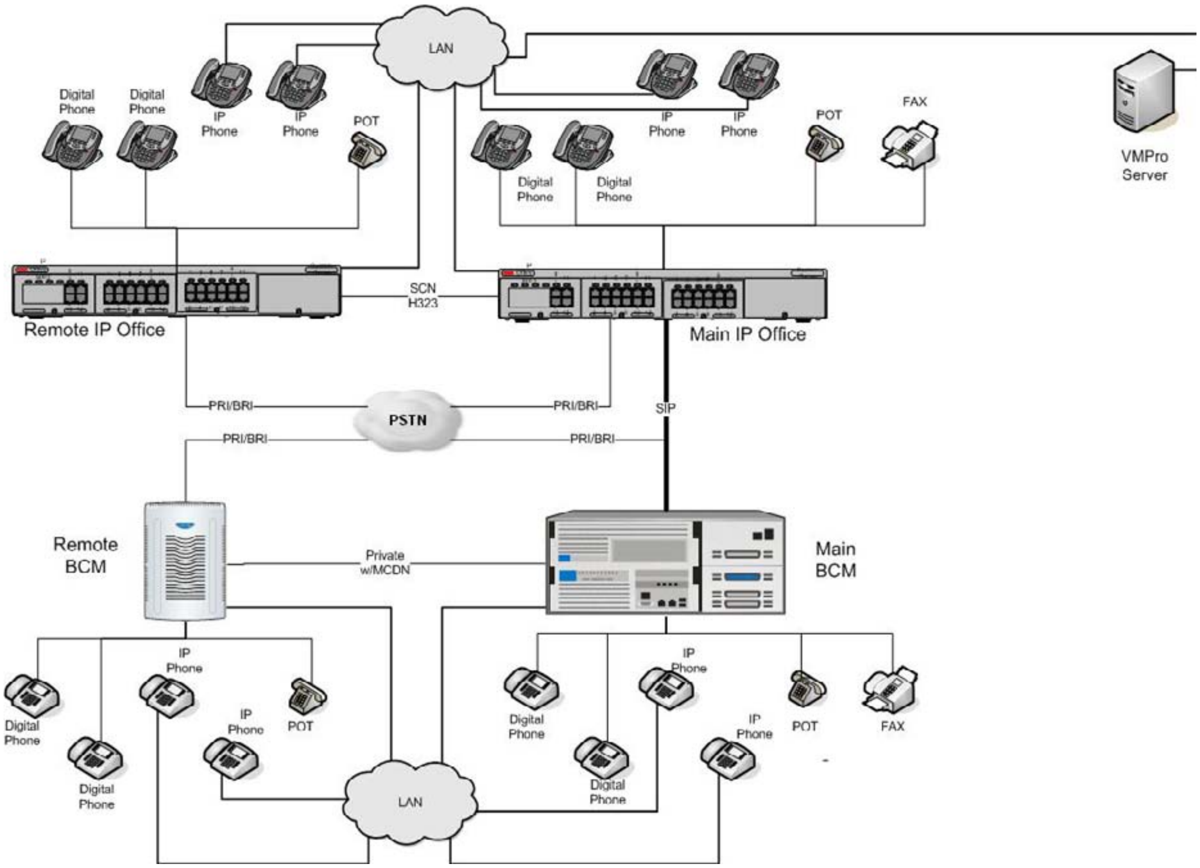
1.4 Network Diagram



2.0 Configuration Guide

2.1 IP Office Software Versions

- IP Office 7.0 release

2.2 BCM Software Versions

This Solution Configuration Guide is applicable to the following BCM Releases:

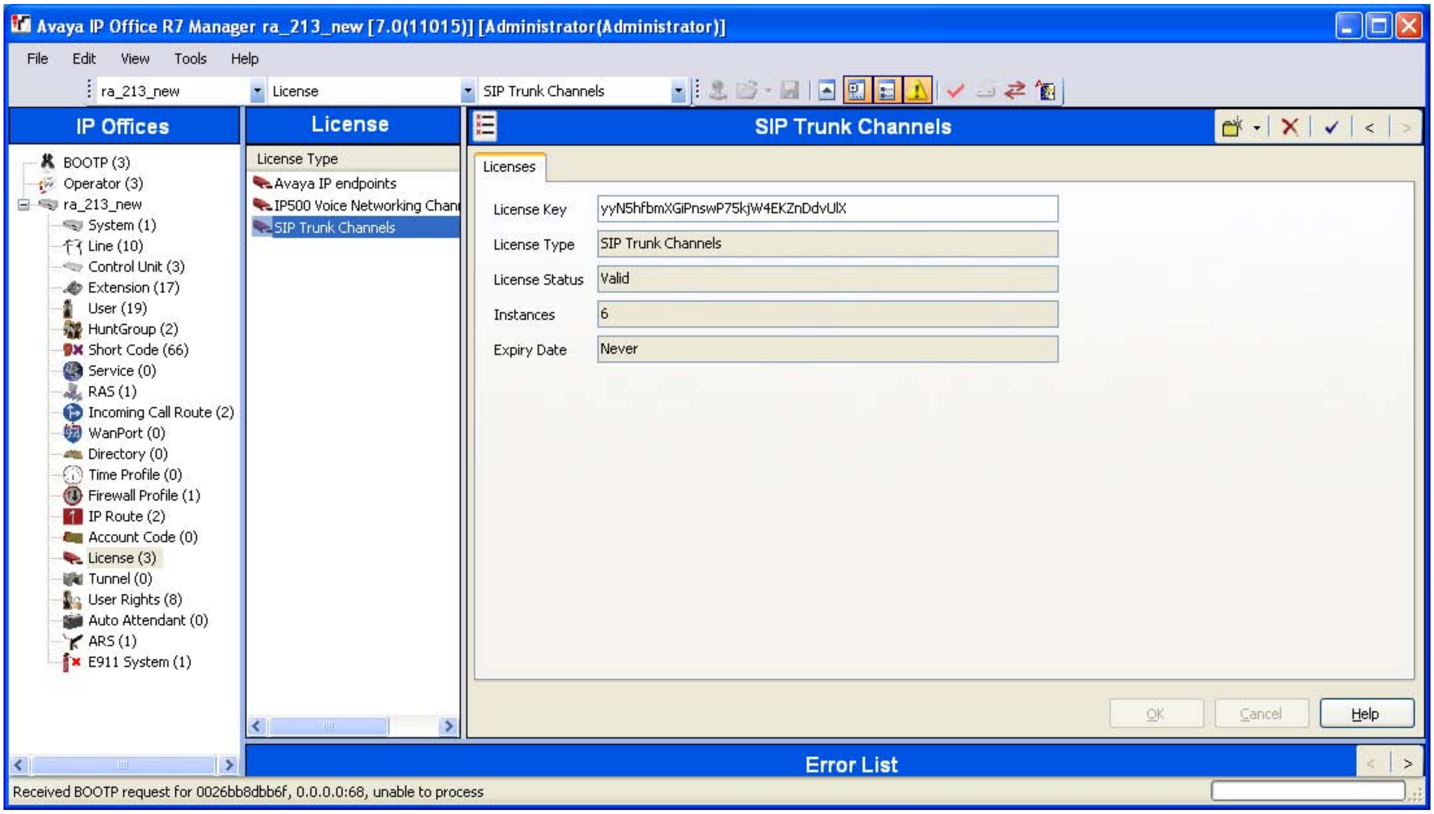
- BCM50 rls 6.0 with the latest SU

- BCM450 Rls 6.0 with the latest SU

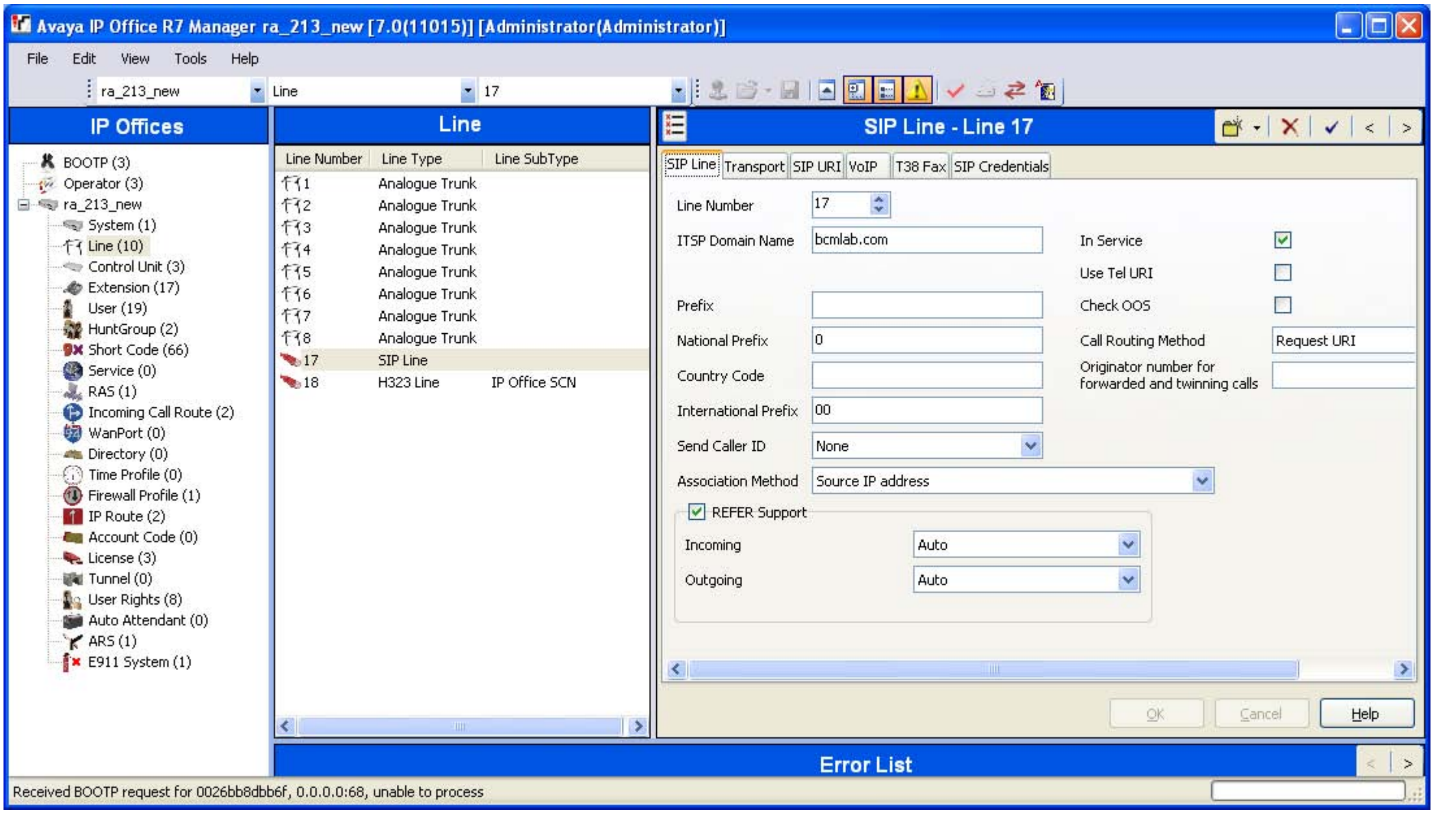
2.3 Provisioning SIP trunks on IP Office

This is only a representative configuration. For more detailed explanations, please refer to the IP Office SIP trunking guide.

