Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Metaswitch MetaSphere CFS and Avaya IP Office – Issue 1.0

Abstract

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunking between a Metaswitch MetaSphere Call Feature Server (CFS) solution and an Avaya IP Office telephony solution.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunking between a Metaswitch MetaSphere Call Feature Server (CFS) solution and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital and analog endpoints.

SIP (Session Initiation Protocol) is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 is the primary specification governing this protocol. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between Avaya IP Office and the Metaswitch MetaSphere CFS solution.

MetaSphere is a broad suite of telephony applications. MetaSphere applications may be deployed individually or in combination to deliver the full spectrum of legacy and next-generation voice services.

MetaSphere makes it easy to deliver the following services:

- POTS - Service equivalence to legacy switches for practically all local calling features in use today.
- Consumer VoIP - A highly flexible solution that enables the creation of compelling consumer VoIP packages by combining a wide range of traditional and next-generation local calling features.
- Hosted PBX - Business voice applications that meet the needs of small, medium and large enterprises, enabling customers to present compelling alternatives to key systems and PBXs.
- Business IP Trunking - A powerful and flexible solution to address the rapidly growing market for IP-PBX connectivity via native SIP trunks.
- Tandem and Long Distance - Meets the needs of local, access and long distance tandem switching, and provide a complete solution for updating legacy core voice networks to VoIP.
- Voicemail / Unified Messaging - The richest possible set of voicemail and unified messaging capabilities, maximizing convenience and ease-of-use across multiple devices and access methods.

1.1. Interoperability Compliance Testing

A simulated enterprise site consisting of an Avaya IP Office telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the generally available SIP trunking solution provided by Metaswitch. This allowed the enterprise site to use SIP trunking for calls to and from the PSTN.

The following features and functionality were covered during the SIP trunking interoperability compliance testing:

- Incoming calls to the enterprise site from the PSTN (using the DID numbers assigned by Metaswitch).
- Outgoing calls from the enterprise site to PSTN destinations via Metaswitch.
- Calls using various H.323, digital, and analog endpoints supported by Avaya IP Office.
- Various call types including: local, long distance, and toll free calls.
- Calls using G.711 µLAW, G.711ALAW, and G.729(a) codecs.
- Inbound and outbound fax calls.
- DTMF tone transmission using RFC 2833 with voice mail navigation.
- Telephone features such as hold, transfer, conference, and call forwarding.
- Mobility Features: Mobile twinning to a mobile phone.
- Calls using IP Office Softphone.

1.2. Support

For technical support for Metaswitch, contact your Metaswitch Networks support representative.

2. Reference Configuration

Figure 1 illustrates an example Avaya IP telephony solution connected to Metaswitch that was utilized for compliance testing. Since public IP addresses were used during compliance testing, those IP address are not show in the figure below and they are masked (at least partially) throughout the document.

The Avaya IP telephony solution comprised of the following equipment and was used to simulate a customer site:

- Avaya IP Office 500 with Phone Expansion Module
- Avaya 5610SW IP Telephone (H.323 protocol)
- Avaya 9620 IP Telephone (H.323 protocol)
- Avaya 2420 Digital Telephone
- Avaya 5420 Digital Telephone
- Avaya 6210 Analog Telephone

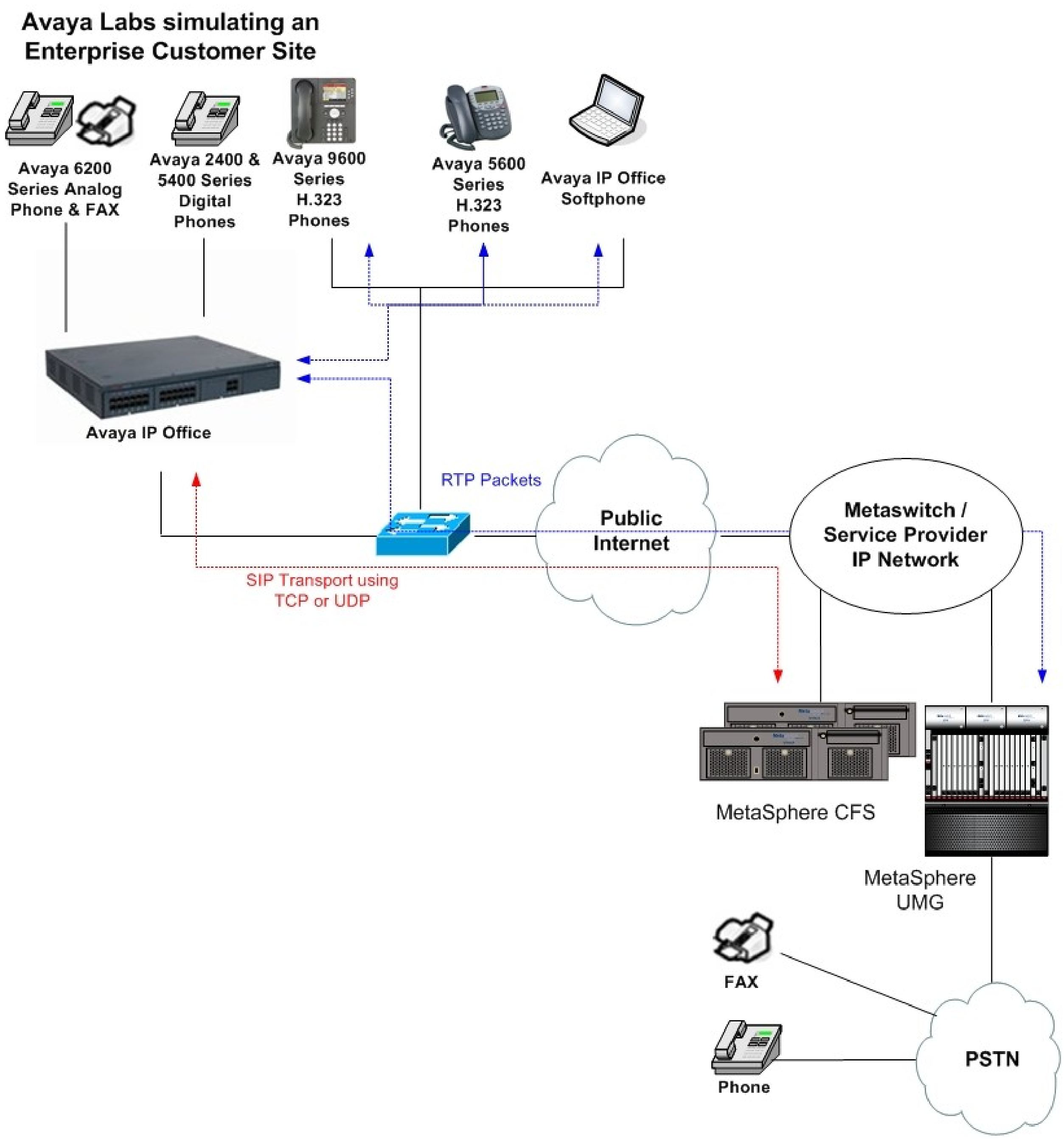


Figure 1: Avaya IP Telephony Network connected to Metaswitch

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya IP Telephony Solution Components	
Equipment	**Software/Firmware**
Avaya IP Office 500	6.0 (8)
Avaya IP Office 500 Phone Expansion Module Analog POTS 30 V2	6.0 (8)
Avaya IP Office Manager (Windows PC)	6.0 (8)
Avaya IP Office Voicemail Pro	6.0 (22)
Avaya 5610SW IP Telephone (H.323)	2.9.1
Avaya 9620 IP Telephone (H.323)	3.002
Avaya 2420 Digital Telephone	R6 Firmware
Avaya 5420 Digital Telephone	R6 Firmware
Avaya 6210 Analog Telephone	n/a
Avaya IP Office Softphone	3.0
Metaswitch Solution Components	
Equipment	**Software/Firmware**
Metaswitch MetaSphere CFS Metaswitch MetaSphere UMG	7.1.01 SU0

