



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Cablevision Optimum Voice SIP Trunking with Avaya IP Office - Issue 1.1

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Cablevision Optimum Voice and Avaya IP Office.

Optimum Voice SIP Trunking provides PSTN access via a SIP trunk between the enterprise and the Cablevision cable network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Cablevision is a member of the Avaya DevConnect Service Provider program. 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 procedures for configuring Session Initiation Protocol (SIP) Trunking between Cablevision Optimum Voice and Avaya IP Office.

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1.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to Optimum Voice SIP Trunking. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Response to SIP OPTIONS queries
- Incoming PSTN calls to various phone types
Phone types included H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types
Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients
Avaya IP Office supports two soft clients: Avaya IP Office Phone Manager and Avaya IP Office Softphone. Avaya IP Office Phone Manager supports two modes (PC softphone and telecommuter). Both clients in each supported mode were tested.
- Various call types including: local, long distance, international, and outbound toll-free.
- Codec G.711MU
- Caller ID presentation and Caller ID restriction
- DTMF transmission using in-band tones
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning

Items not supported by Optimum Voice SIP Trunking or not tested included the following:

- Inbound toll-free, 411 calls and emergency calls (911) are supported but were not tested as part of the compliance test.
- Operator services are not supported.
- DTMF transmission using RFC 2833 is not supported.
- T.38 Fax is not supported.

Interoperability testing of Optimum Voice SIP Trunking was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Twinning – Second Dial Tone:** On a active call answered at an external twinned phone, the Avaya IP Office user should be able to get a second dial tone from Avaya IP Office by dialing **. This was not possible with this solution.

- **Phone Manager Telecommuter Mode:** During testing with the Avaya IP Office Phone Manager (H.323 soft client) in telecommuter mode, one combination of telecommuter phone and destination number resulted in a call with no audio. Calls to all other destination numbers worked properly. The case of no audio resulted when the telecommuter phone and destination number resided on the same external PBX.

1.2. Support

For technical support on Optimum Voice SIP Trunking, contact Cablevision using the Customer Service links at www.optimumbusiness.com.

2. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to Optimum Voice SIP Trunking.

Located at the edge of the enterprise site is an Edgewater Networks EdgeMarc 4552 router provided by Cablevision to serve as a demarcation point for the SIP trunking service. For testing purposes, the WAN interface of the router was connected to a broadband Internet connection with a public IP address to access the service. In an actual customer configuration, the router would connect to a cable modem to access the Cablevision network. The LAN port 1 of the router is connected to the enterprise LAN.

Also located at the enterprise site is an Avaya IP Office 500 with analog expansion module. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 5600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya IP Office Phone Manager, an Avaya IP Office Softphone, an Avaya 5420 Digital Telephone, and an Avaya 6210 Analog Telephone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for voicemail and running Avaya IP Office Manager to configure the Avaya IP Office.