2.3.1 IP Office SIP Trunk Keycodes

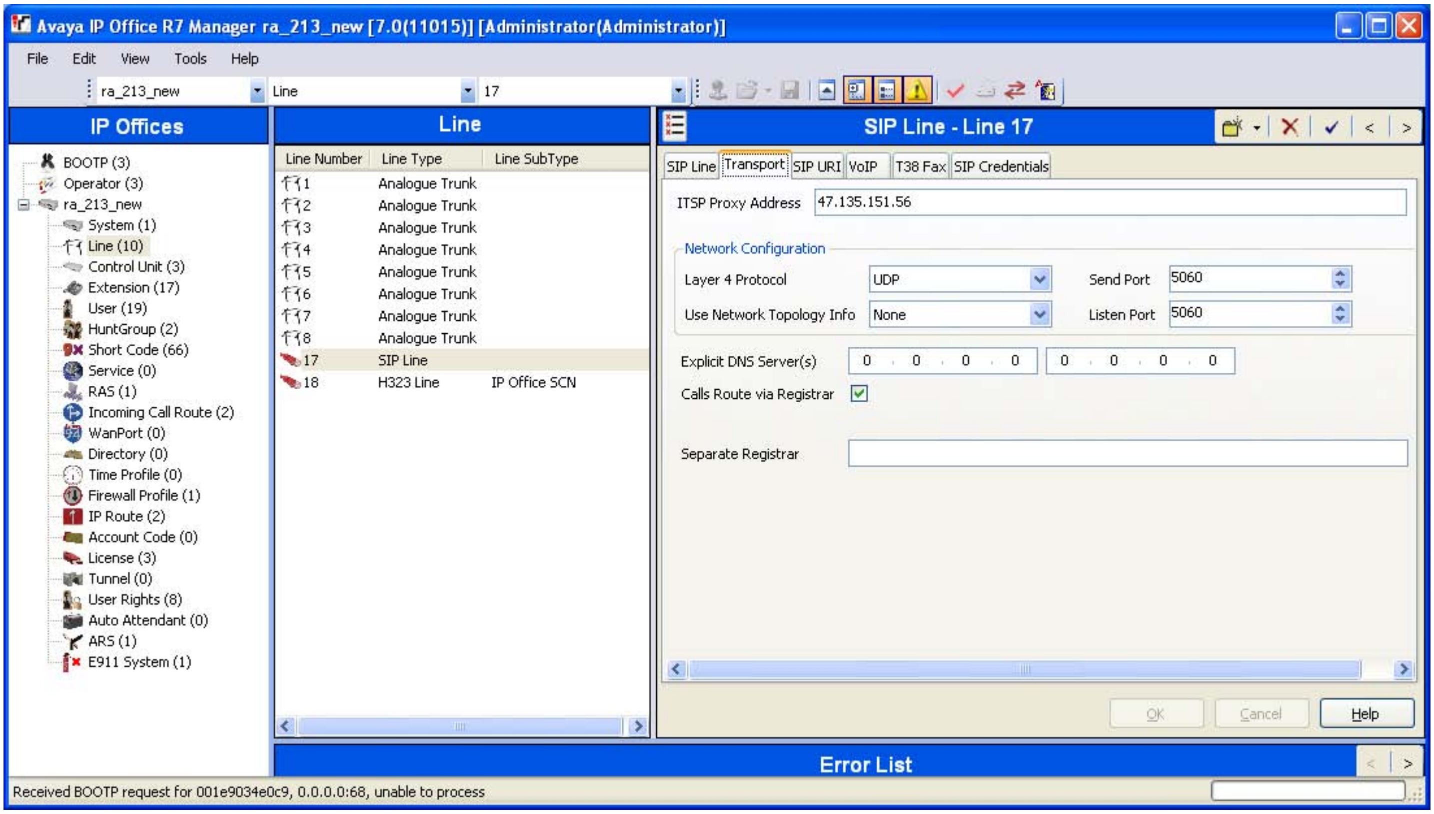


2.3.2 IP Office SIP Line



Some switches require a domain name match. BCM does not require it, but troubleshooting networks may be easier if this information is included.

2.3.3 IP Office SIP Line Transport

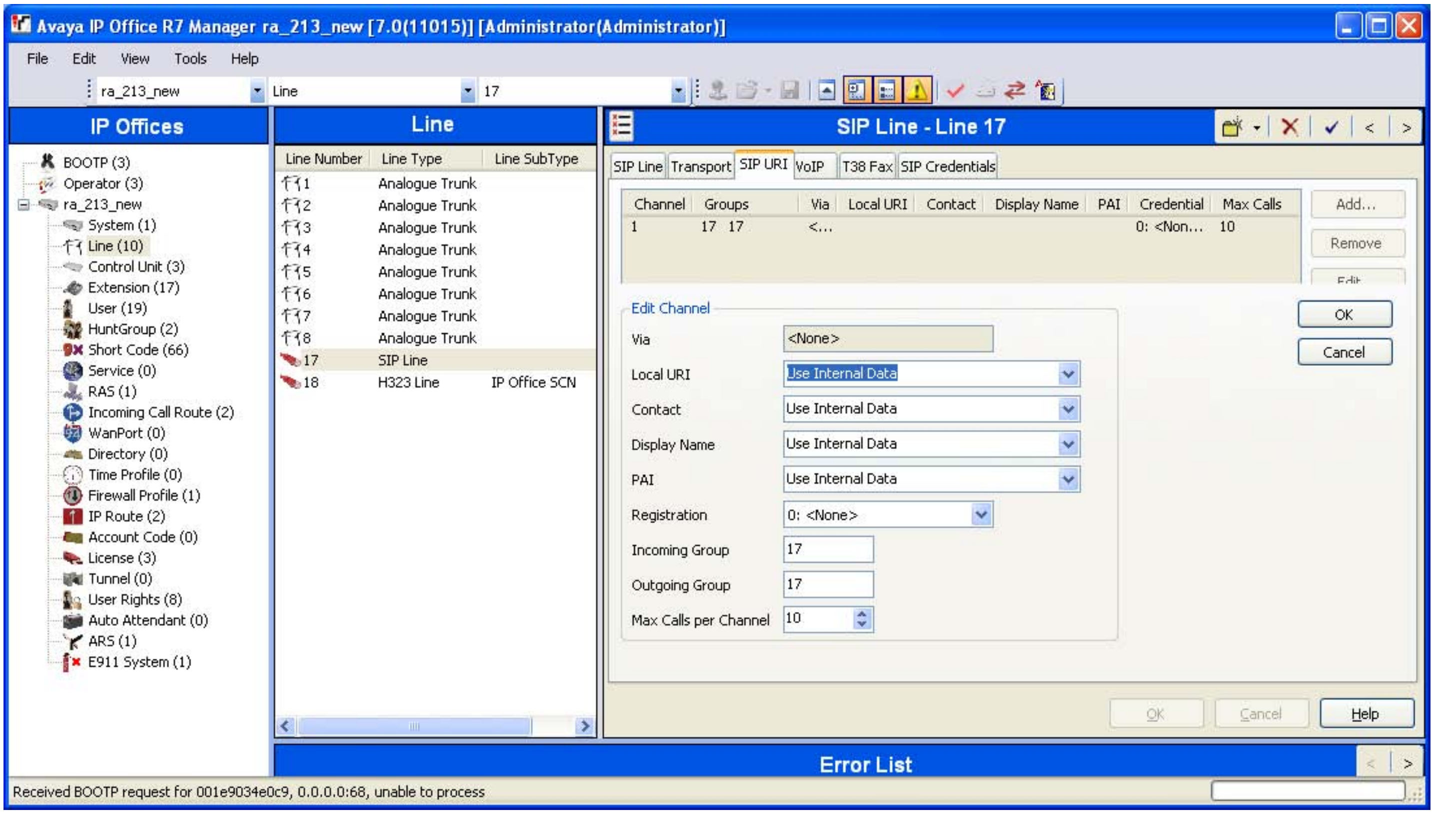


The ITSP Proxy Address would be, in this case, the IP address of the BCM that this route serves.

The BCM does not serve as a registrar, so there will be no registration credentials configured in the IP Office, and the Calls Route Via Registrar checkbox is irrelevant.

