



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Megapath SIP Trunking with Avaya IP Office 8.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Megapath and Avaya IP Office 8.1.

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

Megapath 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 service provider Megapath and an Avaya IP Office solution via an EdgeMarc SIP ALG (Application Layer Gateway). In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.1, Avaya embedded Voicemail, Avaya IP Office Softphone, Avaya H.323, Avaya SIP, digital and analog endpoints.

The Megapath SIP Trunking service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office connecting to the EdgeMarc SIP ALG (Application Layer Gateway) which is delivered as part of service provider. Then, the EdgeMarc SIP ALG connects to Megapath SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**. Note that: NAT devices added between EdgeMarc SIP ALG and the Megapath network should be transparent to the SIP signaling.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

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

- ◆ Incoming PSTN calls to various phone types. Phone types included H.323, SIP, 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, SIP, 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 the Avaya IP Office Softphone.
- ◆ Inbound and outbound long hold time call stability.
- ◆ Various call types including: local, long distance, international, inbound toll-free, outbound toll-free, operator assisted, 411, and 911 services.
- ◆ Codec G.711U and G.729A.
- ◆ Caller number/ID presentation.

- ◆ Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- ◆ DTMF transmission using RFC 2833.
- ◆ Voicemail navigation for inbound and outbound calls.
- ◆ Telephony features such as hold and resume, terminator declined, do not disturb, call waiting, call park and pickup, transfer, and conference.
- ◆ Use of SIP re-INVITE for call transfer to PSTN.
- ◆ FAX using G.711.
- ◆ Off-net call forwarding (Megapath supports Diversion Header).
- ◆ Registration and Authentication.
- ◆ Twinning to mobile phones on inbound calls.

2.2. Test Results

All the applicable test cases were executed. Megapath SIP Trunking passed compliance testing.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit:

<http://support.avaya.com>

For technical support on the Megapath system, please contact customer service or visit:

<http://www.megapath.com/voice/sip-trunking/>

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to Megapath SIP Trunking service through the public IP network. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

Located at the enterprise site are an EdgeMarc SIP ALG and an Avaya IP Office 500v2 with the MOD DGTL STA16 expansion module which provides connections for 16 digital stations to the PSTN, and the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The voicemail service is embedded on Avaya IP Office. The LAN2 port of Avaya IP Office is connected to LAN port of EdgeMarc SIP ALG while the WAN port of EdgeMarc SIP ALG is connected to the public IP network. Endpoints include an Avaya 9600 Series IP Telephone (with H.323 firmware), Avaya 1408D Digital Telephones, an Avaya Symphony 2000 Analog Telephone and an Avaya IP Office Softphone H323/SIP. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer Avaya IP Office.

Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user's phones will also ring and can be answered at the configured mobile phones.