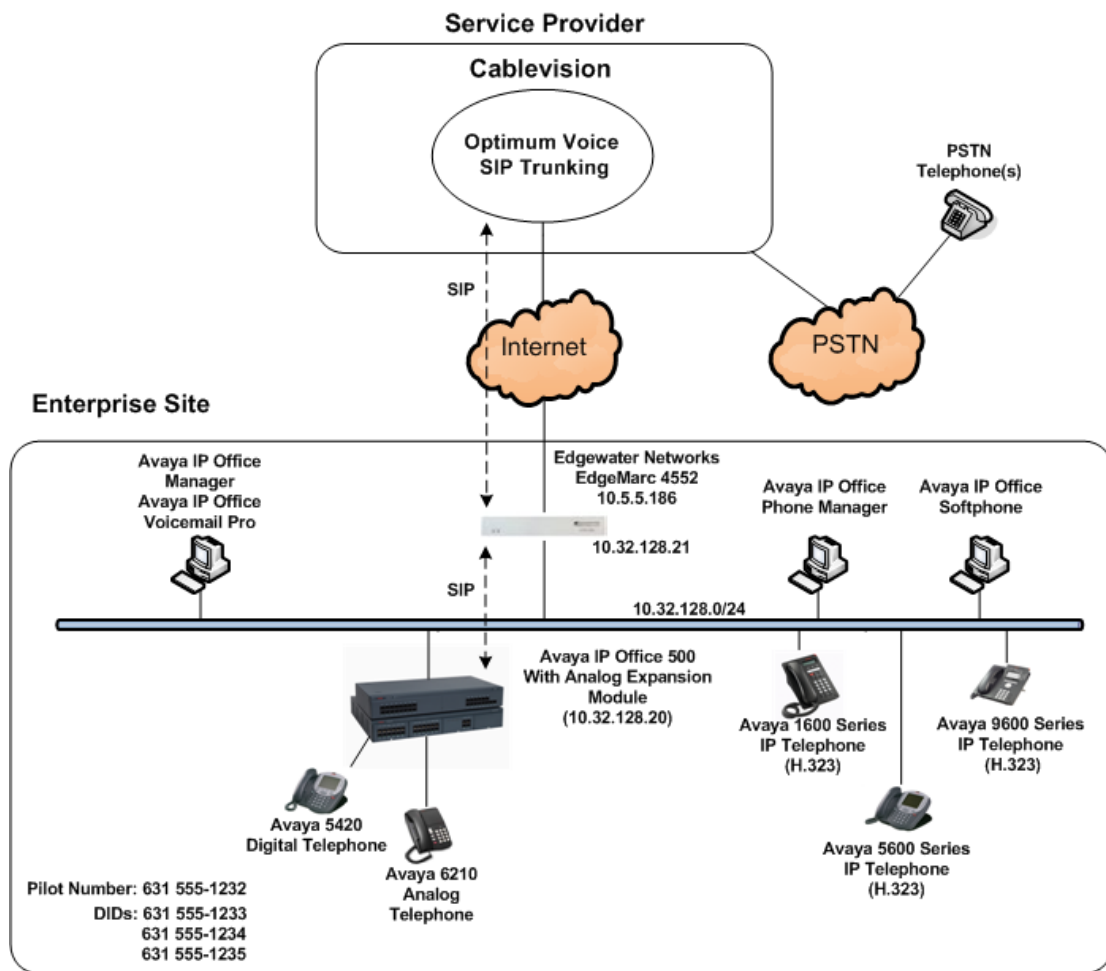


Figure 1: Test Configuration

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that can not be routed by the PSTN.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Cablevision. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Cablevision. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office sent 11 digits in the Request URI and the To field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, Optimum Voice SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

Cablevision uses the phone number in the From header of a SIP INVITE message to authenticate the calling party. Thus, a call will be rejected by the network unless the From header contains a number known to Cablevision. This is especially important for calls inbound from the PSTN which are redirected back to the PSTN by call forwarding or twinning. For call forwarding, Avaya IP Office always sends the number of the forwarding phone in the From header. This is a number known to

Cablevision. As a result, the call display on the destination phone shows the forwarding party not the original caller. For twinning, this behavior can be slightly altered through configuration. See **Section 4.3** and **4.4** for details.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the EdgeMarc router such as a session border controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and EdgeMarc router must be allowed to pass through these devices.

3. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment	Release
Avaya IP Office 500 with Analog Expansion Module	6.0 (8)
Avaya IP Office Analog Expansion Module	8.0 (8)
Avaya Voicemail Pro	6.0.22
Avaya IP Office Manager	8.0 (8)
Avaya 1608SW IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya 5620 IP Telephone (H.323)	
Avaya 9640SW IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.1
Avaya IP Office Phone Manager	4.2.25
Avaya IP Office Softphone	3.0
Avaya 5420 Digital Telephone	N/A
Avaya 6210 Analog Telephone	N/A

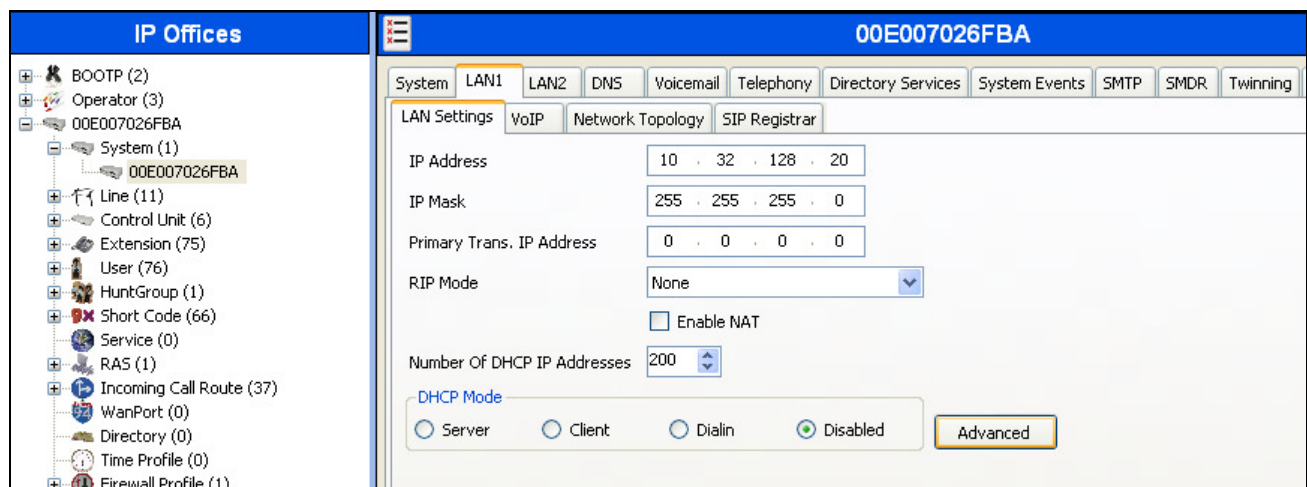
Cablevision Components	
Equipment	Release
Edgewater Networks EdgeMarc 4552	10.2.0.2

4. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to Optimum Voice SIP Trunking. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Softphone support) is assumed to already be in place.