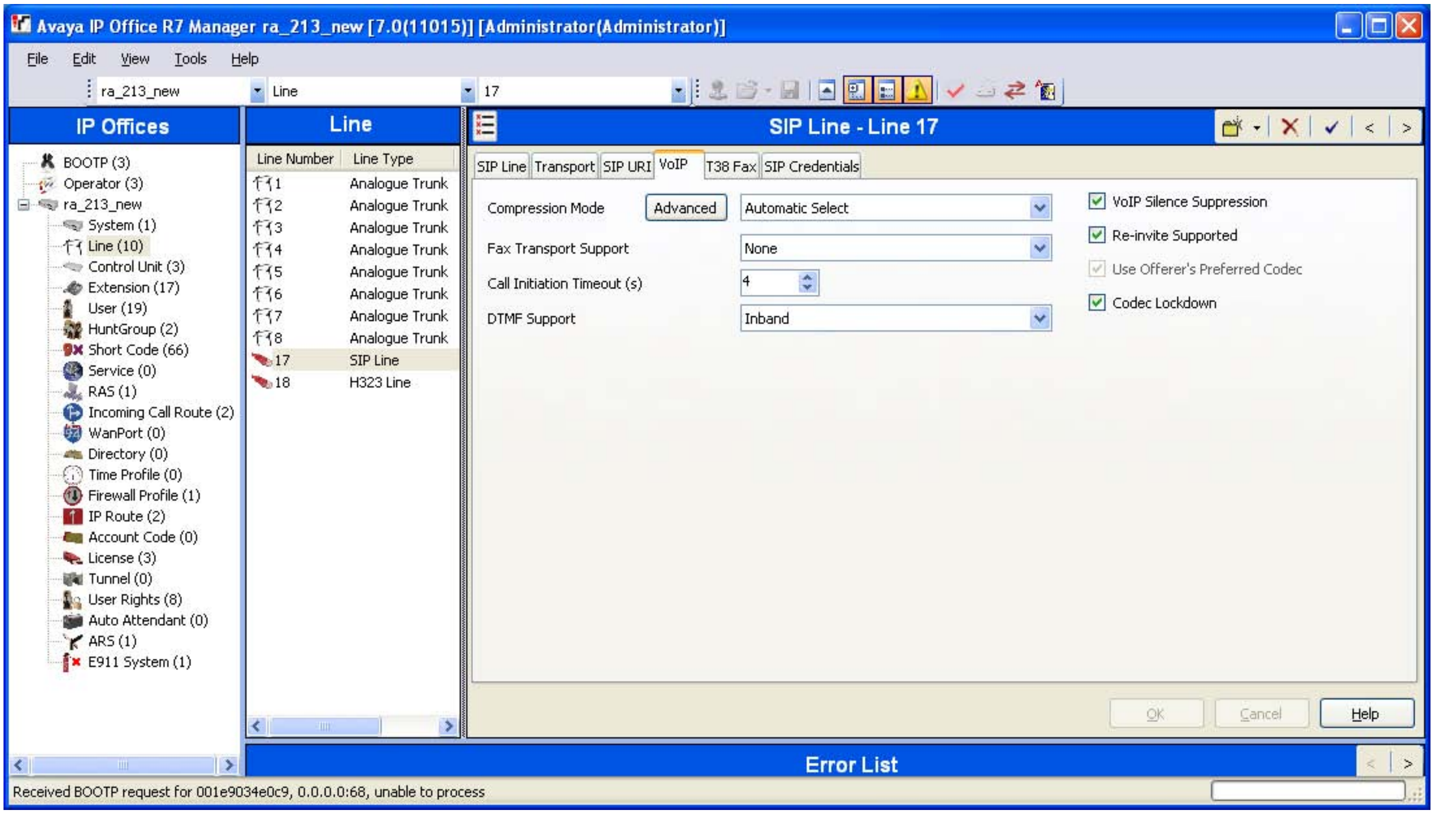
The Use Network Topology Info may cause problems if set to LAN1.

2.3.4 IP Office SIP Line SIP URI



The Incoming and Outgoing Groups are referenced in the Incoming Call Route section, and the Short Code for dialing.

2.3.5 IP Office SIP Line VoIP



Coordinate the VoIP Silence Suppression setting with the BCM's Voice Activity Detection setting. Both endpoints must have this enabled for it to be used by whichever codecs support it, e.g. G.723 and G.729.

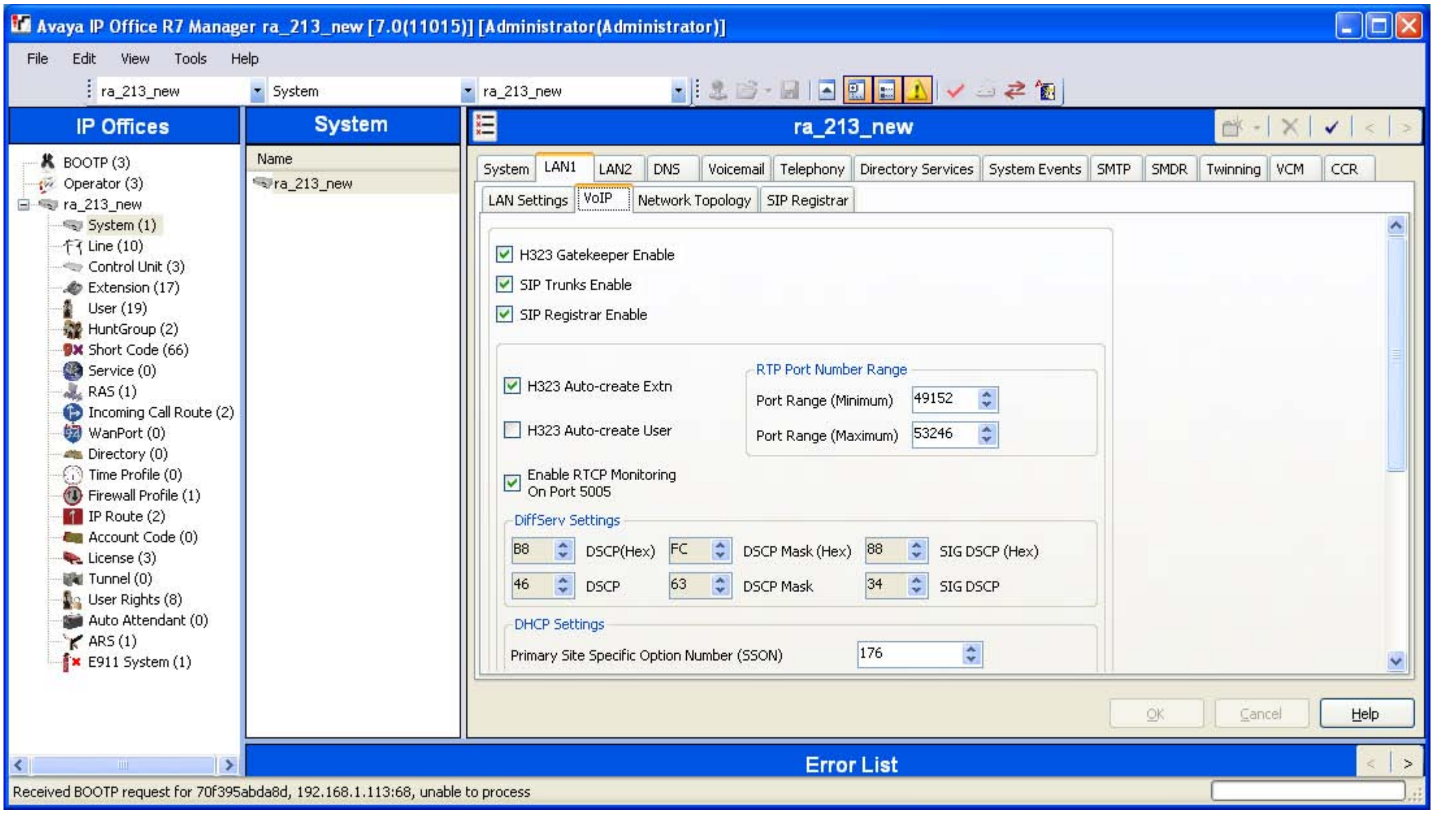
The codec preference order for IP Office is under the Compression Mode Advanced button.

Re-Invite Supported, as BCM will re-invite to put calls on hold, and transfer.

Use Offerer's Preferred Codec should be checked so as to more closely follow the RFC3264 OFFER/ANSWER specification.

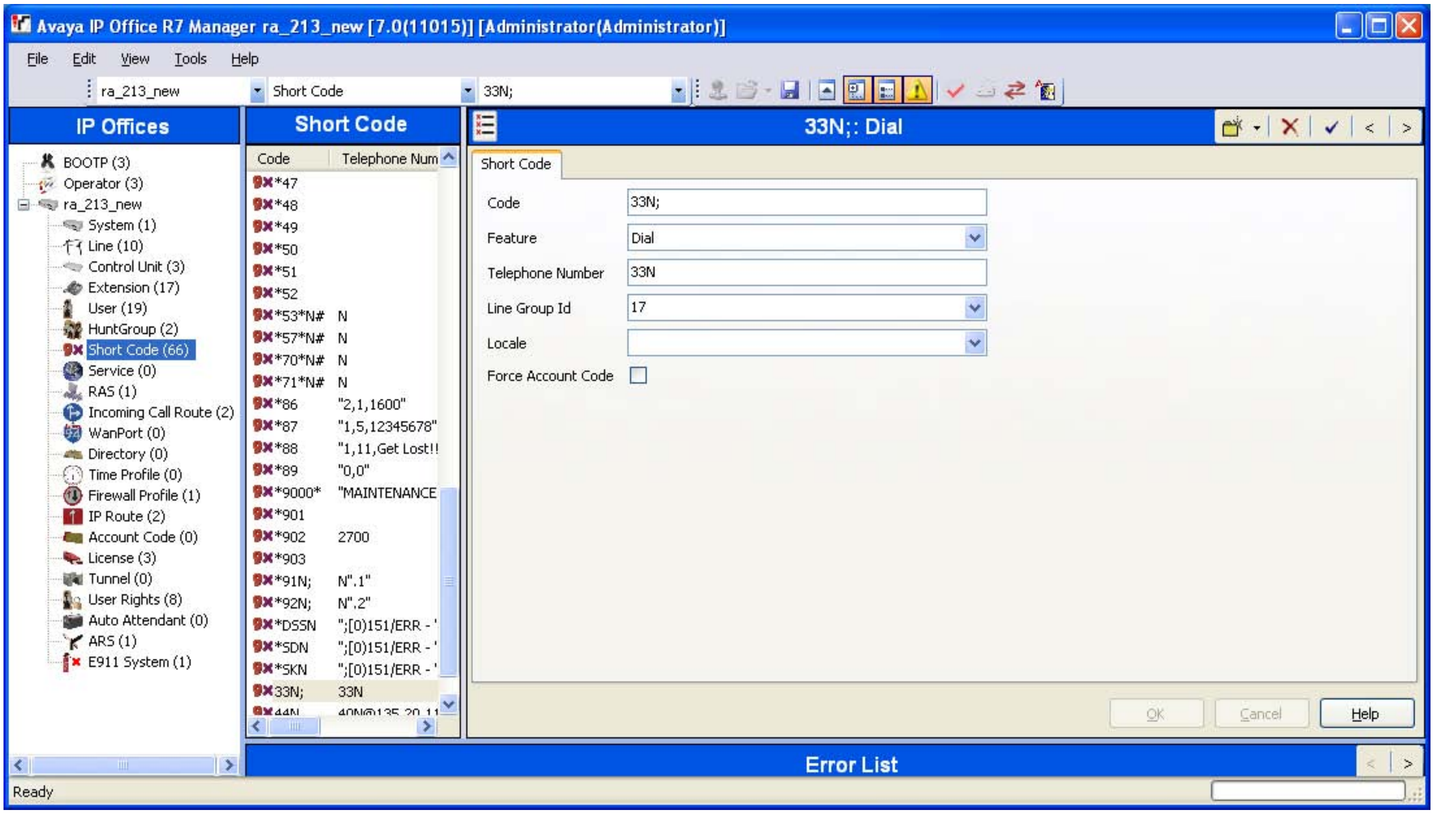
Codec Lockdown is not a requirement for BCM as it always chooses a single codec from the offered list, but this setting does no harm.