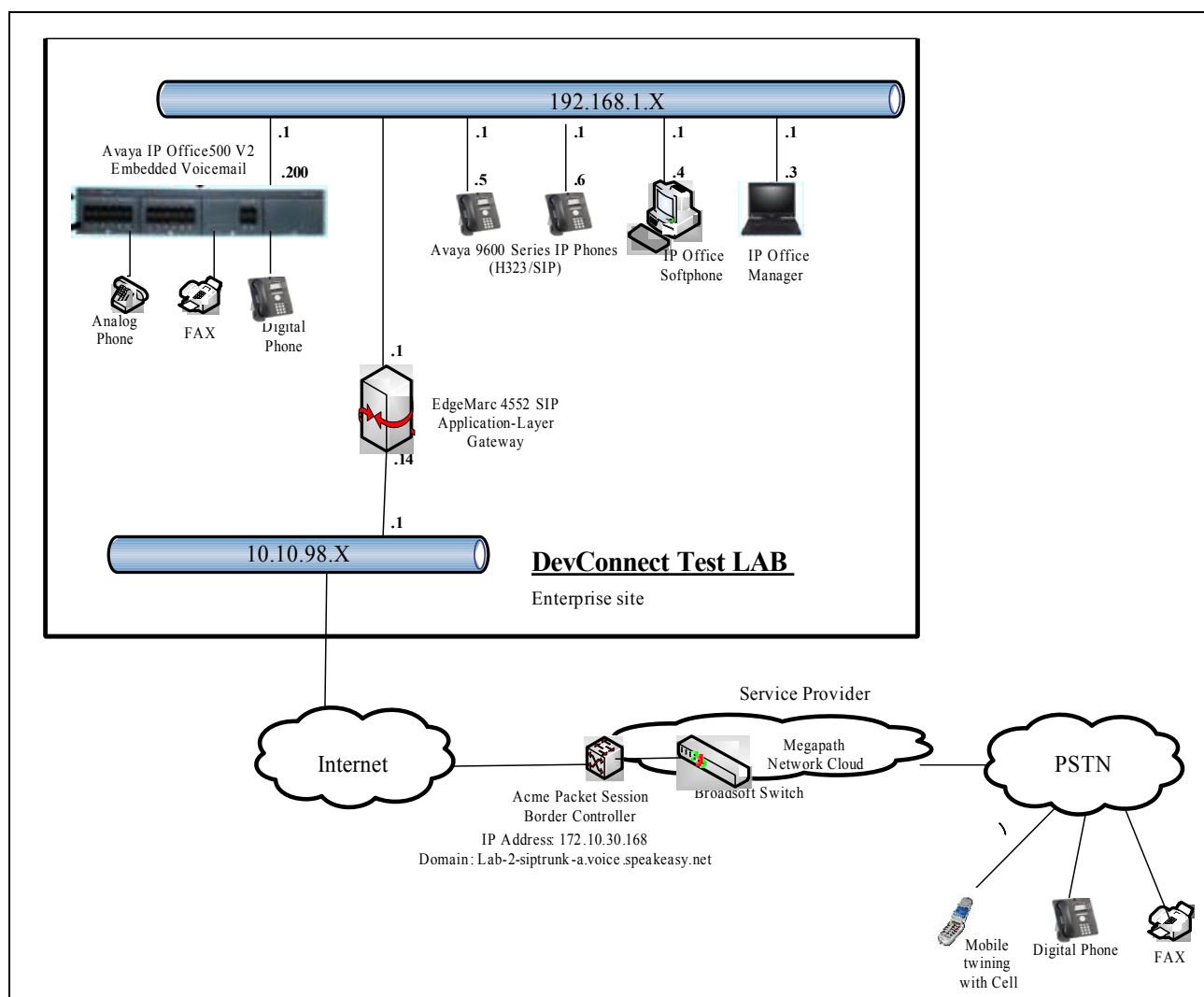


Figure 1: Test Configuration for Avaya IP Office with Megapath SIP Trunking Service

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 6 + N digits to send digits across the SIP trunk to Megapath. The short code of 6 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Megapath. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 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, Megapath SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the Avaya IP Office 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 the Avaya IP Office must be allowed to pass through these devices.

4. 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 500v2	8.1 (43)
Avaya IP Office DIG DCP*16 V2	8.1 (43)
Avaya IP Office Ext Card Phone 8	8.1
Avaya IP Office Manager	10.1 (43)
Avaya 9640 IP Telephone (H.323)	Avaya one-X Deskphone Edition S3.110b
Avaya Digital Telephones (1408D)	N/A
Avaya Symphony 2000 Analog Telephone	N/A
Avaya IP Office Softphone	3.2.3.15 64595
HP Officejet 4500	N/A
Megapath Components	
Equipment	Release
SBC Acme Packet SBC Net-Net	C51.1.1 Patch 21
Broadsoft	R17 SP4