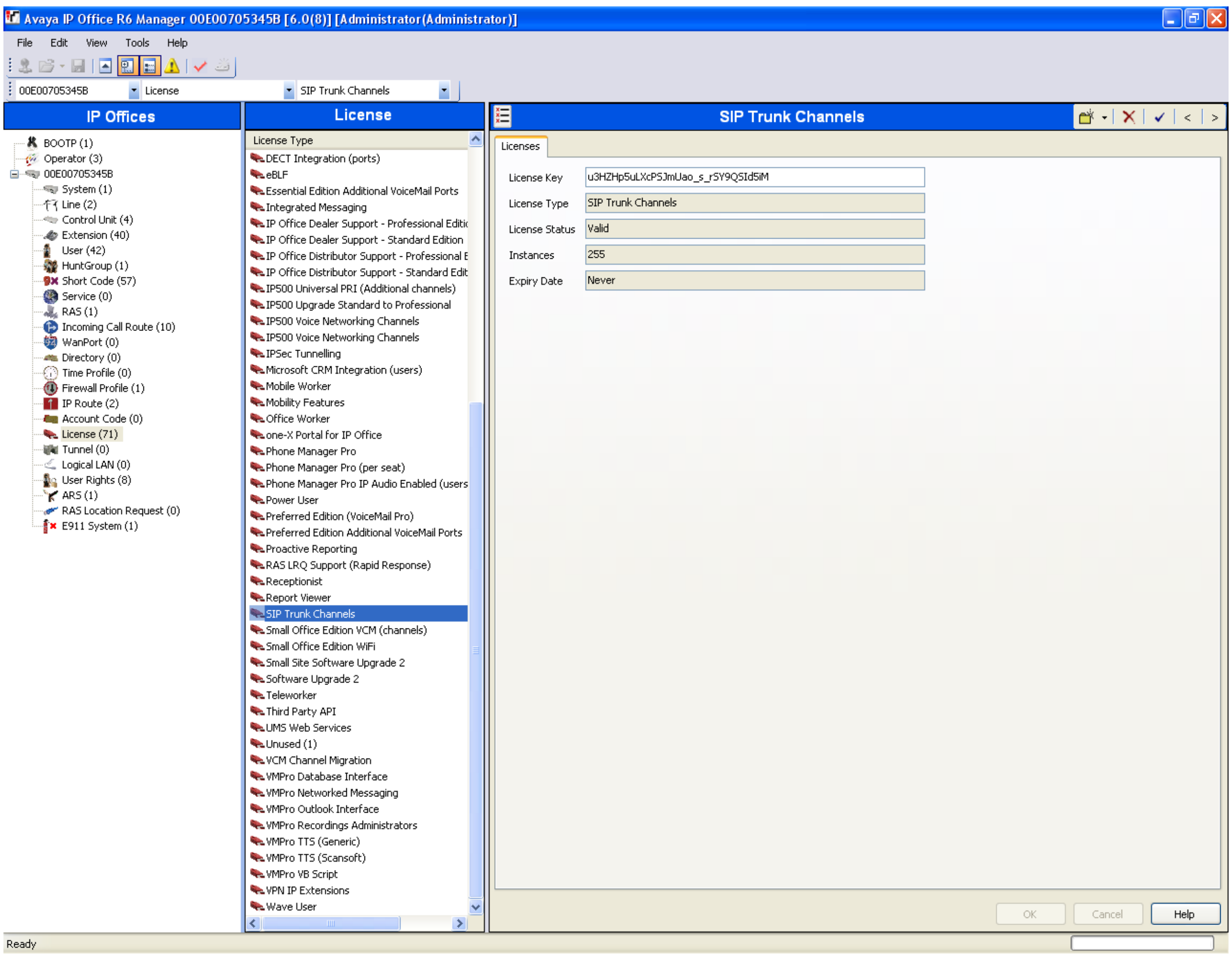
4. Configure the Avaya IP Office

This section describes the steps required for configuring a static SIP trunk on IP Office.

IP Office is configured via the IP Office Manager program. Log into the IP Office Manager PC and select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Log into the Manager application using the appropriate credentials.

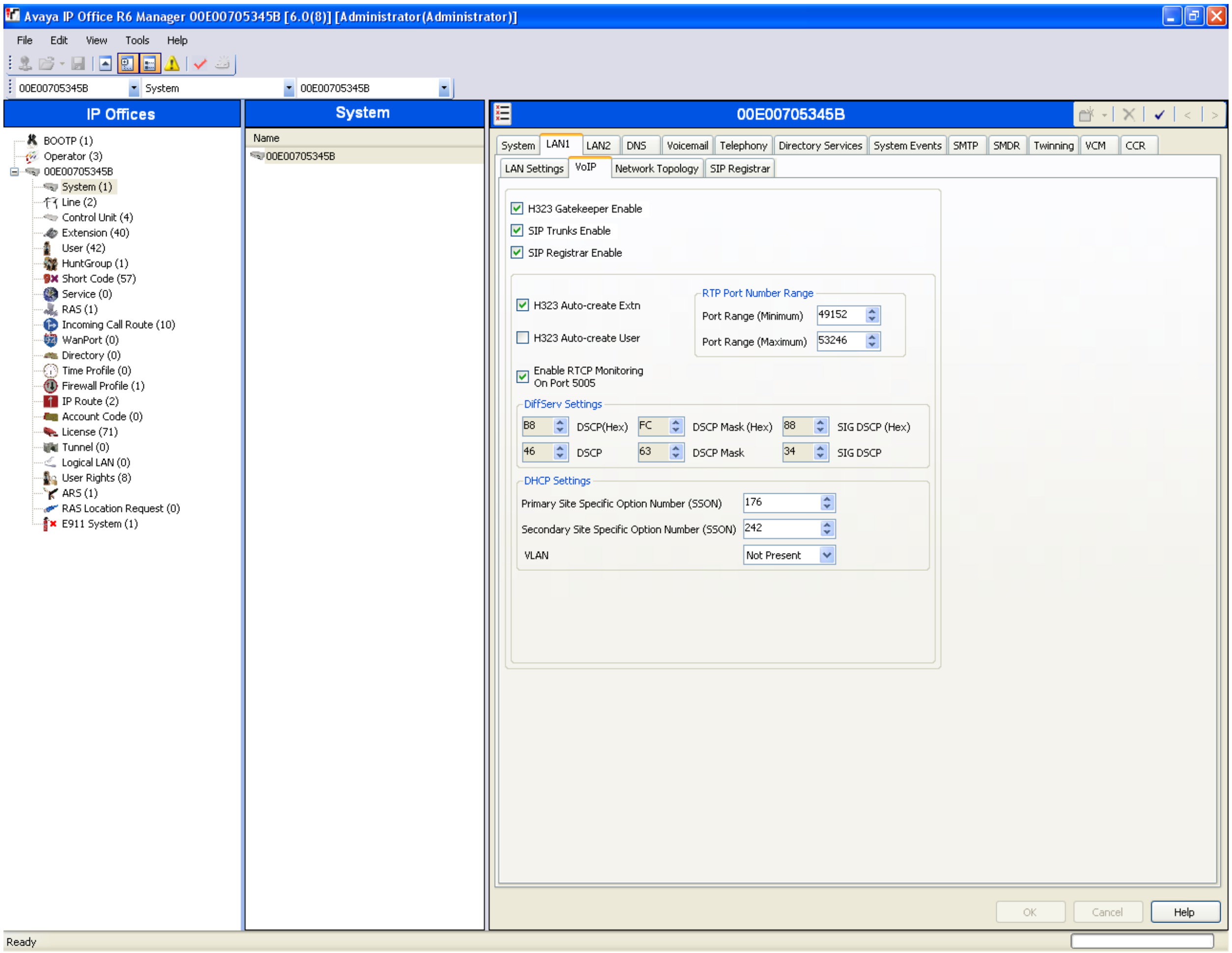
1. *Verify the SIP Trunk Channels License.*
   Click on **License** in the left panel. Confirm that there is a valid **SIP Trunk Channels** entry.

   If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



2. *Enable SIP Trunks*.