2.3.6 IP Office LAN1 VoIP



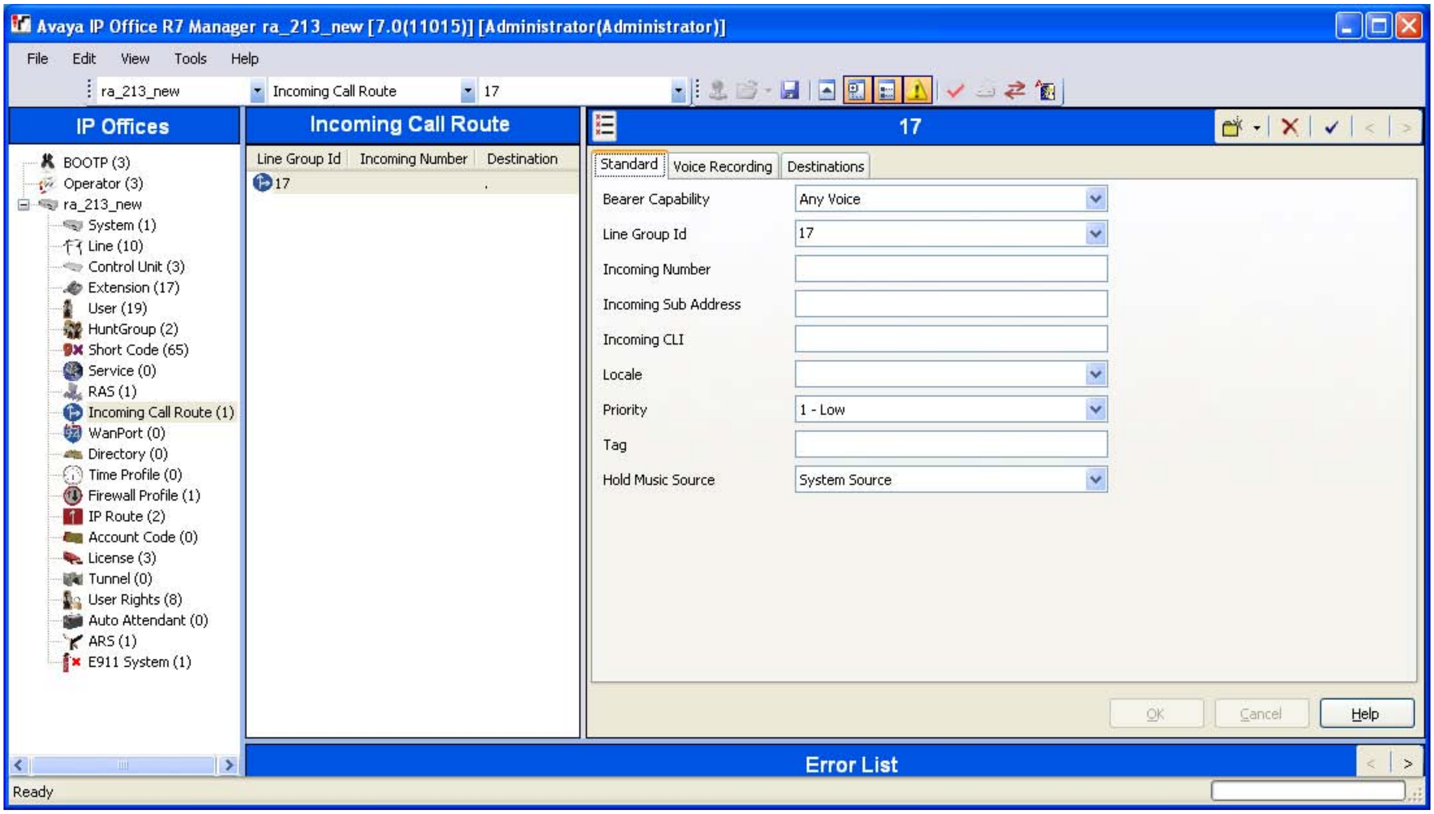
Ensure that SIP Trunks are enabled.

2.3.7 IP Office Short Code



The short code with the dial action matches any dial string that begins with ‘33’, waits for the rest of the dial string ‘N’ until the inter-digit timeout ‘;’, the sends the ‘33’ plus the rest of the dialed number ‘N’ to Line Group Id ‘17’ which was the Outgoing Group Id specified on the SIP Line // SIP URI tab.

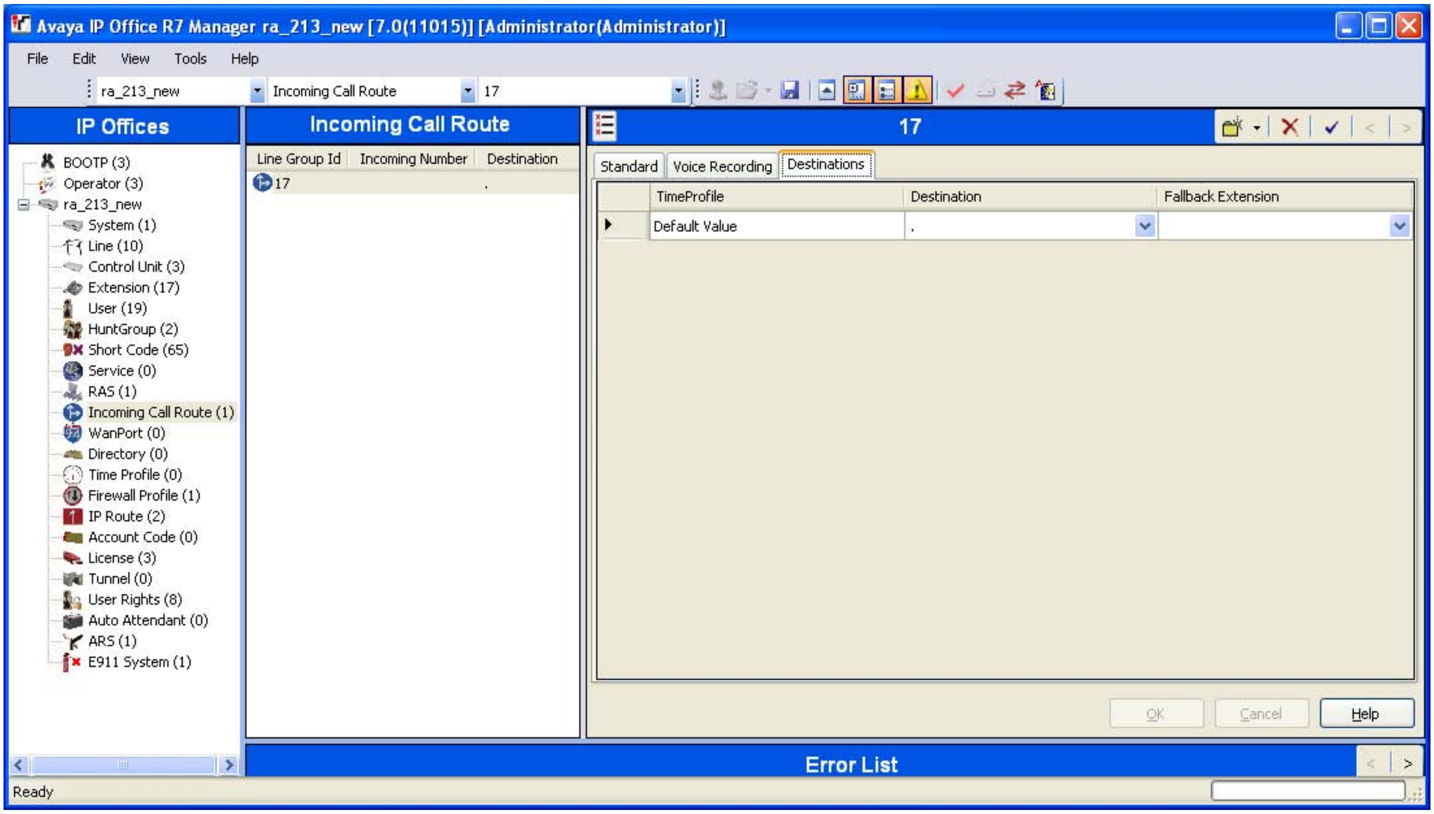
2.3.8 IP Office Incoming Call Route



Any number coming in from Group Id 17 (the Incoming Group Id from the SIP Line // SIP URI tab) will get routed to itself '.'

This destination is configured on the Destinations tab.