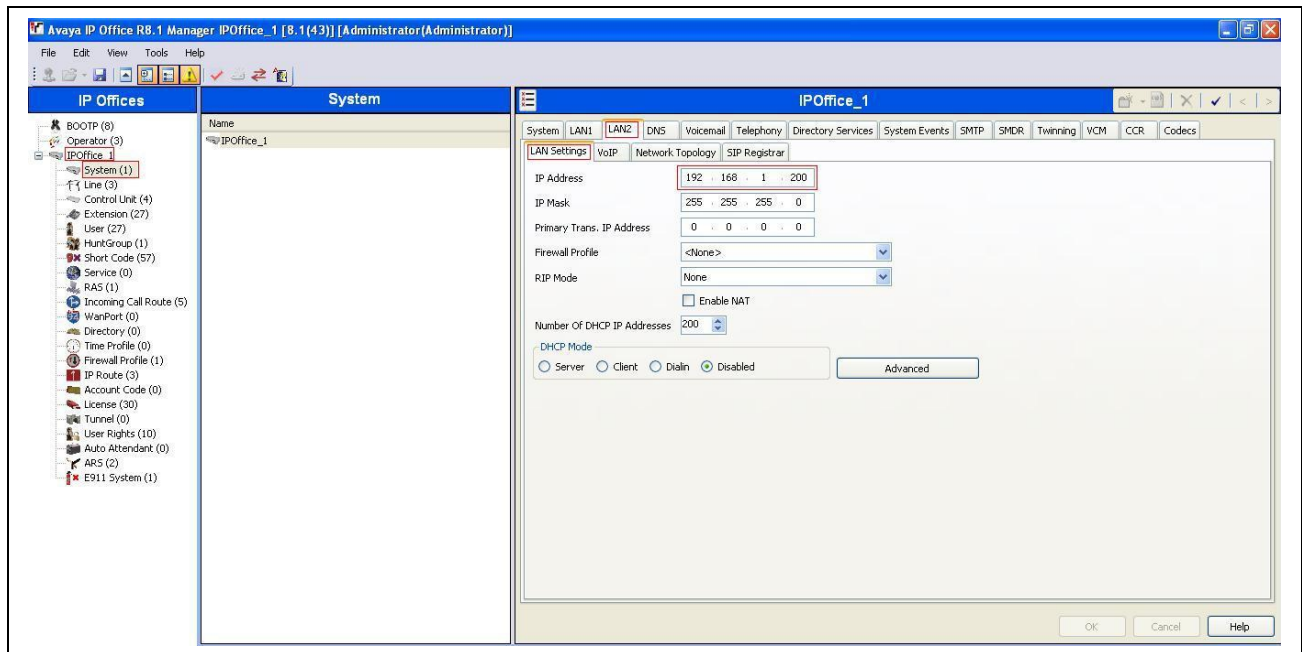
Note: Testing was performed with IP Office 500 R8.1, but it also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints or trunks.

5. Configure IP Office

This section describes the Avaya IP Office configuration to support connectivity to Megapath SIP Trunking service. 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 shown in the next section. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the service provider (such as the LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. LAN2 Settings

In the sample configuration, the **IPOffice_1** was used as the system name and the LAN2 port was used to connect to LAN port of the EdgeMarc SIP ALG. To access the LAN2 settings, first navigate to **System (1) → IPOffice_1** in the Navigation and Group Panes and then navigate to the **LAN2 → LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office LAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.



Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the 9600-Series IP Telephones used in the sample configuration. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Megapath. The **SIP Registrar Enable** box is checked to allow Avaya IP Office Softphone usage. 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 **LAN2**. The **Enable RTCP Monitoring On Port 5005** is checked. 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 Service 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.

The screenshot shows the IPOffice_1 configuration window with the VoIP tab selected. The following settings are visible:

- H.323 Gatekeeper Enable**: ☒
- SIP Trunks Enable**: ☒
- SIP Registrar Enable**: ☒
- H.323 Auto-create Extn**: ☒
- H.323 Auto-create User**: ☐
- H.323 Remote Extn Enable**: ☐
- Enable RTCP Monitoring On Port 5005**: ☒
- RTP Port Number Range**:
 - Port Range (Minimum): 49152
 - Port Range (Maximum): 53246
- DiffServ Settings**:

88	DSCP(Hex)	FC	DSCP Mask (Hex)	88	SIG DSCP (Hex)
46	DSCP	63	DSCP Mask	34	SIG DSCP
- DHCP Settings**:
 - Primary Site Specific Option Number (SSON): 176
 - Secondary Site Specific Option Number (SSON): 242
 - VLAN: Not Present
 - 1100 Voice VLAN Site Specific Option Number (SSON): 232
 - 1100 Voice VLAN IDs:

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**. With this configuration, STUN will not be used.
- ◆ Set the **Binding Refresh Time (seconds)** to **60**. 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. (Refer to **Section 5.10**)
- ◆ Set **Public IP Address** to the IP address of the Avaya IP Office LAN2 port. **Public Port** is set to **5060**.
- ◆ All other parameters should be set according to customer requirements.