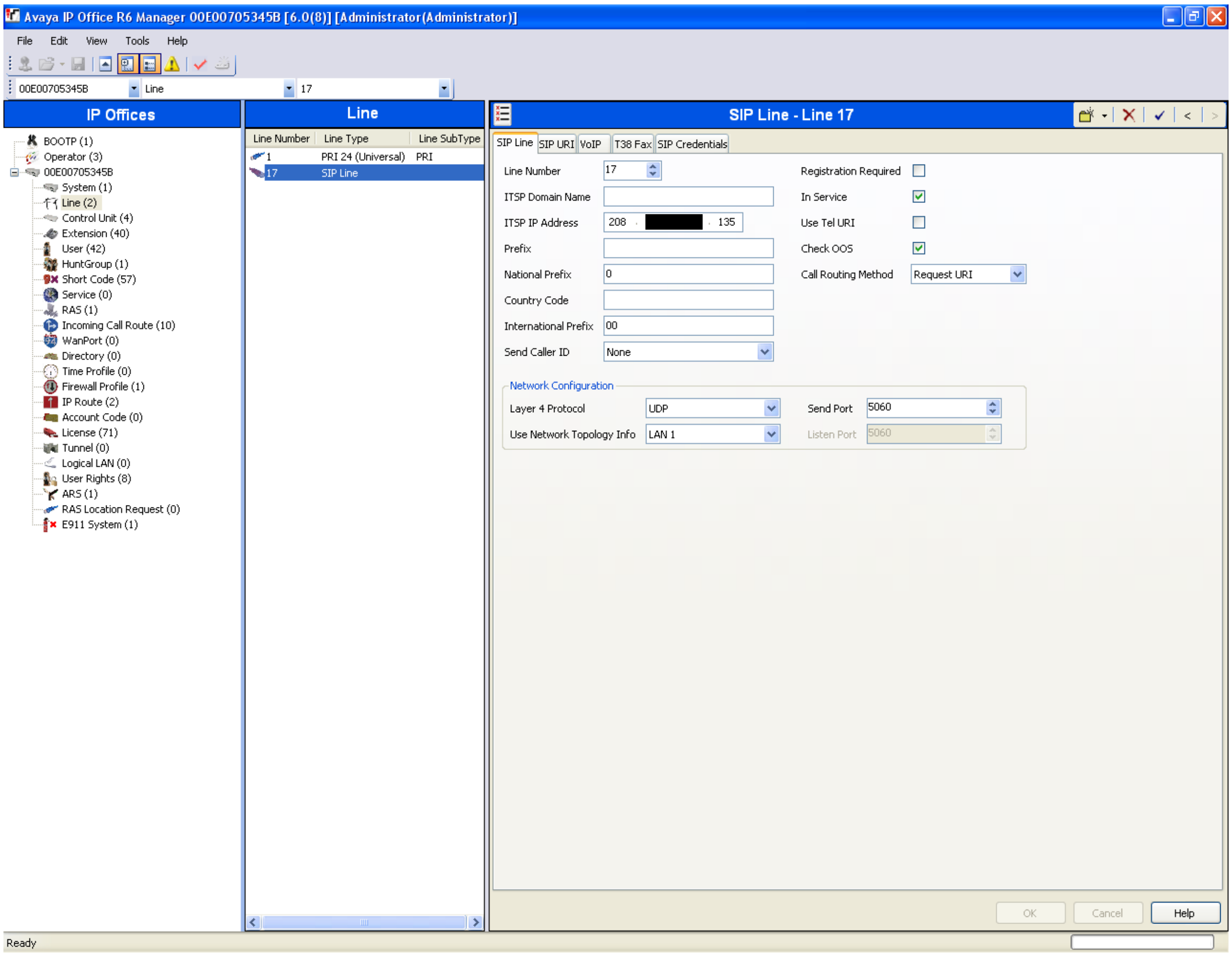
Select **System** in the left panel. Click the **LAN1** tab. Under the **LAN1** tab, select the **VoIP** tab**,** and check the **SIP Trunks Enable** box. Click the **OK** button.



3. *Create the static SIP line for Metaswitch.*
Select **Line** in the left panel. Right-click and select **New** → **SIP Line**.

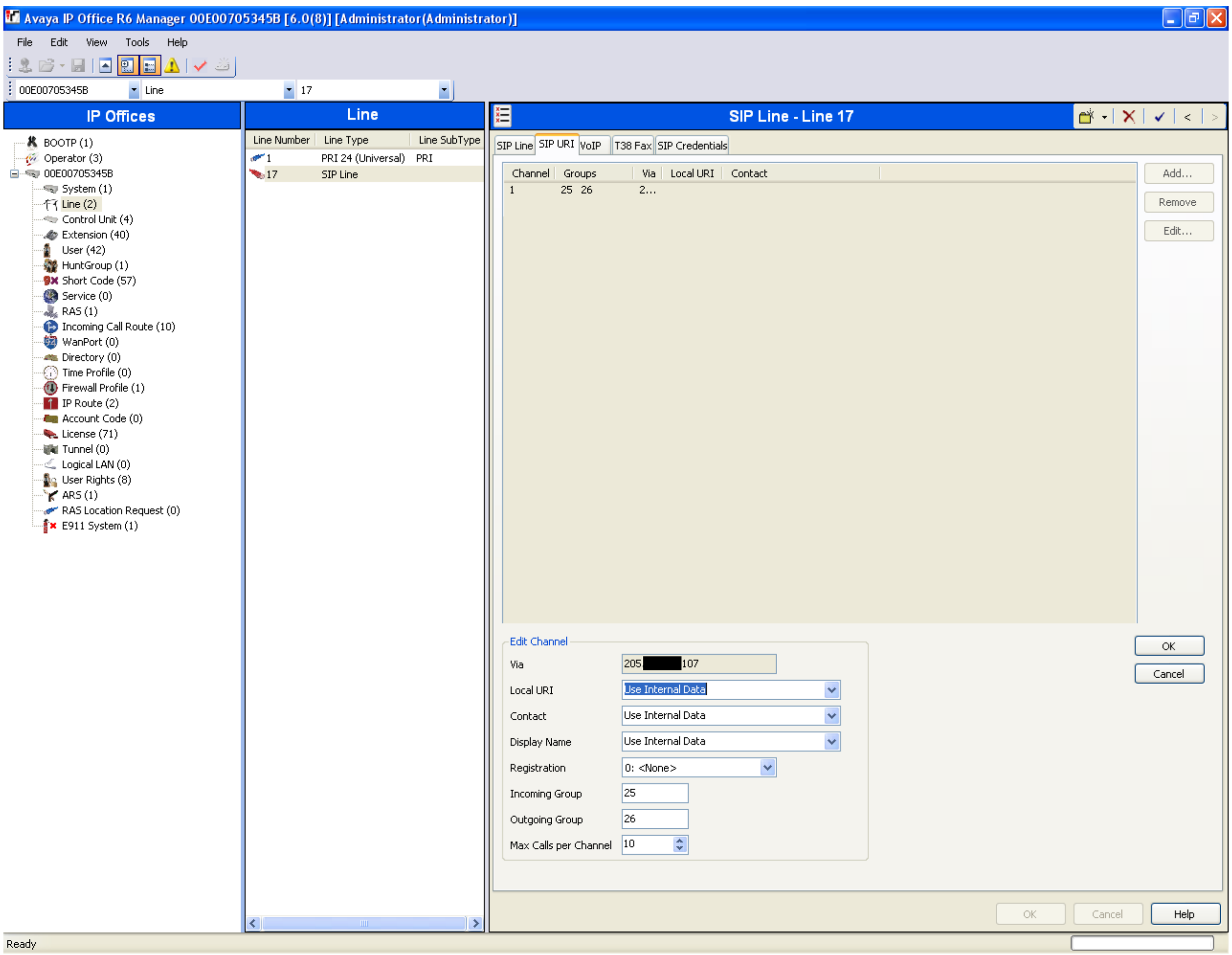
Configure the following:

- For the **ITSP IP Address** field, enter the IP address used for Metaswitch MetaSphere CFS solution.
- Use default values for other fields.



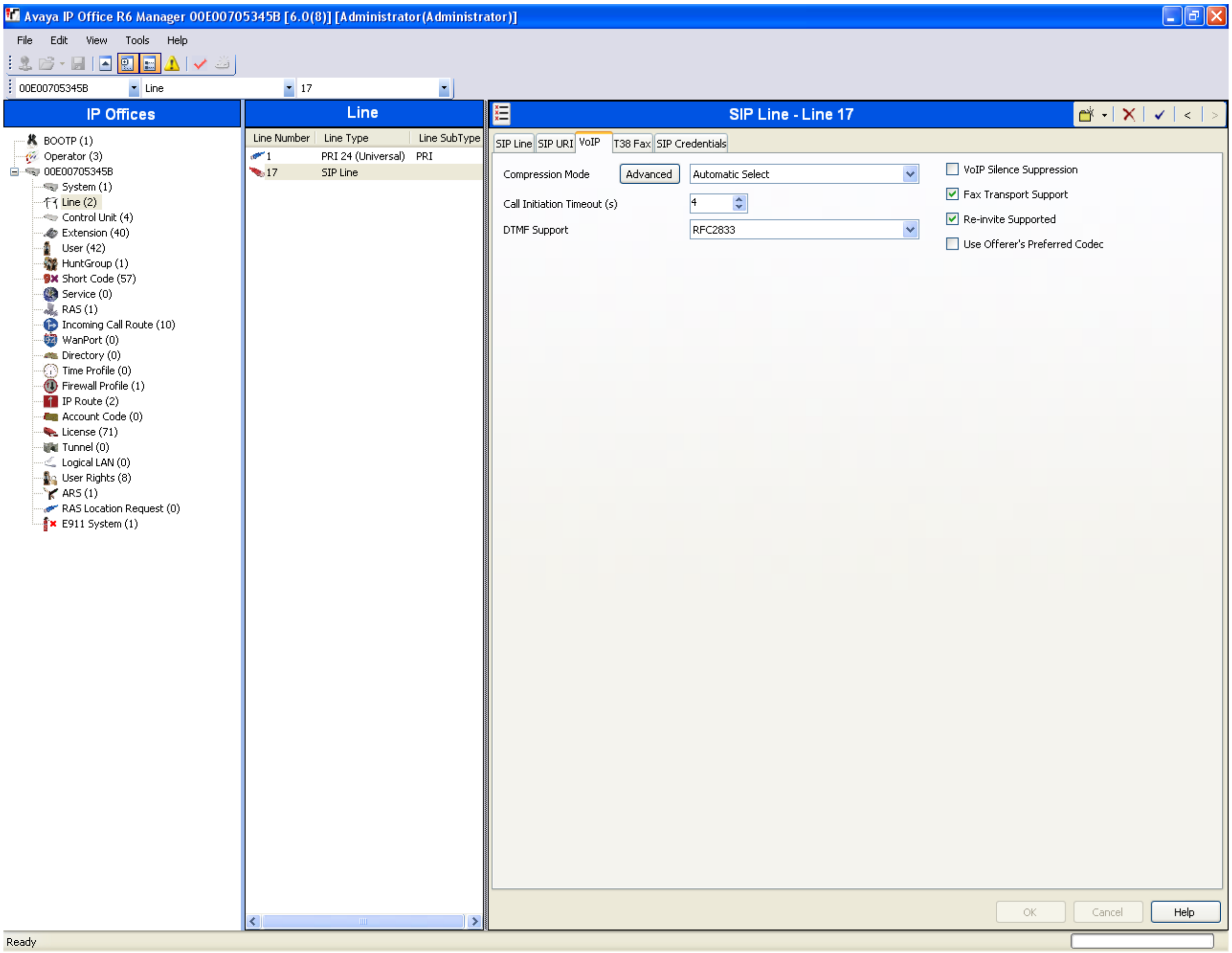
4. *Configure SIP URI parameters for the SIP Line.*
Select the **SIP URI** tab. Click the **Add** button.