4.1. LAN1 Settings

In the sample configuration, the MAC address **00E007026FBA** was used as the system name and the LAN1 port was used to connect the Avaya IP Office to the enterprise network. To access the LAN1 settings, first navigate to **System → 00E007026FBA** in the Navigation Pane and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are set from values shown in **Figure 1**. All other parameters should be set according to customer requirements.



On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN1. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.

00E007026FBA

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP

LAN Settings VoIP Network Topology SIP Registrar

☒ H323 Gatekeeper Enable

☒ SIP Trunks Enable

☒ SIP Registrar Enable

☐ H323 Auto-create Extn

☐ H323 Auto-create User

☒ Enable RTCP Monitoring On Port 5005

RTP Port Number Range

Port Range (Minimum) 49152

Port Range (Maximum) 53246

DiffServ Settings

B8 DSCP(Hex) FC DSCP Mask (Hex) B8 SIG DSCP (Hex)

46 DSCP 63 DSCP Mask 46 SIG DSCP

DHCP Settings

Primary Site Specific Option Number (SSON) 176

Secondary Site Specific Option Number (SSON) 242

VLAN Not Present

On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**.
- Set **Binding Refresh Time (seconds)** to **300**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. See **Section 4.9** for complete details.
- All other parameters should be set according to customer requirements.

The screenshot displays the Avaya IP Office configuration window with the title bar '00E007026FBA'. The 'System' tab is selected, and within it, the 'Network Topology' sub-tab is active. The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server IP Address:** A text box containing '0 . 0 . 0 . 0'.
- STUN Port:** A spinner box set to '3478'.
- Firewall/NAT Type:** A pull-down menu currently showing 'Open Internet'.
- Binding Refresh Time (seconds):** A spinner box set to '300'.
- Public IP Address:** A text box containing '0 . 0 . 0 . 0'.
- Public Port:** A spinner box set to '0'.
- Run STUN:** A button.
- Cancel:** A button.
- Run STUN on startup:** An unchecked checkbox.

4.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab on the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. Choose the **Companding Law** typical for the enterprise location. For North America, **ULAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk.

The screenshot shows the 'Telephony' configuration page for a system identified as '00E007026FBA'. The interface includes a top navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony (selected), Directory Services, System Events, SMTP, SMDR, and Twinning. Below this, there are sub-tabs for Telephony, Tones & Music, and Call Log. The main configuration area is divided into two columns. The left column, titled 'Analogue Extensions', contains settings for call sequences (Default Outside Call Sequence: Normal, Default Inside Call Sequence: Ring Type 1, Default Ring Back Sequence: Ring Type 2), timing parameters (Dial Delay Time: 4s, Dial Delay Count: 0, Default No Answer Time: 15s, Hold Timeout: 0s, Park Timeout: 300s, Ring Delay: 5s, Call Priority Promotion Time: Disabled), and other options (Default Currency: USD, Automatic Codec Preference: G.711 ULAW 64K). The right column, titled 'Companding Law', has two sections: 'Switch' and 'Line'. Both sections have radio buttons for ULAW (selected) and ALAW. Below these are several checkboxes: DSS Status, Auto Hold, Dial By Name (checked), Show Account Code (checked), Inhibit Off-Switch Forward/Transfer (unchecked), Restrict Network Interconnect, Drop External Only Impromptu Conference, and Visually Differentiate External Call.

Category	Setting	Value
Analogue Extensions	Default Outside Call Sequence	Normal
	Default Inside Call Sequence	Ring Type 1
	Default Ring Back Sequence	Ring Type 2
	Dial Delay Time (secs)	4
	Dial Delay Count	0
	Default No Answer Time (secs)	15
	Hold Timeout (secs)	0
	Park Timeout (secs)	300
	Ring Delay (secs)	5
	Call Priority Promotion Time (secs)	Disabled
Default Currency	USD	
Automatic Codec Preference	G.711 ULAW 64K	
Companding Law - Switch	ULAW	Selected
	ALAW	Unselected
Companding Law - Line	ULAW Line	Selected
	ALAW Line	Unselected
Other Settings	DSS Status	Unchecked
	Auto Hold	Unchecked
	Dial By Name	Checked
	Show Account Code	Checked
	Inhibit Off-Switch Forward/Transfer	Unchecked
	Restrict Network Interconnect	Unchecked
	Drop External Only Impromptu Conference	Unchecked
Visually Differentiate External Call	Unchecked	

4.3. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affects twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 4.4**).