The screenshot shows the 'IPOffice_1' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields and values:

Field	Value
STUN Server IP Address	192 . 168 . 10 . 13
Firewall/NAT Type	Open Internet
Binding Refresh Time (seconds)	60
Public IP Address	192 . 168 . 1 . 200
Public Port	5060
STUN Port	3478

Below the fields are two buttons: 'Run STUN' and 'Cancel'. At the bottom, there is a checkbox labeled 'Run STUN on startup' which is currently unchecked.

In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with Megapath SIP Trunking service, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, **U-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. Set **Hold Timeout (secs)** to **600**.

The screenshot shows the 'IPOffice_1' configuration window with the 'Telephony' tab selected. The 'Telephony' sub-tab is also active. The 'Companding Law' section is highlighted, showing 'U-Law' selected for the Switch and 'U-Law Line' selected for the Line. The 'Hold Timeout (secs)' is set to 600. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked.

Setting	Value
Default Outside Call Sequence	Normal
Default Inside Call Sequence	Ring Type 1
Default Ring Back Sequence	Ring Type 2
Restrict Analogue Extension Ringer Voltage	<input type="checkbox"/>
Dial Delay Time (secs)	4
Dial Delay Count	0
Default No Answer Time (secs)	15
Hold Timeout (secs)	600
Park Timeout (secs)	300
Ring Delay (secs)	5
Call Priority Promotion Time (secs)	Disabled
Default Currency	USD
Default Name Priority	Favor Trunk
Companding Law Switch	U-Law
Companding Law Line	U-Law Line
DSS Status	<input type="checkbox"/>
Auto Hold	<input checked="" type="checkbox"/>
Dial By Name	<input checked="" type="checkbox"/>
Show Account Code	<input checked="" type="checkbox"/>
Inhibit Off-Switch Forward/Transfer	<input type="checkbox"/>
Restrict Network Interconnect	<input type="checkbox"/>
Drop External Only Impromptu Conference	<input type="checkbox"/>
Visually Differentiate External Call	<input type="checkbox"/>
Unsupervised Analog Trunk Disconnect Handling	<input type="checkbox"/>
High Quality Conferencing	<input checked="" type="checkbox"/>

5.3. Twinning Calling Party Settings


When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect 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, as shown below. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.4**).

If **Send original calling party information for Mobile Twinning** on the **System→Twinning** tab is optioned, the setting of the second parameter is ignored and Avaya IP Office will send the following in the SIP From Header:

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

If this option is unchecked, the value sent in the SIP From header is determined by the setting of the second parameter mentioned above.

For the compliance test, the **Send original calling party information for Mobile Twinning** box in 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 titled 'IPOffice_1'. The 'Twinning' tab is selected and highlighted with a red box. Within this tab, the checkbox 'Send original calling party information for Mobile Twinning' is checked and also highlighted with a red box. Below this checkbox, the label 'Calling party information for Mobile Twinning' is followed by an empty text input field.