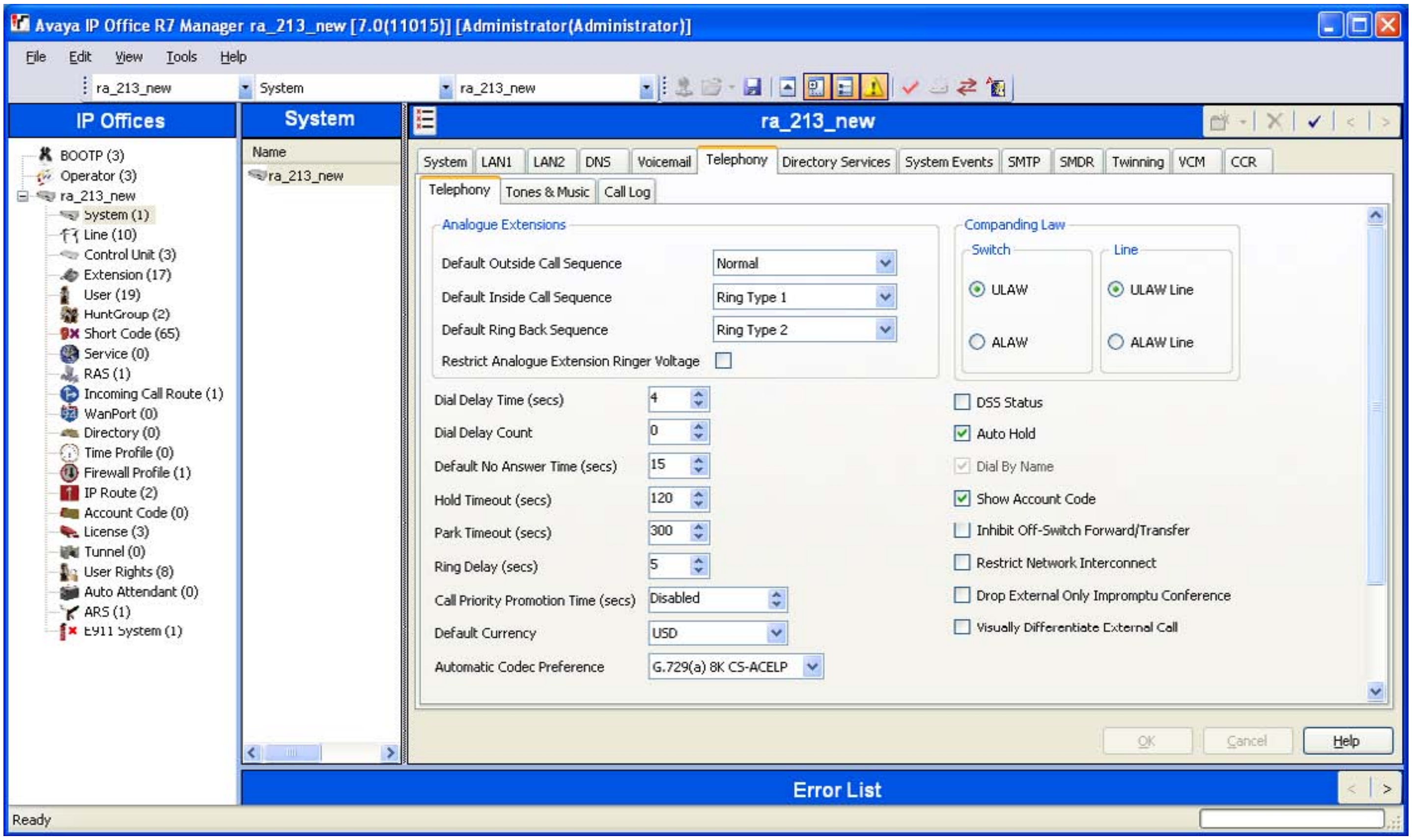
2.3.9 IP Office Incoming Call Route Destination



The addition of the "." will look for an incoming digit match, User or Hunt Group. The drop down list can be used to associate an incoming call route to a specific user. The "." must be manually input, it is not part of the drop down list.

The Line Group ID relates to the ‘Incoming Group’ on the SIP Line SIP URI tab.

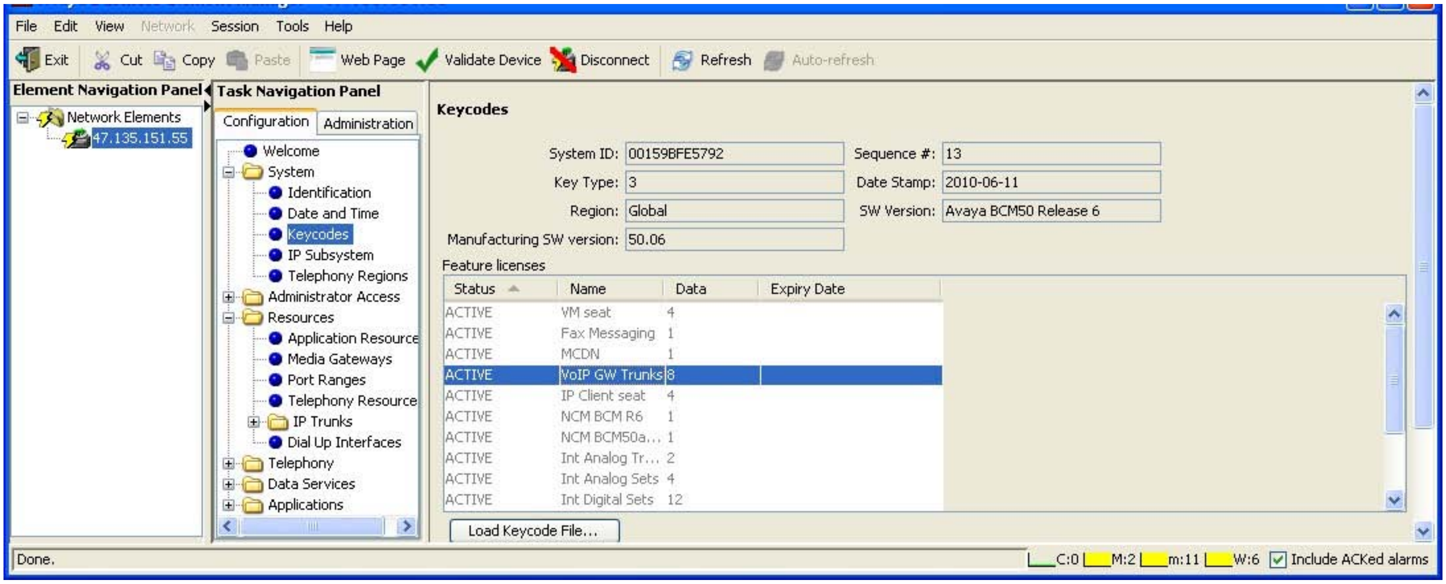
2.3.10 IP Office System Telephony



Choose the appropriate Companding Law depending on region, and the preferred Automatic Codec Preference.