Select **Use Internal Data** for the **Local URI, Contact**, and **Display Name** fields. This tells the system to use the information configured on the SIP tab for each individual user. Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. The **Incoming Group** field will be used for mapping inbound calls from the SIP trunk to local stations. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in Step 8. Use default values for all other fields. Click the **OK** button.



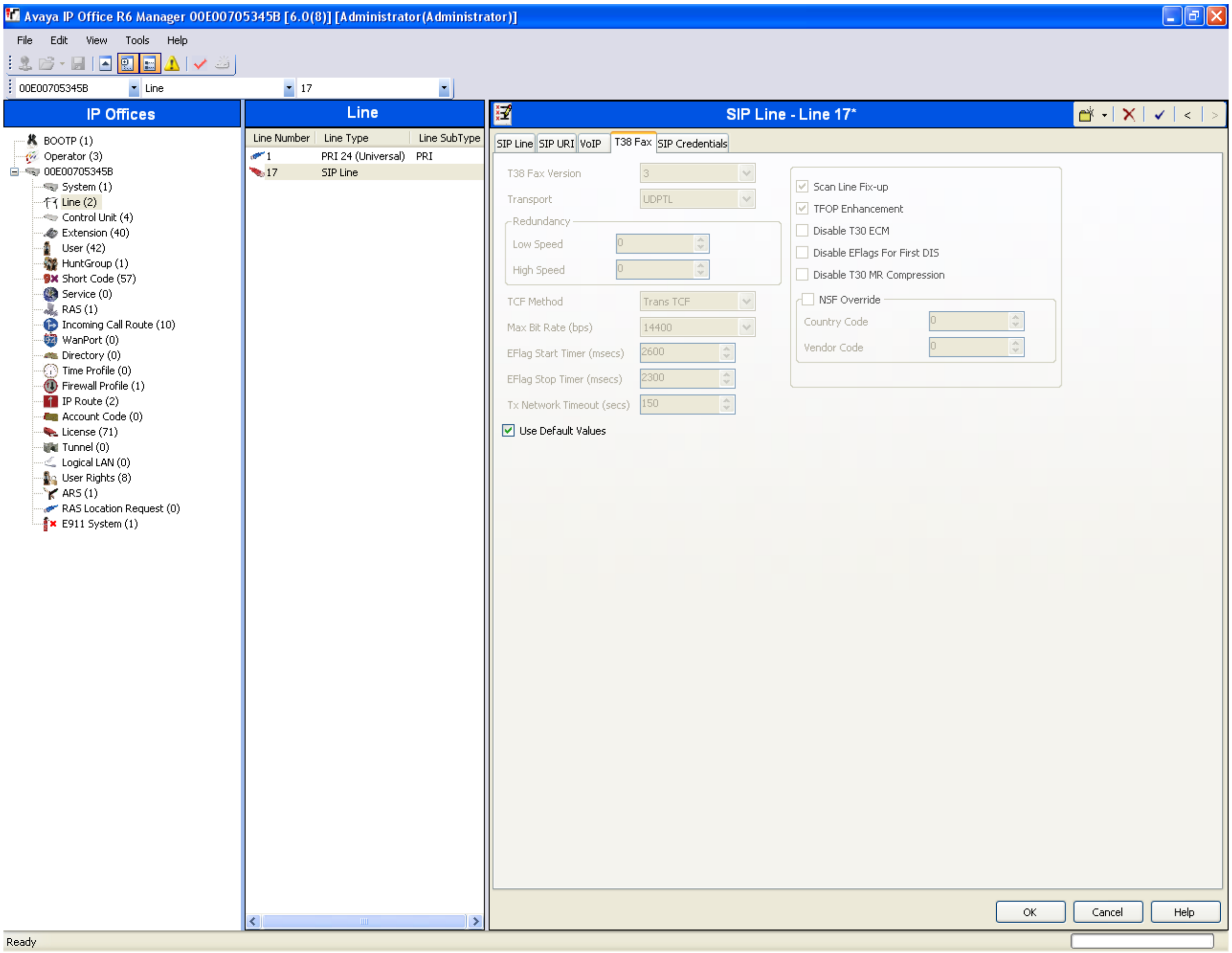
5. *Configure VOIP parameters for the SIP Line.*
   Select the **VOIP URI** tab. Click the **Add** button.

   For **Compression Mode**, select **Automatic Select** or the desired codec from the drop-down list. Check the **Fax Transport Support** and **Re-invite Supported** check boxes. Click the **OK** button.



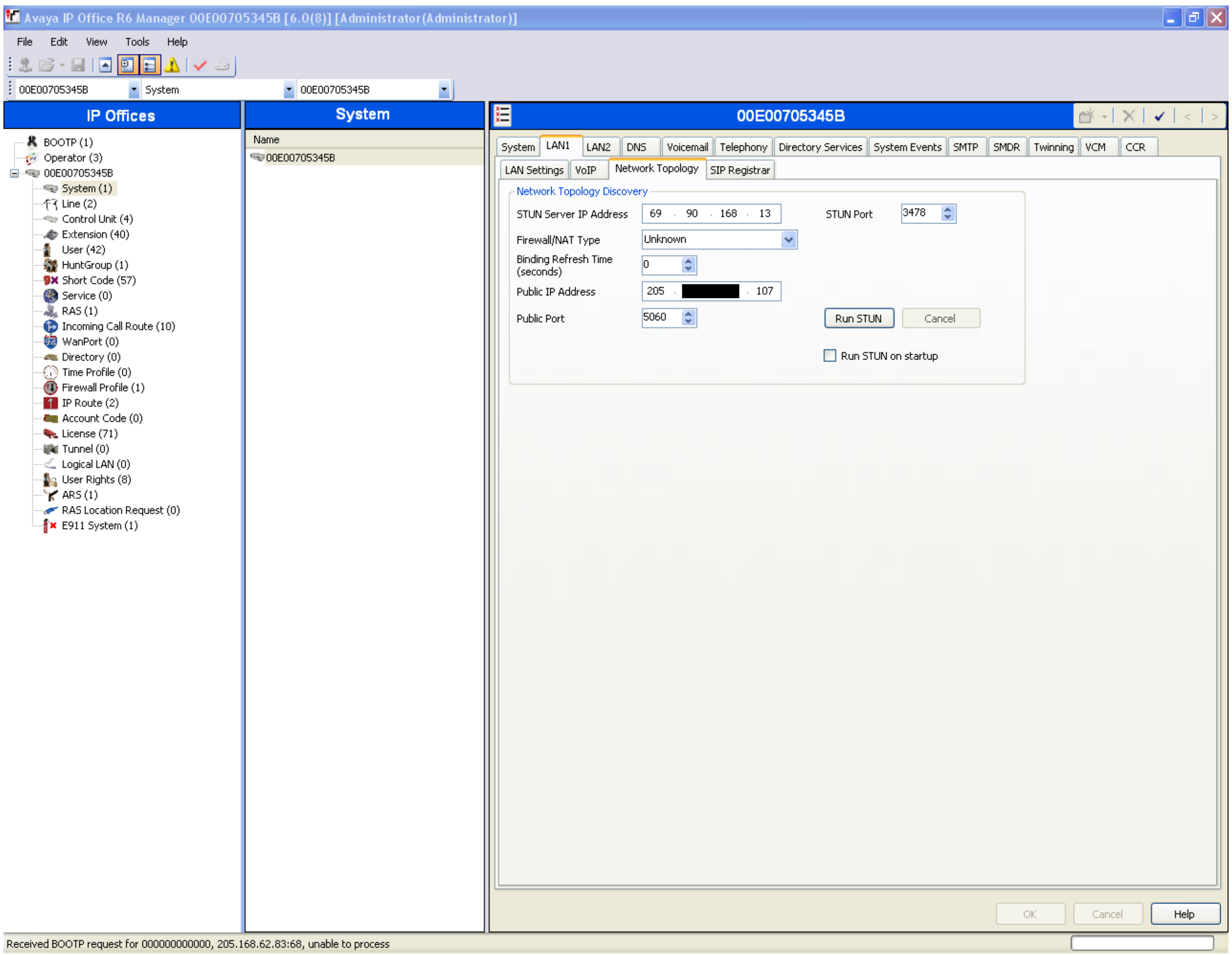
6. *Configure T38 Fax parameters for the SIP Line.*
Select the **T38 Fax** tab.