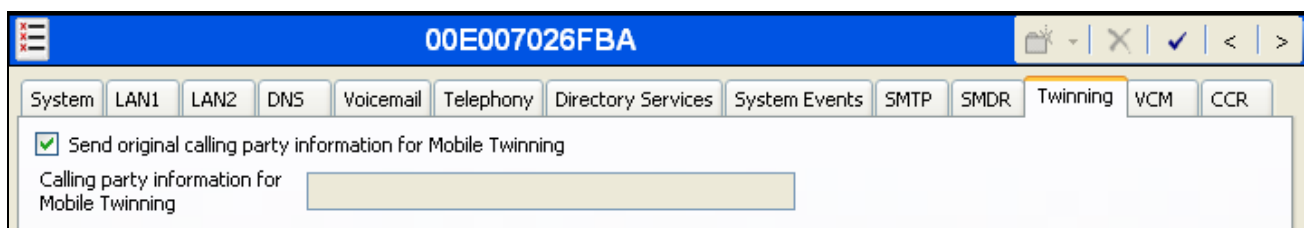
If this box (representing the first parameter) is checked, the setting of the second parameter is ignored and Avaya IP Office will send the following in the SIP From Header. The value in the From header determines what is displayed as the calling party number:

- On calls from an internal extension to an enterprise phone with twinning enabled, Avaya IP Office will send the calling party number of the originating extension to the twinned destination.
- On calls from the PSTN to an enterprise phone with twinning enabled, Avaya IP Office will send the calling party number of the enterprise host phone (instead of the number of the originating caller) to the twinned destination.

If this box is unchecked, the value sent in the SIP From header is determined by the setting of the second parameter (the **Send Caller ID** value in **Section 4.4**). Cablevision supports the SIP Diversion header for re-directed calls which is one of the choices available for setting the **Send Caller ID** parameter in **Section 4.4**. If this option is chosen, the calling party number is changed in the following way:

- On all calls to an enterprise phone with twinning enabled (internal or PSTN), the calling party number appears as the pilot number on the twinned destination. The EdgeMarc router overwrites the SIP From header from Avaya IP Office with the pilot number. The pilot number is described in **Section 4.4** as part of the Contact header.

For the compliance test, the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab was checked which overrides any setting of the **Send Caller ID** parameter on the **SIP Line** form.



The screenshot shows the Avaya IP Office configuration window for extension 00E007026FBA. The 'Twinning' tab is selected. The checkbox 'Send original calling party information for Mobile Twinning' is checked. Below this checkbox is a text input field labeled 'Calling party information for Mobile Twinning'.

4.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Optimum Voice SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- Set the **ITSP Domain Name** to the IP address of the LAN port 1 interface of the EdgeMarc router.
- Set **ITSP IP Address** to the IP address of the LAN port 1 interface of the EdgeMarc router.
- Set **Send Caller ID** to **None**. For the compliance test, this parameter was ignored since the **Send original calling party information for Mobile Twinning** box was checked in **Section 4.3**.
- Check the **In Service** box.
- Uncheck the **Check OOS** box. Optimum Voice SIP Trunking does not respond to the SIP OPTIONS messages sent by Avaya IP Office. Thus, this setting will prevent Avaya IP Office from taking the SIP trunk out of service.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to the network port configured in **Section 4.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with 'Line (11)' selected. The main area is titled 'SIP Line - Line 24' and contains several tabs: 'SIP Line', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is active, showing the following configuration fields:

Field	Value	Field	Value
Line Number	24	Registration Required	<input type="checkbox"/>
ITSP Domain Name	10.32.128.21	In Service	<input checked="" type="checkbox"/>
ITSP IP Address	10 . 32 . 128 . 21	Use Tel URI	<input type="checkbox"/>
Prefix		Check OOS	<input type="checkbox"/>
National Prefix	0	Call Routing Method	Request URI
Country Code			
International Prefix	00		
Send Caller ID	None		

Below these fields is a 'Network Configuration' section with the following settings:

Field	Value	Field	Value
Layer 4 Protocol	UDP	Send Port	5060
Use Network Topology Info	LAN 1	Listen Port	5060

Each SIP URI that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. For the compliance test, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

- Set **Local URI**, and **Display Name** to *Internal Data*. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 4.6**.
- Set **Contact** to the pilot number provided by Cablevision. This number is used to authenticate all calls from the enterprise. This number must not be assigned to any internal extension.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **24** was defined that only contains this line (line 24).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

The screenshot displays the 'SIP Line - Line 24' configuration window. The 'SIP URI' tab is active, showing a table with columns: Channel, Groups, Via, Local URI, and Contact. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is the 'New Channel' section, which contains the following fields and values:

Via	10.32.128.20
Local URI	Use Internal Data
Contact	6315551232
Display Name	Use Internal Data
Registration	0: <None>
Incoming Group	24
Outgoing Group	24
Max Calls per Channel	10

At the bottom right of the window are 'OK' and 'Cancel' buttons.

Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below.

- Set the **Compression Mode** to *Automatic Select*.
- Set the **DTMF Support** field to *Inband*. This directs Avaya IP Office to send DTMF tones as inband tones in the audio stream instead of RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** box.
- Uncheck the **Fax Transport Support** box.
- Uncheck the **Re-invite Supported** box.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 24' configuration window with the 'VoIP' tab selected. The window has a blue title bar and a toolbar with icons for help, cancel, OK, and navigation. Below the title bar are tabs for 'SIP Line', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'VoIP' tab is active, showing the following settings:

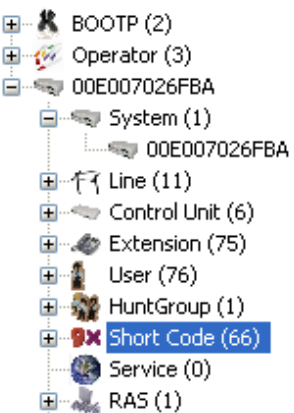
Compression Mode	Advanced	Automatic Select	<input type="checkbox"/> VoIP Silence Suppression
Call Initiation Timeout (s)	4		<input type="checkbox"/> Fax Transport Support
DTMF Support	Inband		<input type="checkbox"/> Re-invite Supported
			<input type="checkbox"/> Use Offerer's Preferred Codec

4.5. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@10.32.128.21"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The IP address 10.32.128.21 is the IP address of the LAN port 1 interface of the EdgeMarc router.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 4.4**. This short code will use this line group when placing the outbound call.

Click the **OK** button (not shown).

IP Offices	9N;; Dial												
	<div>Short Code</div> <table><tr><td>Code</td><td>9N;;</td></tr><tr><td>Feature</td><td>Dial</td></tr><tr><td>Telephone Number</td><td>N"@10.32.128.21"</td></tr><tr><td>Line Group Id</td><td>24</td></tr><tr><td>Locale</td><td>United States (US English)</td></tr><tr><td>Force Account Code</td><td><input type="checkbox"/></td></tr></table>	Code	9N;;	Feature	Dial	Telephone Number	N"@10.32.128.21"	Line Group Id	24	Locale	United States (US English)	Force Account Code	<input type="checkbox"/>
Code	9N;;												
Feature	Dial												
Telephone Number	N"@10.32.128.21"												
Line Group Id	24												
Locale	United States (US English)												
Force Account Code	<input type="checkbox"/>												

4.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 4.4**. To configure these settings, first navigate to **User→Name** in the Navigation Pane where *Name* is the name of the user to be modified. In the example below, the name of the user is Extn370. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 4.4**). The example below shows the settings for user Extn370. The **SIP Name** is set to one of the DID numbers assigned to the enterprise from Cablevision. The **Contact** is set to the pilot number. The **Contact** will be set to the pilot number for all users. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click the **OK** button (not shown).

The screenshot shows the Avaya SIP configuration interface. On the left is a navigation pane titled 'IP Offices' with a tree structure containing the following items: BOOTP (2), Operator (3), 00E007026FBA, System (1), 00E007026FBA, Line (11), Control Unit (6), Extension (75), User (76), HuntGroup (1), and Short Code (66). The 'User (76)' item is selected. The main pane is titled 'Extn370: 370' and contains a series of tabs: User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button, Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, SIP, and Personal D. The 'SIP' tab is active. It contains three text input fields: 'SIP Name' with the value '6315551235', 'SIP Display Name (Alias)' with the value 'Extn370', and 'Contact' with the value '6315551232'. Below these fields is an unchecked checkbox labeled 'Anonymous'.

4.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider except the DID number used for trunk registration. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capacity** to *Any Voice*.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 4.4**.
- Set the **Incoming Number** to the incoming number on which this route should match. Matching is right to left.
- Default values can be used for all other fields.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Standard' tab selected. The title bar indicates the line group '24' and the incoming number '6315551235'. The left navigation pane shows a tree structure with 'Incoming Call Route (38)' selected. The main area contains the following fields:

Field	Value
Bearer Capability	Any Voice
Line Group Id	24
Incoming Number	6315551235
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 6315551235 on line 24 are routed to extension 370.