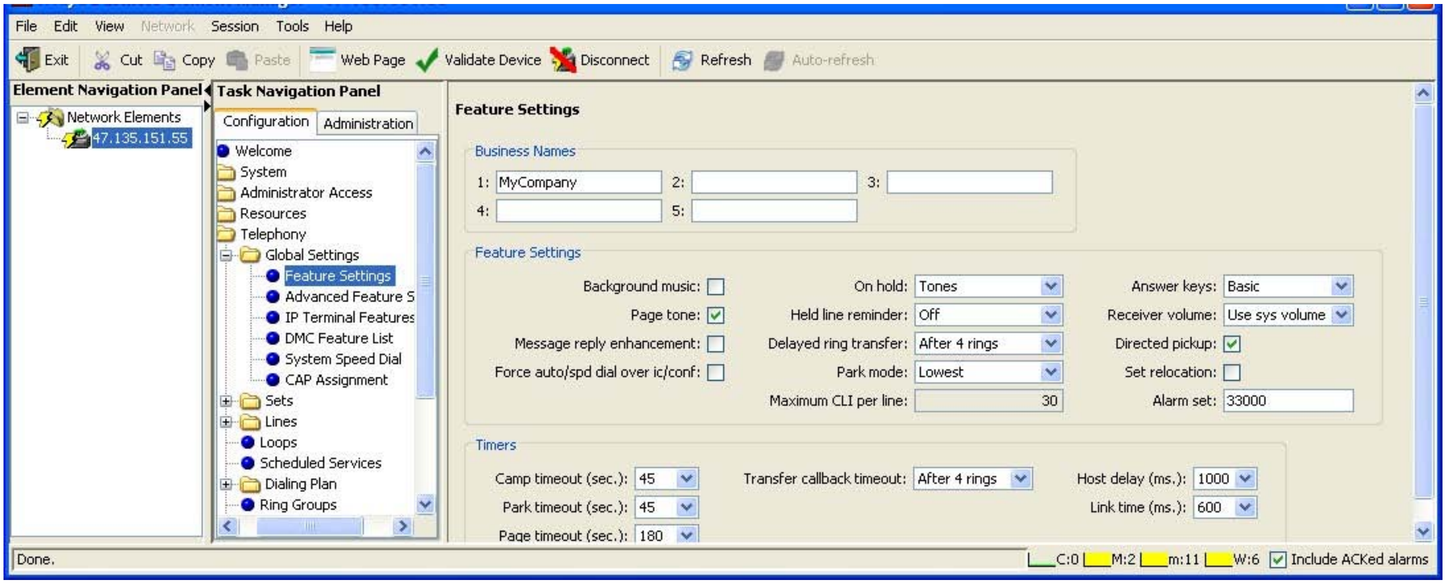
2.4 Provisioning SIP trunks on BCM

2.4.1 BCM SIP Trunk Keycodes



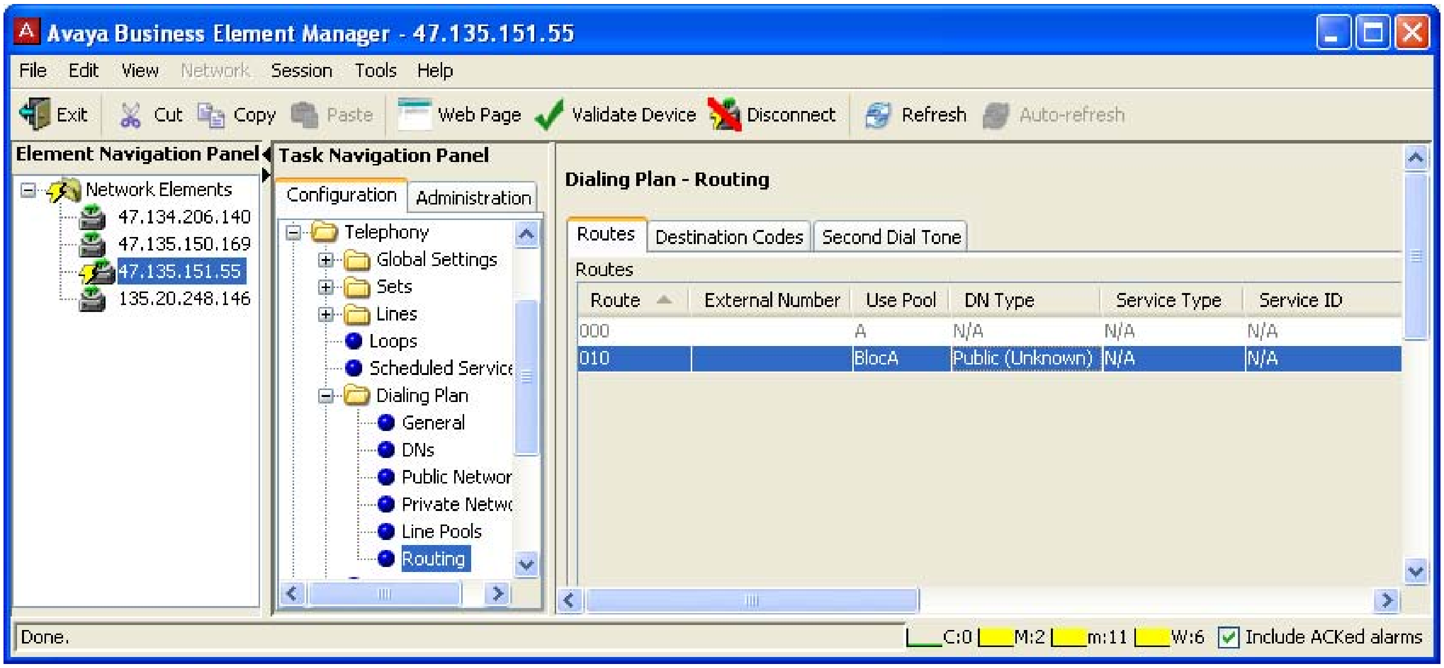
Keycodes are required in order to activate IP trunks on BCM. The ‘VoIP GW Trunks’ license enables H.323 and SIP trunks. There is, alternatively, a separate keycode to enable only SIP trunks.

2.4.2 BCM Business Name



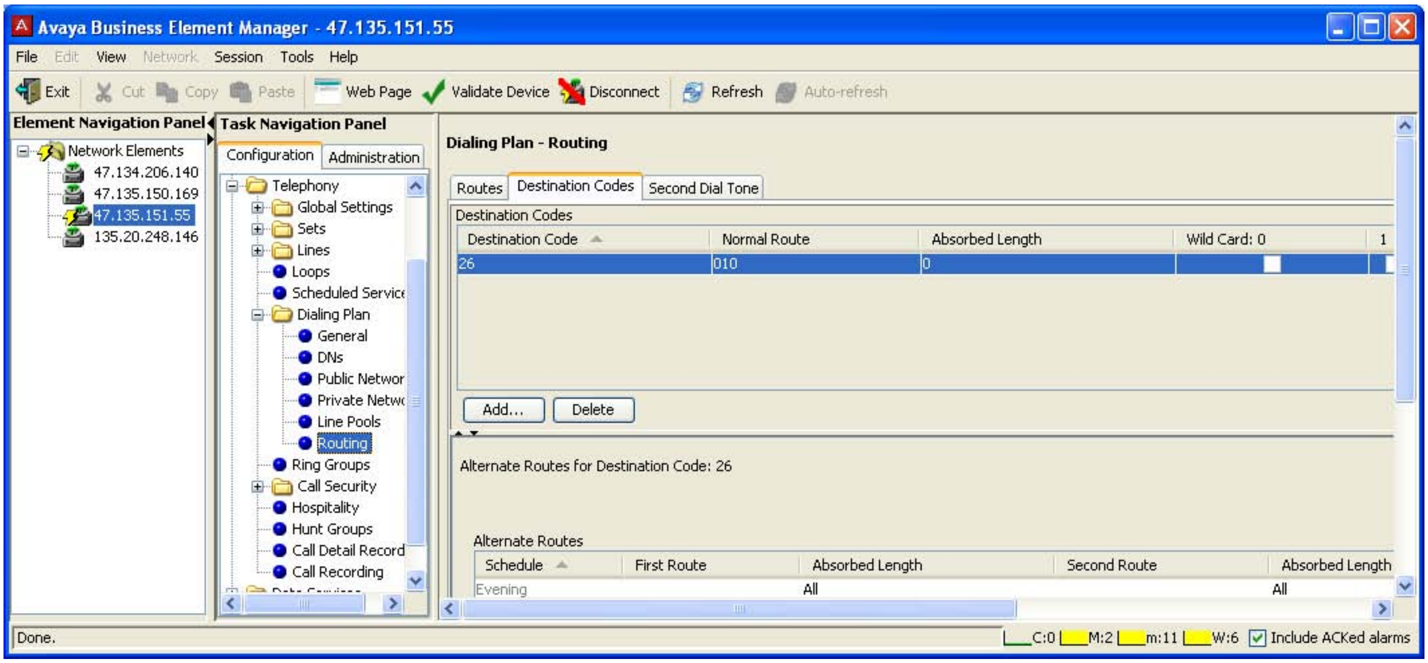
Before BCM will send CLID, a business name must be configured for the system.

2.4.3 BCM Trunk Routes



There IP Office does not make use of the Phone-context in SIP messages to determine NPI/TON, so the DN Type field will be set to Public(Unknown), and will be routed using a public SIP Trunk profile.

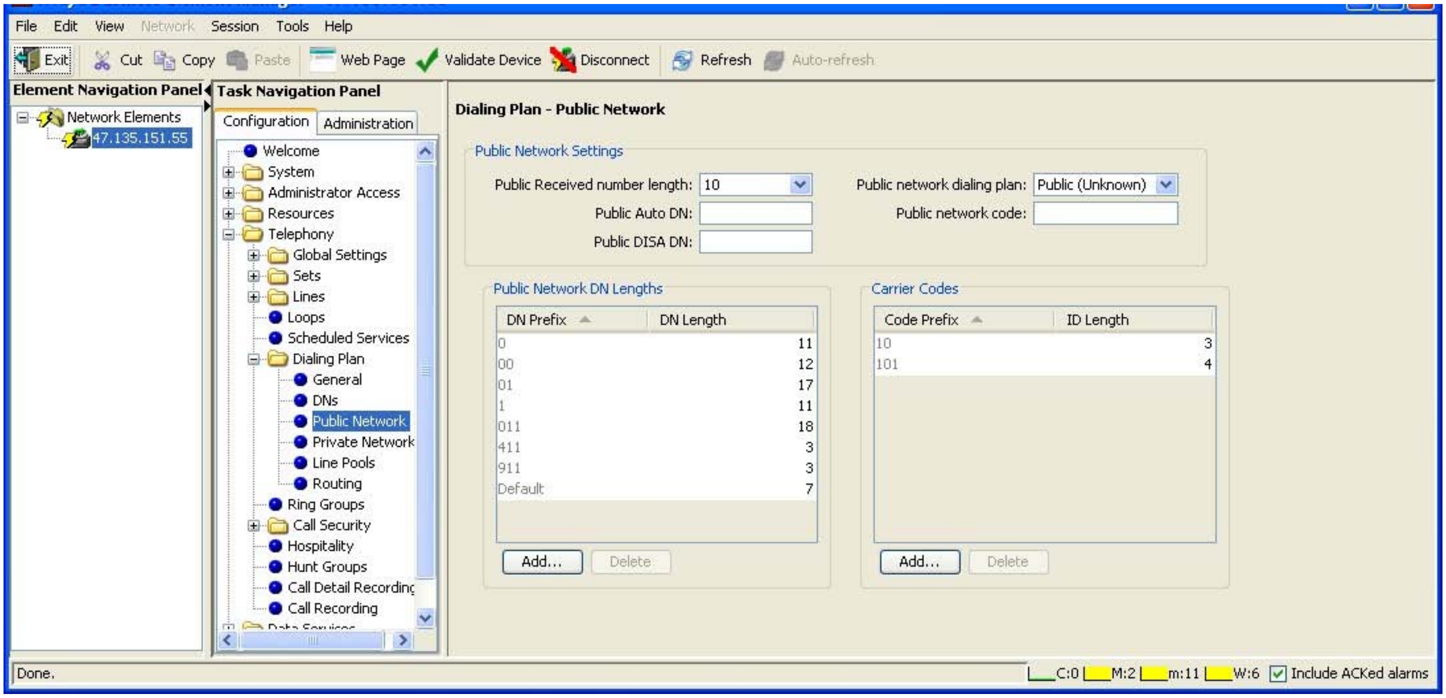
2.4.4 BCM Destination Codes



The leading digits are not stripped off the dialed string as they will be used by the IP Office to route the call to the destination Set or Hunt Group.

There will also be a corresponding entry for this (‘26’) in the SIP trunk routing table.

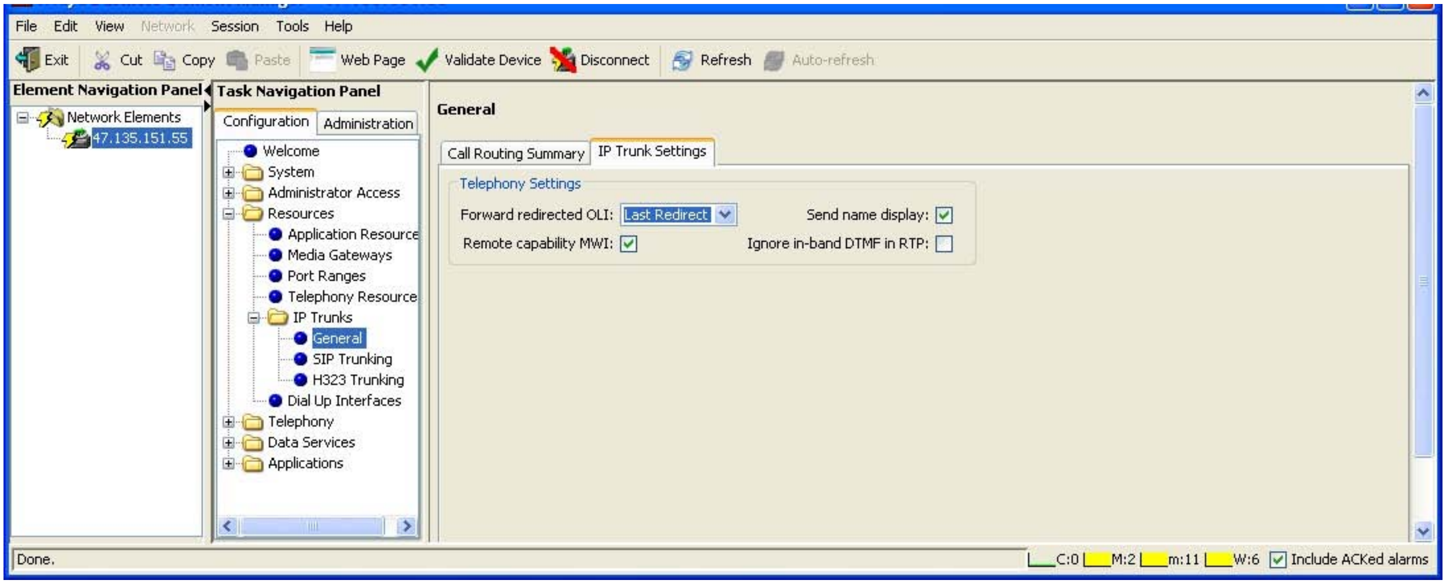
2.4.5 BCM Public Networking



The received number length has been set to 10, assuming a full NPA/NXX/Extn North American dialing plan will be used.