Check the **Use Default Values** check box. Click the **OK** button.



7. *Configure SIP OPTIONS timer for "keep alive" function with Metaswitch.*
   Select **System** in the left panel. Under the **LAN1** tab, select the **Network Topology** tab.

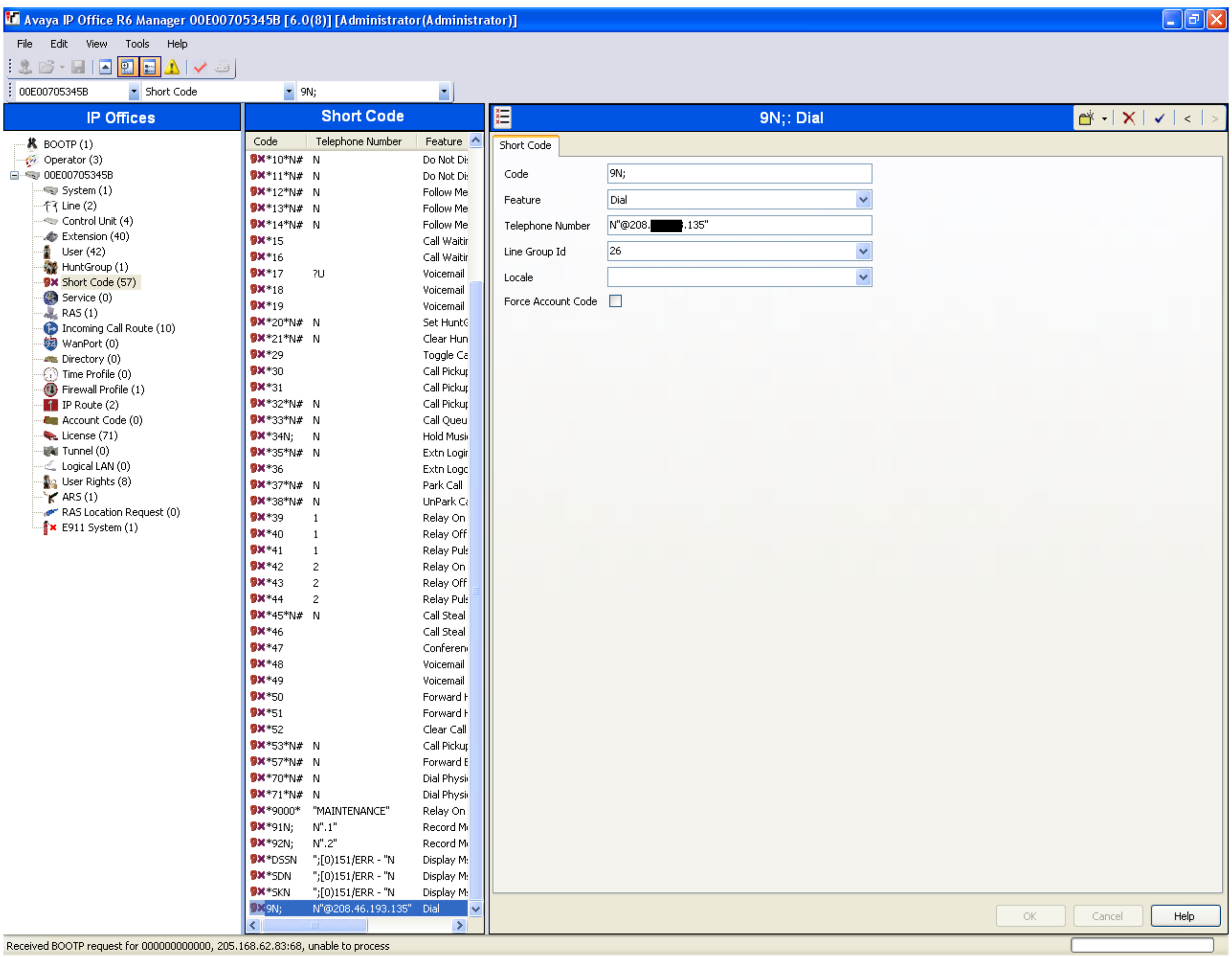
   Set the **Binding Refresh Time** to the desired interval which determines the frequency with which OPTIONS messages will be sent to Metaswitch. For this compliance testing, it is set to the default value of 0 which disables OPTIONS messages. For **Public IP Address**, enter the Avaya IP Office system IP address. Confirm that **Public Port** is set to 5060 and accept the default values for all other fields. Click the **OK** button.



8. *Configure a short code to route calls to Metaswitch*.
Select **Short Code** in the left panel. Right click and select **Add**. Enter **[x]N;**, where **[x]** is a valid number, in the **Code** text box. The number **9** is used for **[x]** in the example below. This code requires the user to dial the digit **9** followed by the destination's telephone number symbolized by **N** in order to route the call out the SIP Trunk.

Note: N can be any number other than a local IP Office extension. For example, a 10-digit number, a toll-free number, directory assistance, information service, etc.

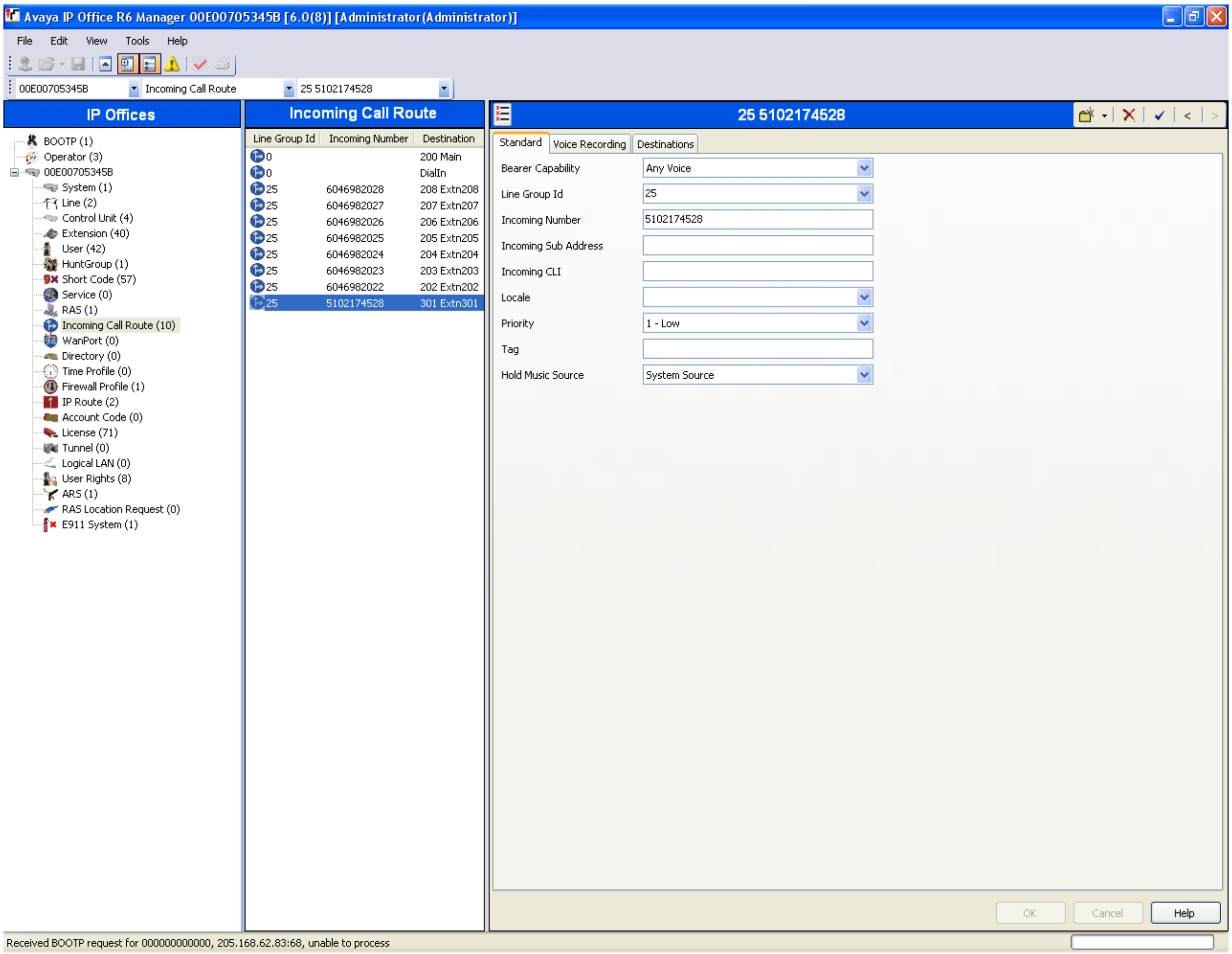
Select **Dial** for the **Feature**. Enter the **Outgoing Group Id** created in Step 4 for the **Line Group Id** field. Enter the dialed number **N** followed by **"@<IP address of Metaswitch >"** for the **Telephone Number** field. The **Telephone Number** field is used to construct the SIP URI in the **To** field of the outgoing SIP INVITE message. Use default values for all other fields. Click the **OK** button.