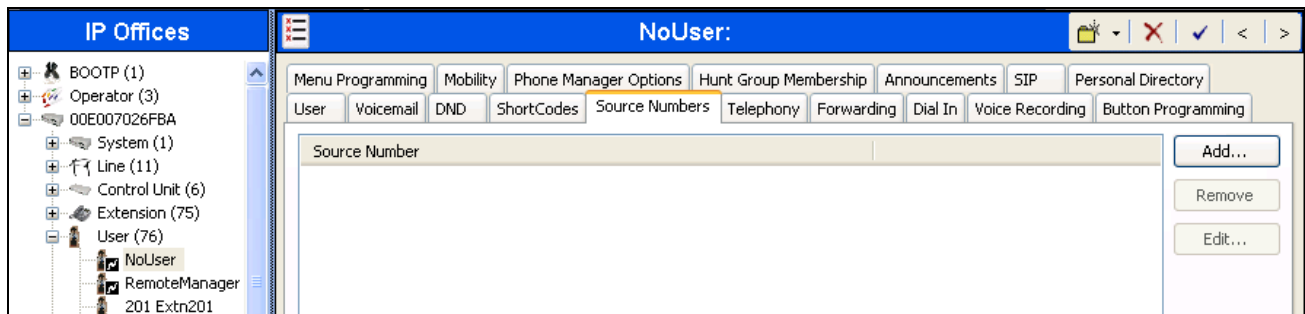
The screenshot shows the 'Incoming Call Route' configuration window with the 'Destinations' tab selected. The title bar is the same as the previous screenshot. The main area contains a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	370 Extn370	

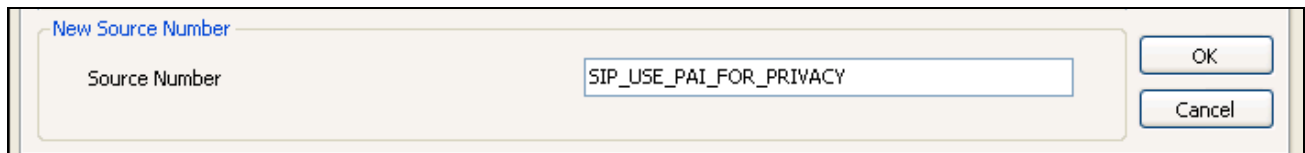
4.8. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “restricted” and “anonymous” respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. By default, Avaya IP Office will use PPI for privacy. Optimum Voice SIP Trunking supports PAI for the purposes of privacy.

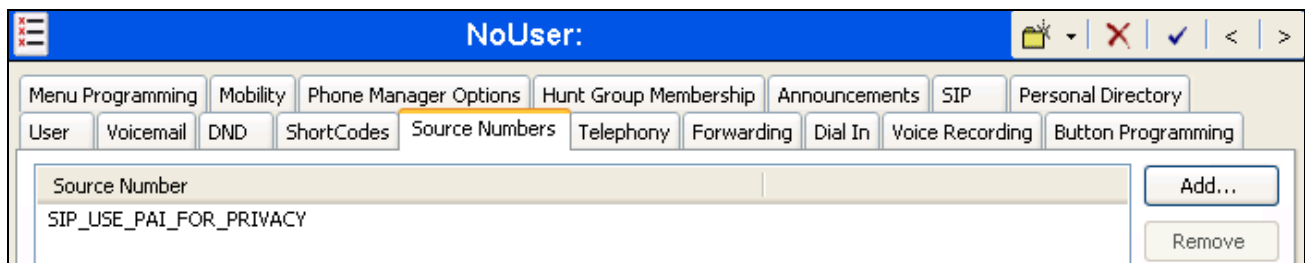
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_USE_PA1_FOR_PRIVACY**. Click **OK**.



The **SIP_USE_PA1_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.

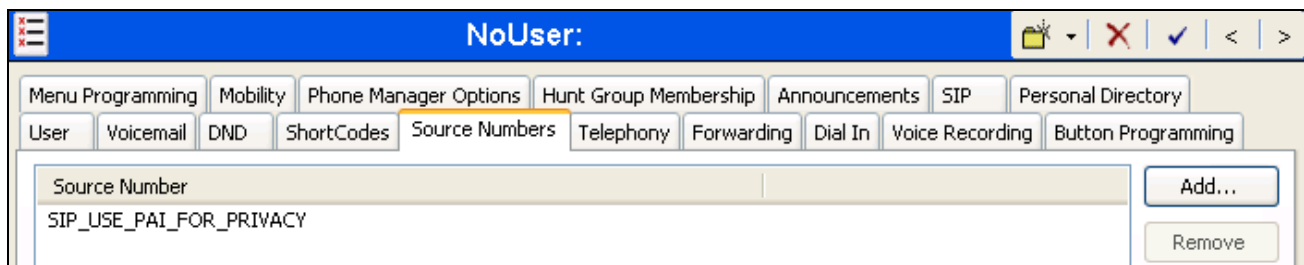


4.9. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 4.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