On the outgoing side, the public network dialing plan type has been set to Public (Unknown) as private trunking is not supported.

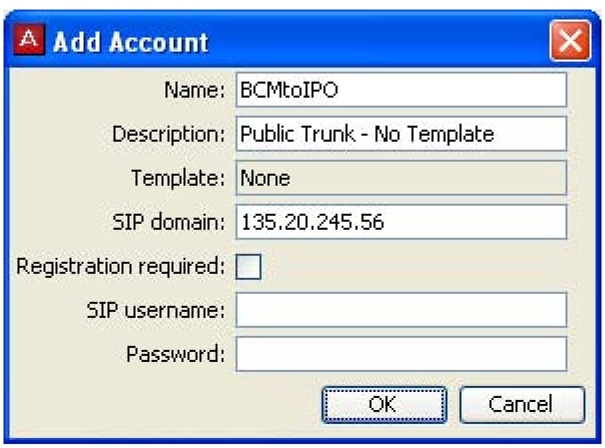
2.4.6 BCM SIP Trunks General Settings



Set the 'Forward redirected OLI' to Last or First Redirect depending on the expected treatment from voicemail.

The 'Remote capability MWI' indicates that Message Waiting indications will be sent across the SIP trunk if there is a message for a set on the remote switch, though centralized voicemail is not supported between IP Office and BCM.

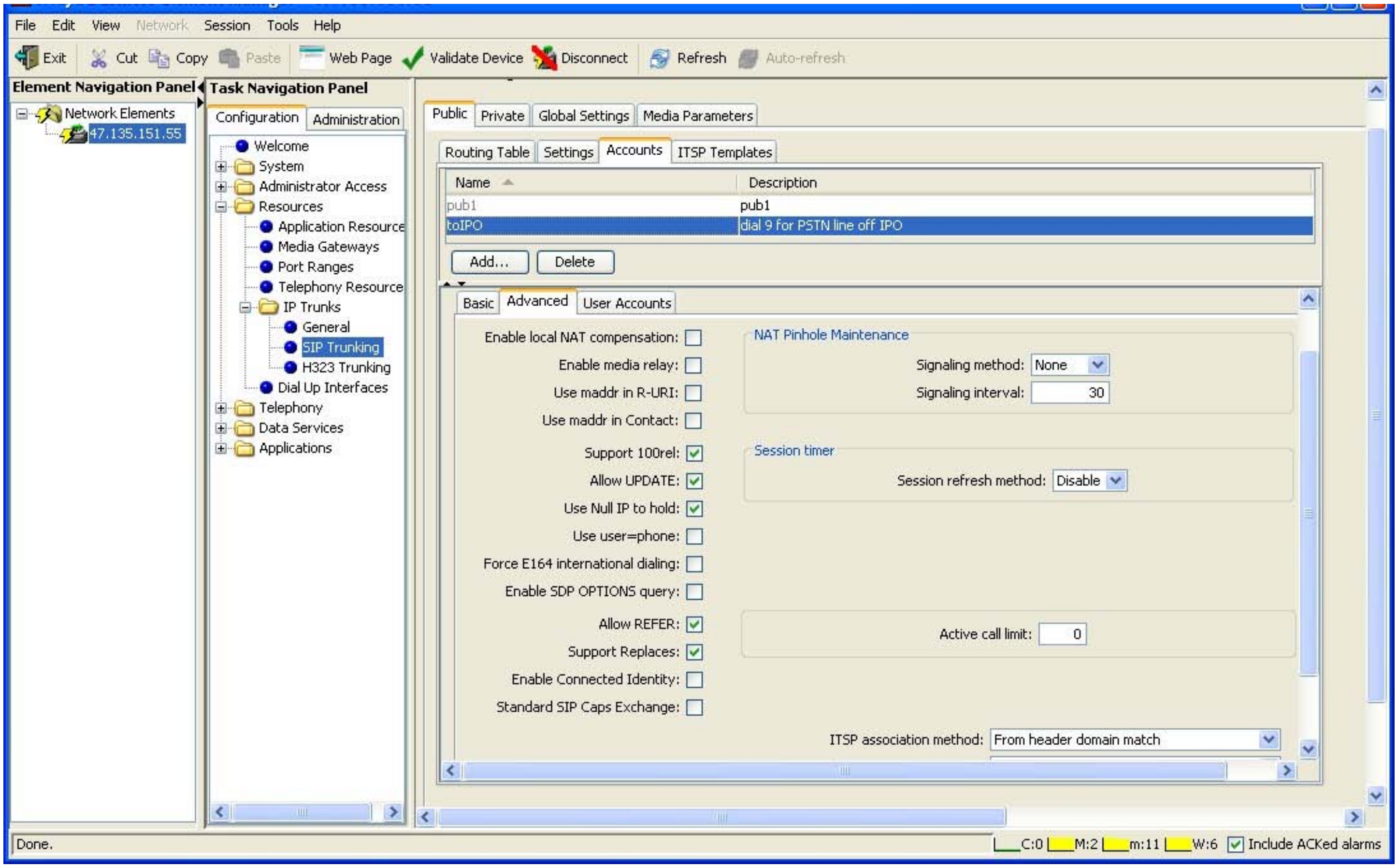
2.4.7 BCM SIP Trunks Public Account Basic



Create a public account for this network.

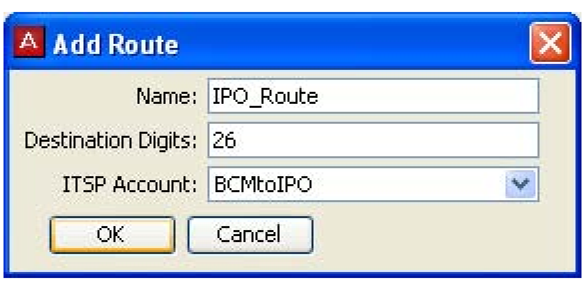
Enter the IP Address of the IP Office in the SIP Domain field.

2.4.8 BCM SIP Trunks Public Account Advanced



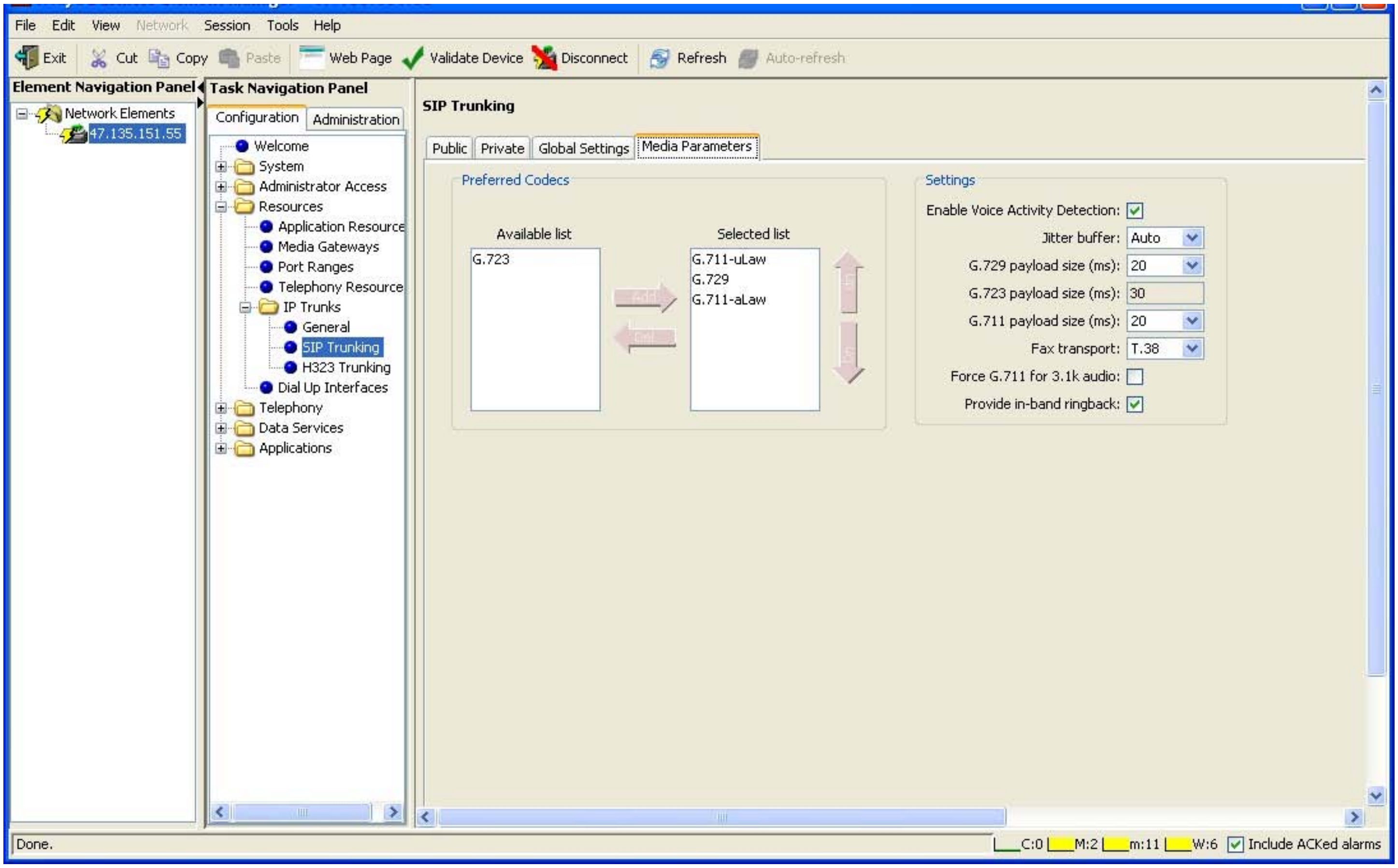
The default settings on this page, assuming the Account was not created based on a template, should be correct for successful interoperability with IP Office.