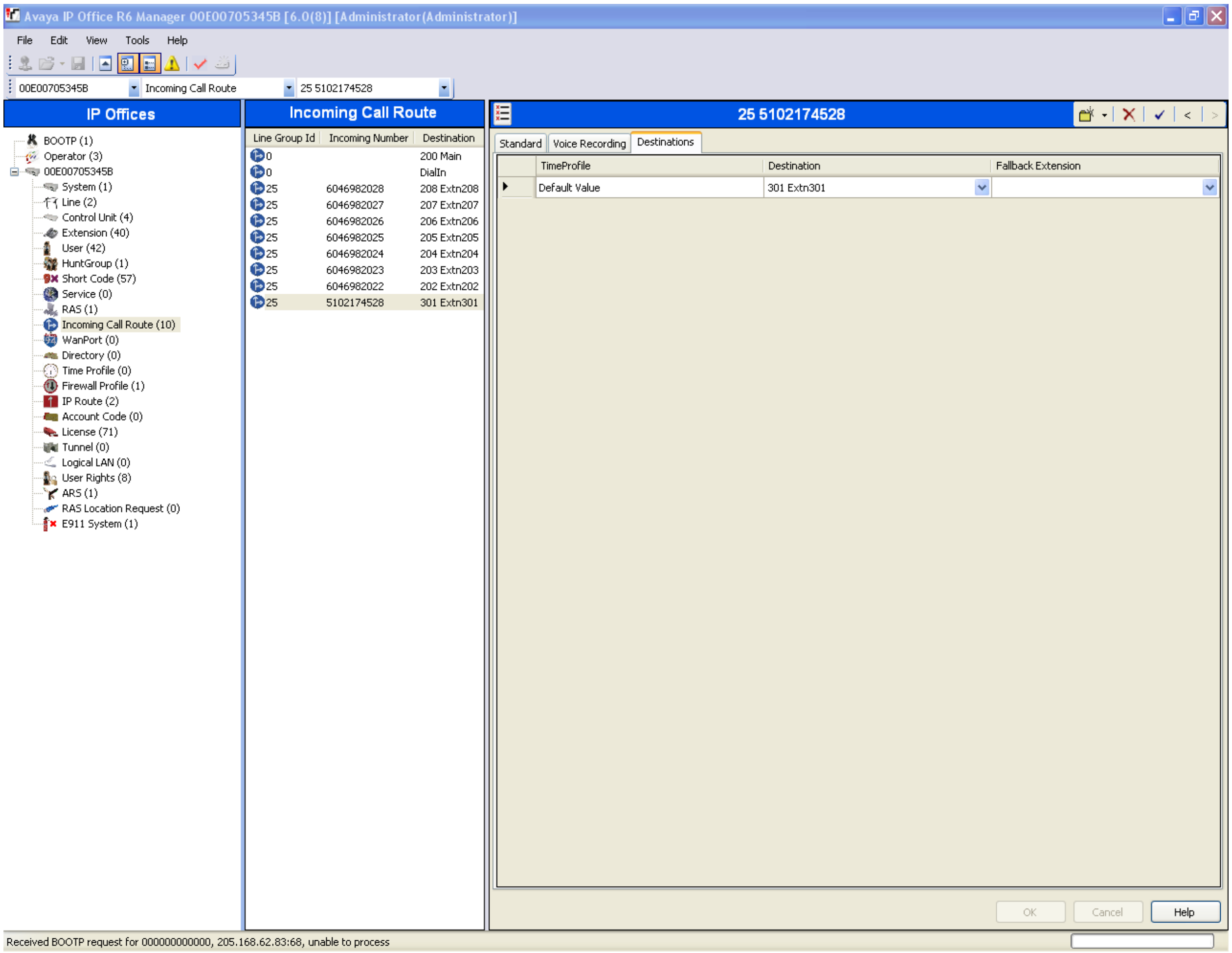
9. *Create an Incoming Call Route for the Inbound SIP calls*.
Select **Incoming Call Route** in the left panel. Right-click and select **New**.

Enter the following:

- **Any Voice** for the **Bearer Capability** field.
- The Incoming Group created for the URI in Step 4 in the **Line Group Id** field.
- The 10-digit DID number provided by Metaswitch that is mapped back to a local IP Office extension in the **Incoming Number** field.
- Use default values for all other fields.



- Next, navigate to the Destinations tab and select the desired local extension number from the drop down menu.
- Click the **OK** button.



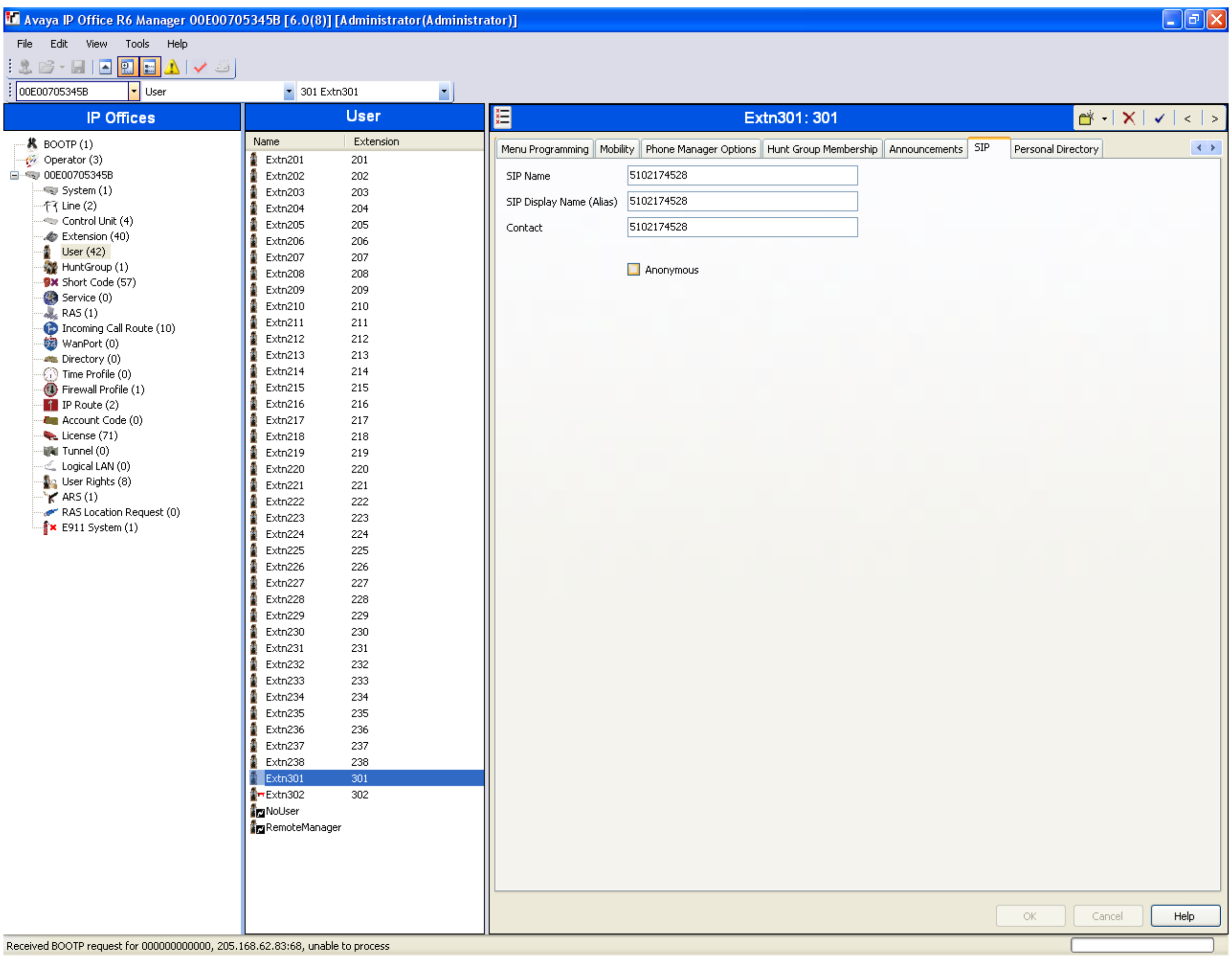
10. *Configure Users' SIP names*.
Select **User** in the left panel. Select the desired user by double-clicking on an entry in the right panel. Select the **SIP** tab.