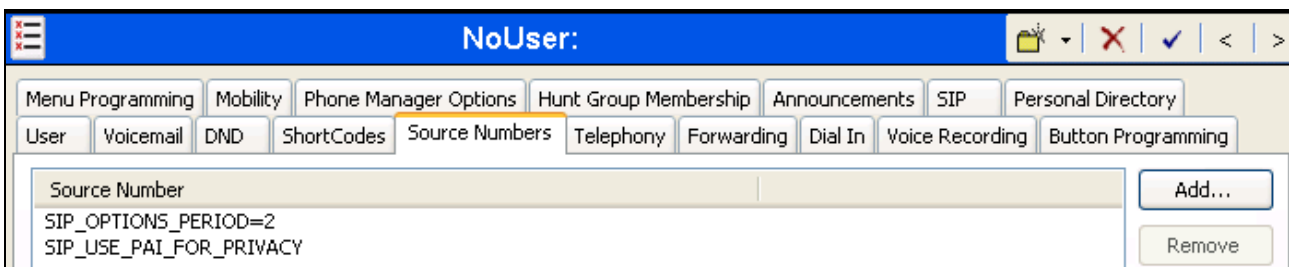
The screenshot shows the configuration window for the 'noUser' user. The 'Source Numbers' tab is selected in the Details Pane. The 'Source Number' field contains the text 'SIP_USE_PAI_FOR_PRIVACY'. There are 'Add...' and 'Remove' buttons to the right of the field.

At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_OPTIONS_PERIOD=X**, where **X** is the desired value in minutes. Click **OK**.



The screenshot shows a 'New Source Number' dialog box. The 'Source Number' field contains the text 'SIP_OPTIONS_PERIOD=2'. There are 'OK' and 'Cancel' buttons to the right of the field.

The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 2 minutes was desired. The **Binding Refresh Time** was set to **300** seconds (5 minutes) in **Section 4.1**. The **SIP_OPTIONS_PERIOD** was set to **2** minutes. Avaya IP Office chose the OPTIONS period as the smaller of these two values (2 minutes). Click the **OK** button (not shown).



4.10. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

5. Cablevision Optimum Voice SIP Trunking Configuration

Cablevision is responsible for the configuration of Optimum Voice SIP Trunking including the WAN interface of the EdgeMarc router. Cablevision will configure the EdgeMarc router before shipping the device to the customer. Once installed on the customer site, Cablevision will be able to access the EdgeMarc router remotely via this interface. In addition, Cablevision will provide the customer the necessary information to configure the Avaya IP Office SIP connection including:

- Supported codecs
- Pilot number
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

The customer will be responsible for configuring the voice network interface of the EdgeMarc router, along with basic SIP trunk parameters. LAN port 1 is used for the voice network and will communicate with the Avaya IP Office. LAN port 4 of the EdgeMarc router is used for management and is pre-configured with IP address 10.10.200.1. To perform the customer configuration of the EdgeMarc router, connect a PC to LAN port 4 of the router and assign the PC an IP address on the same subnet as the router. From the PC, launch a Web browser and go to <http://10.10.200.1>. A login screen will appear. Enter login credentials provided by Cablevision in the Optimum Business SIP Trunk Set-UP guide provided by the Optimum Business technician at install. A copy of the set-up guide can also be downloaded from www.optimumbusiness.com/sip.

The **LAN Settings** screen will appear as shown below. Enter the following:

- **IP Address:** Enter the IP address assigned to LAN port 1 which will be used for SIP and RTP traffic. This IP address should be on the same subnet as the Avaya IP Office.
- **Subnet Mask:** Enter the subnet mask appropriate for the network where port 1 will be connected.

Default values were used for all other settings. Click **Submit**. Optionally, the **DHCP Server** functionality can be enabled. However, the compliance test did not use this functionality. Once complete, connect the voice network to LAN port 1.

The screenshot shows the 'LAN Settings' page of the Optimum Business web interface. On the left is a 'Configuration Menu' with links for Customer, LAN Settings, SIP Trunk Configuration, Diagnostics, and System. The main content area is titled 'LAN Settings' and includes a 'Help' link. Under 'LAN Interface Settings', there is a note that the IP address must be on the same subnet as the PBX. The IP Address field is set to 10.32.128.21 and the Subnet Mask field is set to 255.255.255.0. Under 'DHCP Server', the 'Enable DHCP Server' checkbox is unchecked, and the Start and End Address fields are set to 10.32.128.100 and 10.32.128.200 respectively. A red warning message states: 'The voice network must be connected to LAN port 1. LAN port 4 is dedicated to management and maintenance.' At the bottom are 'Submit' and 'Reset' buttons.