2.4.9 BCM SIP Trunks Public Routing Table



Add a route matching the '26' dialed digits from the Telephony // Dialing Plan // Routes // Destination Digits tab, and specifying the account created above.

2.4.10 BCM SIP Trunks Media Parameters



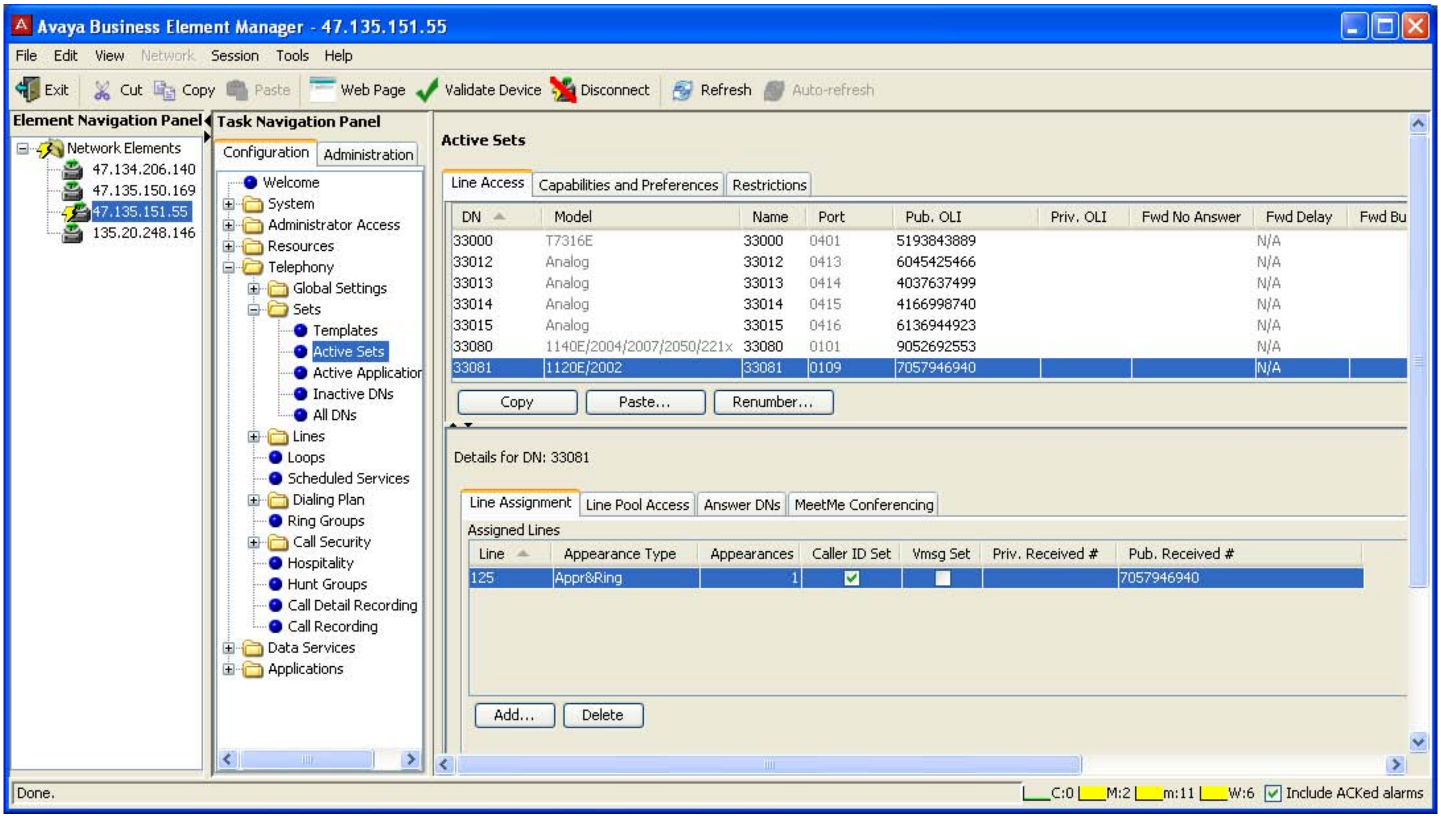
Choose the codecs and put them in the preferred order.

Select whether Voice Activity Detection (Silence Suppression) will be available for those codecs which support it.

Choose whether T.38 will be preferred for fax transport.

Choose whether in-band ringback should always be provided by the BCM on incoming SIP calls.

2.4.11 BCM Set Line Assignments

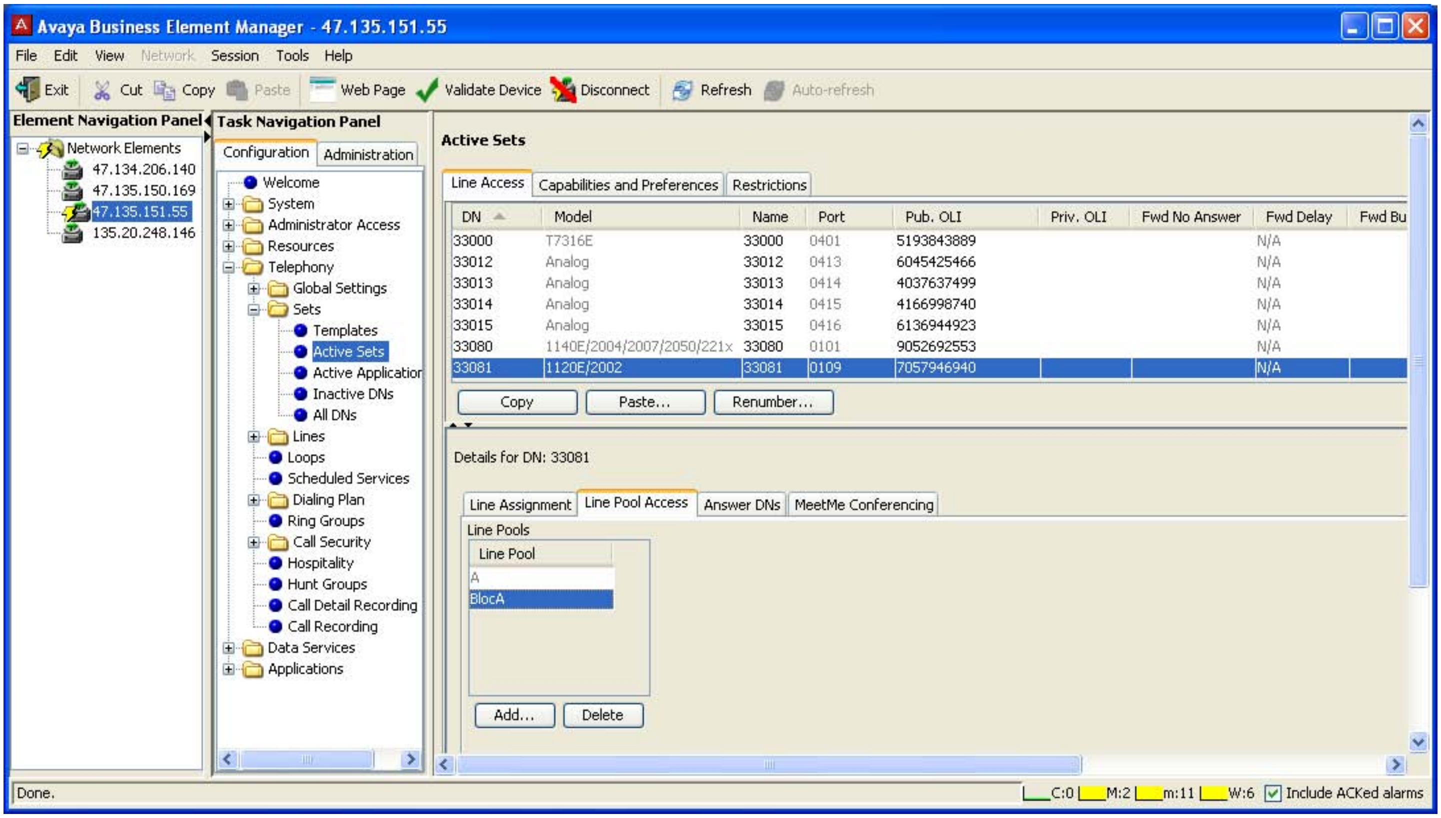


Calls into BCM are routed to Target lines. If the BCM had been set up with the DID configuration, it would have had a target line assigned to each set, and populated the sets' intercom numbers in the Received # fields. A BCM in 'PBX" profile will have to have these assigned and configured manually.

Calls with public, or without numbering plan information will be matched to the digits in the Public Received #.

Configure the Public OLI for each set to be used as its CLID.

2.4.12 BCM Set Line Pool Access



By default, SIP lines/trunks are put into pool BlocA on BCM. Sets do not automatically have access to this pool, so it must be added on a set by set basis.

3.0 References

IP Office 7.0 Installation Manual, Document Number 15-601042
https://support.avaya.com/css/P8/documents/100119958

IP Office Release 7.0 Manager 9.0, Document Number 15-601011
https://support.avaya.com/css/P8/documents/100119917

IP Office Release 6.0 System Status Application, Issue 05a, February 12, 2010
Document Number 15-601758
http://support.avaya.com/css/P8/documents/100073300

IP Office System Monitor, Document Number 15-601019
http://support.avaya.com/css/P8/documents/100073350