Modify the **SIP Name** and **Contact** fields to the DID number provided by Metaswitch that is used for this particular extension. These settings instruct the system to use this DID number to construct the:

- user part of the SIP URI in the From header of an outgoing SIP INVITE message
- user part of the SIP URI in the Contact header of an outgoing SIP INVITE message

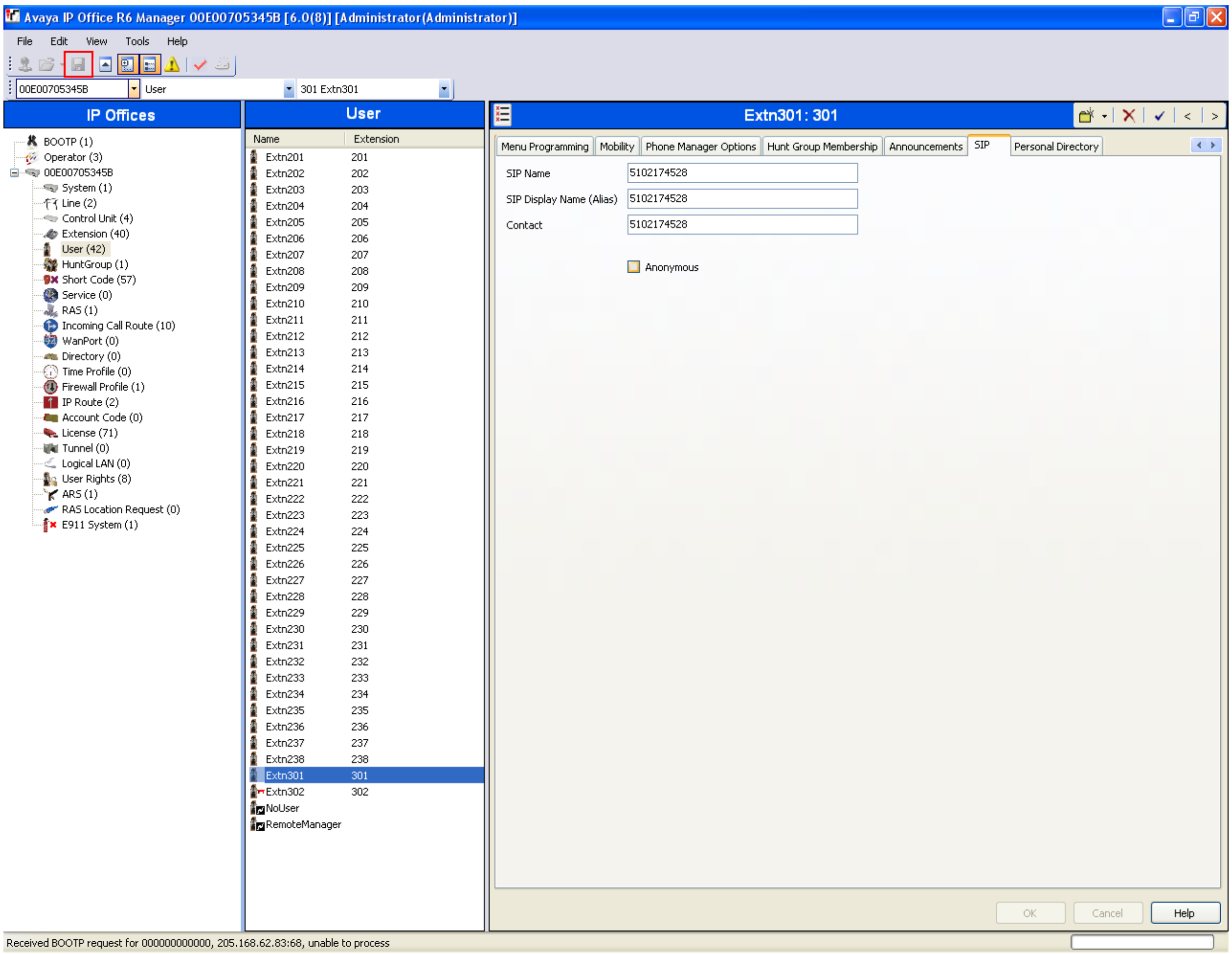
Modify the **SIP Display Name (Alias)** that will be used for the SIP Display info.

The other fields can be left as defaults. Click the **OK** button.



11. Repeat Steps 9 and 10 for all users that will be sending/receiving SIP calls on the system.

12. After making the changes, click on the floppy disk icon (3rd from left) to push the changes to the IP Office system and have them take effect. **Changes will not take effect until this step is completed. ** NOTE ** This may cause a reboot of Avaya IP Office causing service disruption.**



5. Configure Metaswitch

During the test effort, the Metaswitch network was protected by a pair of Acme Packet Net-Net SD 3820 session border controllers. The session border controllers are not required as part of the solution. For brevity, only the configuration of the MetaSphere CFS is discussed below. If a session border controller is used between the MetaSphere CFS solution and the Avaya IP Office solution, contact a Metaswitch Networks support representative for additional configuration details.

5.1. Media Gateway Model

A truncated text dump of the Remote Media Gateway Model used for the Avaya IP Office testing is shown below. For an importable version, contact a Metaswitch customer service representative.

```
begin MediaGatewayModel MediaGatewayModel.177 // Remote Media Gateway Model "avaya ip
office"
  Category                                        SIP
  ModelName                                       avaya ip office
  ControlProtocol                                 SIP
  DefaultModel                                    False
  SupportedHighBandwidthMediaFormats              {G.711 u-law,G.711 A-law}
  SupportedLowBandwidthMediaFormats               {G.726 32kbps,G.729 AB}
  PreferredLowBandwidthMediaFormats               {G.726 32kbps,G.729 AB}
  AdvancedVoiceCodecsPermitted                    Any codecs
  VideoCodecsPermitted                            Any codecs
  PacketizationInterval                           0
  SilenceSuppressionAllowed                       False
  MaximumSimultaneousTransactionsOutstanding      100
  DigitOverhangTime                               250
  FixBitsMGCPMeGaCoSIP                            {Cannot be hub,Simple
                                                   contexts,Cannot play
                                                   ringback,Cannot control endpoint
                                                   connectivity,Cannot move
                                                   contexts,Connections always
                                                   receive,Cannot report detection of
                                                   call-type discrimination tones,T.38
                                                   supported}
  DynamicFixBitsMGCPMeGaCoSIP                     {}
  FixBitsSIP                                      {Supports SDP connectivity
                                                   requests,Supports receiving INVITEs
                                                   with no SDP,Supports receiving SIP
                                                   Reason header over tandem trunk
                                                   calls}
  FixBitsSIP2                                     {}
end //MediaGatewayModel
```