Optimum Business™

Configuration Menu

- ◆ Customer
 - ▶ [LAN Settings](#)
 - ▶ [SIP Trunk Configuration](#)
 - ▶ [Diagnostics](#)
 - ▶ [System](#)

LAN Settings [Help](#)

LAN Interface Settings:

This address must be on the same subnet as the IP Address of the PBX

IP Address:

Subnet Mask:

DHCP Server:

Enable DHCP Server: ☐

Start Address:

End Address:

The voice network must be connected to LAN port 1. LAN port 4 is dedicated to management and maintenance.

Select the **SIP Trunk Settings** link from the navigation menu in the left pane. In the right pane, select **Avaya IP Office** (latest version) from the pull-down menu for the **Select your PBX** field. Select the connection option labeled **Passive connection using the local, private IP address of the PBX**. In the **PBX address** field, enter the IP address of the Avaya IP Office. Click **Submit**.



The screenshot shows the 'SIP Trunk Configuration' page of the Optimum Business web interface. On the left is a 'Configuration Menu' with links for Customer, LAN Settings, SIP Trunk Configuration (selected), Diagnostics, and System. The main area is titled 'SIP Trunk Configuration' and includes a 'Help' link. It features a 'Select your PBX:' dropdown menu set to 'Avaya IP Office v7.0 (18)'. Below this are two radio button options: 'Passive connection using the local, private IP address of the PBX' (selected) and 'Active connection using registration'. The passive option includes a note about subnet matching and a 'PBX Address:' text field containing '10.32.128.20'. The active option includes 'User Id:' and 'Password:' text fields. At the bottom are 'Submit' and 'Reset' buttons.

6. General Test Approach and Test Results

This section describes the general test approach used during compliance testing and the test results.

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to Optimum Voice SIP Trunking. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 1.1**.

Optimum Voice SIP Trunking passed compliance testing.

7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is *Idle* for each channel.

The screenshot shows the Avaya IP Office System Status application. The left pane lists various system components, with 'Line: 24' selected under 'Trunks (11)'. The main pane displays the 'SIP Trunk Summary' for Line 24. The summary includes the following information:

- Peer Domain Name: 10.32.128.21
- Gateway Address: 10.32.128.21
- Line Number: 24
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: Auto
- Silence Suppression: Off
- SIP Trunk Channel Licences: Unlimited
- SIP Trunk Channel Licences in Use: 0
- SIP Device Features: 0%

Below the summary is a table showing the status of the channels:

Chann	UR	Call	Current	Time in	Remote	Code	Conne	Caller	Other	Part	Directi	Round	Receiv	Receiv	Transr	Tr
Numbe	Grc	Ref	State	State	Address	Type	Dialed	on Call	of Call	Delay	Jitter	Loss	Fi	Jitter	Lo	
1			Idle	18:34:31												
2			Idle	18:34:31												
3			Idle	18:34:31												
4			Idle	18:34:31												
5			Idle	18:34:31												
6			Idle	18:34:31												
7			Idle	18:34:31												
8			Idle	18:34:31												

At the bottom of the application, there are buttons for 'Trace', 'Trace All', 'Pause', 'Ping', 'Call Details', 'Print...', and 'Save As...'. The status bar at the bottom right shows the time as 10:56:17 AM and the system is Online.

- Select the **Alarms** tab and verify that no alarms are active on the SIP line.

The screenshot shows the 'Alarms' tab selected in the Avaya IP Office System Status application. The title of the tab is 'Alarms for Line: 24 SIP 10.32.128.21'. Below the title is a table with the following columns:

Last Date Of Error	Occurrences	Error Description
--------------------	-------------	-------------------

The table is currently empty, indicating that there are no active alarms.

- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Verify that a phone connected to PSTN can successfully place a call to the Avaya IP Office with two-way audio.

8. Conclusion

The Optimum Voice SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and the Cablevision Optimum Voice SIP Trunking as shown in **Figure 1**.

9. Additional References

- [1] *IP Office 6.0 Documentation CD*, February 2010.
- [2] *IP Office Installation*, Document number15-601042, May 2010.
- [3] *IP Office Manager*, Document number15-601011, May 2010.
- [4] *System Status Application*, Document number15-601758, February 2010.

Product documentation for Avaya products may be found at <http://support.avaya.com>.
Product documentation for Optimum Voice SIP Trunking is available from Cablevision at www.optimumbusiness.com/sip.

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