5.2. Configured SIP Binding

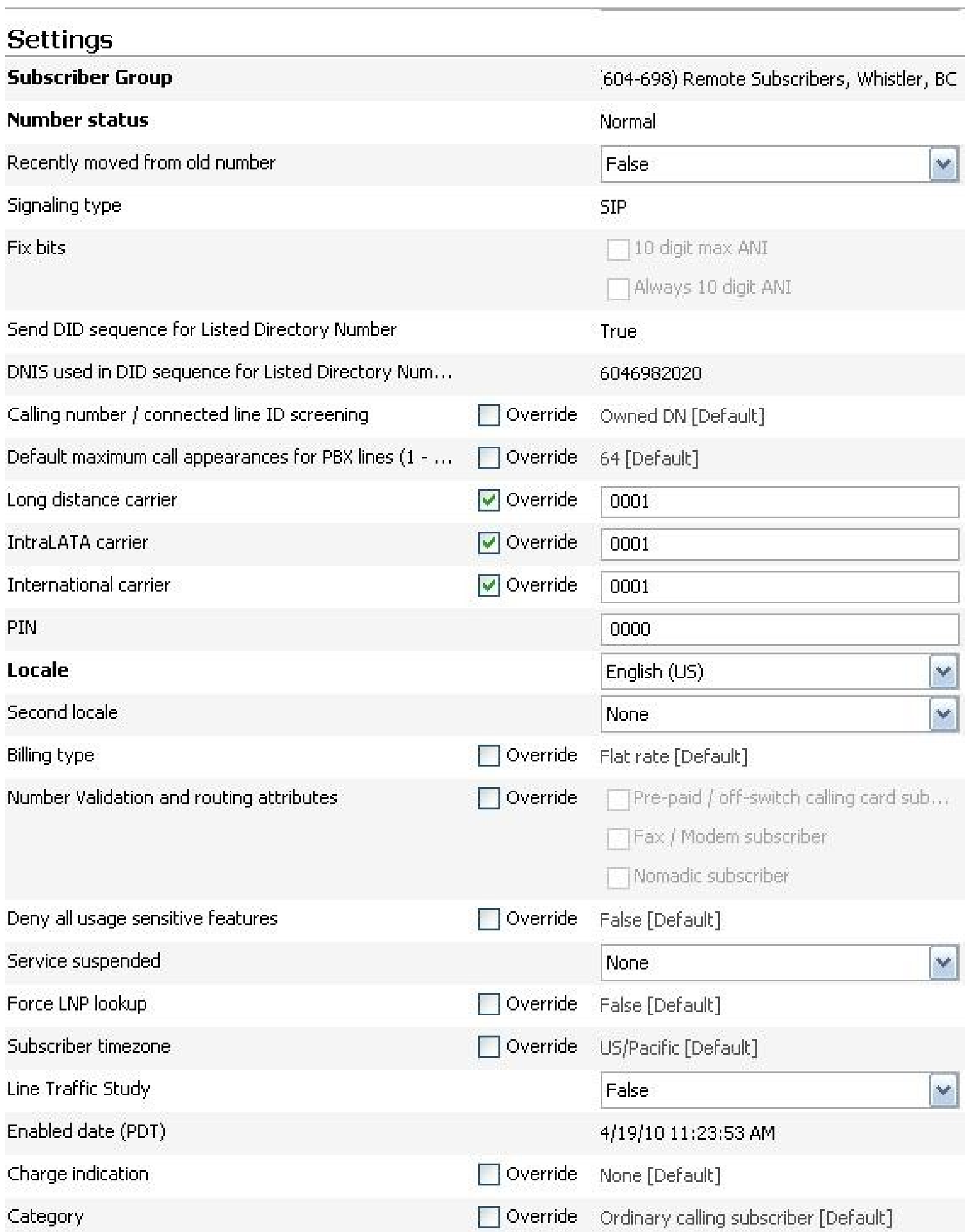
The connection to Avaya IP Office is modeled as a configured SIP binding. During compliance testing, the configured SIP binding was configured as shown below.

Name	Value
Name	Avaya IP Office
Customer information	
Customer information 2	
Customer information 3	
Customer information 4	
Customer information 5	
Customer information 6	
Usage	Subscriber
Use DN for identification	True
SIP authentication required	False
SIP domain name	208.[redacted].135
IP address match required	False
Contact IP address (Format: w.x.y.z)	205.[redacted].107
Contact IP port (0 - 65535)	5060
Supported incoming trunk group parameter type	None
Trunk group parameter type on outgoing messages	None
Proxy IP address (Format: w.x.y.z)	10.220.20.25
Proxy IP port (0 - 65535)	5060
Transport protocol	UDP
Media Gateway model	ledia Gateway Model "avaya ip office"
Network Node ☐ Override	None [Default]
Preferred location of Trunk Gateway	None
ESA Protection Domain	None
Trusted	True
Use caller name provided by SIP device	False
Play announcements when error conditions occur	True
Use static NAT mapping	False
Maximum call appearances (1 - 2147483647)	1024
Maximum concurrent high bandwidth call appearanc...	0
Poll peer device	True
Polling interval (1 - 3600 seconds)	30
Current number of call appearances in use	0

5.3. PBX object configuration

The Avaya IP Office is modeled in MetaView as a PBX. The settings used during compliance testing are shown in the sections below.

5.3.1. PBX Object



Settings

Subscriber Group		(604-698) Remote Subscribers, Whistler, BC
Number status		Normal
Recently moved from old number		False
Signaling type		SIP
Fix bits		☐ 10 digit max ANI ☐ Always 10 digit ANI
Send DID sequence for Listed Directory Number		True
DNIS used in DID sequence for Listed Directory Num...		6046982020
Calling number / connected line ID screening	☐ Override	Owned DN [Default]
Default maximum call appearances for PBX lines (1 - ...	☐ Override	64 [Default]
Long distance carrier	☑ Override	0001
IntraLATA carrier	☑ Override	0001
International carrier	☑ Override	0001
PIN		0000
Locale		English (US)
Second locale		None
Billing type	☐ Override	Flat rate [Default]
Number Validation and routing attributes	☐ Override	☐ Pre-paid / off-switch calling card sub... ☐ Fax / Modem subscriber ☐ Nomadic subscriber
Deny all usage sensitive features	☐ Override	False [Default]
Service suspended		None
Force LNP lookup	☐ Override	False [Default]
Subscriber timezone	☐ Override	US/Pacific [Default]
Line Traffic Study		False
Enabled date (PDT)		4/19/10 11:23:53 AM
Charge indication	☐ Override	None [Default]
Category	☐ Override	Ordinary calling subscriber [Default]

5.3.2. PBX Line Object

Settings		
Configured SIP Binding		Avaya IP Office
Maximum call appearances (1 - 2147483647)	☑ Override	10
Line usage		Voice and fax
PBX plays ringback		False

5.3.3. DID objects

Two DID ranges were configured during compliance testing due to setup in Metaswitch's test lab. Two DID ranges are not required.

Type	DID range
Description	normal
Range size (1 - 1000000000)	9
(First) Directory number	6046982021
Last Directory number	6046982029
First code	6046982021
Last code	6046982029

Type	DID range
Description	PSTN
Range size (1 - 1000000000)	1
(First) Directory number	5102174528
Last Directory number	5102174528
First code	5102174528
Last code	5102174528

6. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Metaswitch MetaSphere CFS solution and the Avaya IP Office solution. This section covers the general test approach and the test results.

Avaya IP Office was connected using SIP trunking to the Metaswitch MetaSphere CFS solution. The general test approach included the following:

- Inbound Calls – Verify that calls placed from a PSTN telephone to a DID or toll free number are properly routed via the SIP trunk to the expected extension on Avaya IP Office. Verify the talk-path exists in both directions, that calls remain stable for several minutes and disconnect properly.

- Outbound Calls – Verify that calls placed to a PSTN telephone are properly routed via the SIP trunk. Verify that the talk-path exists in both directions and that calls remain stable and disconnect properly.

- Inbound and Outbound fax calls – Verify successful transmission of single and multi-page faxes.

- Inbound DTMF Digit Navigation – Verify inbound DID calls can properly navigate the Avaya IP Office voice mail menus.

- Outbound DTMF Digit Navigation – Verify outbound calls can properly navigate a voice mail or interactive response system reached via a PSTN number.

Interoperability testing of the sample configuration was completed with successful results.

The following observations were noted:

1. The Metaswitch test lab did not support x11 calls (e.g 411, 911, international, etc.).

2. Due to limitations of the Metaswitch test lab configuration, the caller-id did not always display the proper calling party number for calls to and from the PSTN (rather, an administered 10 digit number was display).

7. Verification Steps

This section provides verification steps that may be performed to verify that the H.323, digital and analog endpoints can place outbound calls and receive inbound calls through Metaswitch MetaSphere CFS via a SIP trunking solution.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.

2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.

3. Verify that the user on the PSTN can terminate an active call by hanging up.

4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

8. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony solution to a Metaswitch MetaSphere CFS solution. Metaswitch offers a flexible VoIP solution for customers with a SIP based telephone network. SIP trunks may be used to connect company networks to the public telephone network via converged IP access, providing an alternative to traditional hardwired telephony trunk lines.

9. Additional References

This section references the documentation relevant to these Application Notes.

The following Avaya product documentation is available at http://support.avaya.com.

[1] *IP Office 6.0 Installation, Issue 21j,* April 2010
Document Number 15-601042

[2] *IP Office Release 6 Manager 8.0, Issue 24k,* April 2010
Document Number 15-601011

[3] Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

[4] RFC 3261 *SIP: Session Initiation Protocol* http://www.ietf.org/

[5] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
http://www.ietf.org/

Metaswitch product documentation is available at http://www.metaswitch.com/support/.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.