5.4. Administer SIP Line

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

- ◆ Set **ITSP Domain Name** to the enterprise domain so that IP Office uses this domain as the host portion of the SIP URI in SIP headers such as the From header.
- ◆ Check the **In Service** box.
- ◆ Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- ◆ Set **Call Routing Method** to **Request URI**.
- ◆ Check **Caller ID from From header** box.
- ◆ Set **Send Caller ID** to **Diversion Header**. For the compliance test, this parameter was used for call forwarding and it was ignored in Mobility Twinning since **Send original calling party information for Mobile Twinning** is optioned in **Section 5.3**.
- ◆ Set **Association Method** to **By Source IP address**.
- ◆ The area of the screen entitled **REFER Support** is used to enable/disable SIP REFER for call transfers. The default values of “Auto” for **Incoming** and **Outgoing** effectively disable the use of SIP REFER.
- ◆ Set **UPDATE Supported** to **Never**. This method is not supported by Megapath SIP Trunk Interface at this time.
- ◆ Default values may be used for all other parameters.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows the hierarchy: IP Office (8) > Operator (3) > JPOffice_1 > System (1) > Line (3) > Line 17. The 'Line' table in the center lists three lines: Line 1 (PRI 24 Universal, T1), Line 2 (PRI 24 Universal, T1), and Line 17 (SIP Line). The right pane shows the 'SIP Line - Line 17' configuration tab. The 'SIP Line' sub-tab is active, showing various configuration fields. The 'ITSP Domain Name' is set to 'megapathvoice.com'. The 'In Service' checkbox is checked. The 'Check OOS' checkbox is checked. The 'Call Routing Method' is set to 'Request URI'. The 'Caller ID from From header' checkbox is checked. The 'Send Caller ID' is set to 'Diversion Header'. The 'Association Method' is set to 'By Source IP address'. The 'REFER Support' section has 'Incoming' and 'Outgoing' both set to 'Auto'. The 'UPDATE Supported' is set to 'Never'. The 'Send From In Clear' checkbox is unchecked. The 'User-Agent and Server Headers' field is empty. The 'Prefix', 'National Prefix', 'Country Code', and 'International Prefix' fields are empty. The 'Originator number for forwarded and twinning calls' field is empty. The 'Name Priority' is set to 'System Default'. The 'Send From In Clear' checkbox is unchecked. The 'User-Agent and Server Headers' field is empty. The 'OK', 'Cancel', and 'Help' buttons are at the bottom right.

Line Number	Line Type	Line SubType
1	PRI 24 (Universal)	T1
2	PRI 24 (Universal)	T1
17	SIP Line	

SIP Line - Line 17

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

Line Number: 17

ITSP Domain Name: megapathvoice.com

Prefix:

National Prefix:

Country Code:

International Prefix:

Send Caller ID: Diversion Header

Association Method: By Source IP address

REFER Support

Incoming: Auto

Outgoing: Auto

UPDATE Supported: Never

OK Cancel Help

Select the **Transport** tab. The **ITSP Proxy Address** is set to the EdgeMarc SIP ALG LAN IP address. As shown in **Figure 1**, this IP Address is **192.168.1.1**. In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Megapath, in this case the well known SIP port of **5060** was used. The **Use Network Topology Info** parameter is set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters retain default values in the screen below.

Line Number	Line Type	Line SubType
1	PRI 24 (Universal)	T1
2	PRI 24 (Universal)	T1
17	SIP Line	

SIP Line - Line 17

SIP Line | **Transport** | SIP URI | VoIP | T38 Fax | SIP Credentials

ITSP Proxy Address: 192.168.1.1

Network Configuration

Layer 4 Protocol: UDP | Send Port: 5060

Use Network Topology Info: LAN 2 | Listen Port: 5060

Explicit DNS Server(s): 0 . 0 . 0 . 0 | 0 . 0 . 0 . 0

Calls Route via Registrar: ☒

Separate Registrar:

A SIP Credentials entry must be created for Digest Authentication used by Megapath SIP trunking service to authenticate calls from the enterprise to the PSTN. To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the Add button and the New Channel area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the bottom of the screen, the **Edit Channel** area will be opened. In the example screen below, a previously configured entry is edited. The entry was created with the parameters shown below:

- ◆ Set **User name/ Authentication Name/Contact/Password** to the values provided by the service provider.
- ◆ **Expiry (mins)** is set to **60**.
- ◆ Check the **Registration required** option (Megapath requires registration for Digest Authentication).

The screenshot shows a configuration window titled "SIP Line - Line 17". It has several tabs: "SIP Line", "Transport", "SIP URI", "VoIP", "T38 Fax", and "SIP Credentials". The "SIP Credentials" tab is selected. Below the tabs is a table with the following data:

Index	UserName	Authentication Name	Contact	Expiry (mins)	Register
1	7039399245	7039399245	7039399245	60	True

To the right of the table are buttons: "Add...", "Remove", and "Edit...". The "Edit..." button is highlighted with a red box. Below the table is a dialog box titled "Edit SIP Credentials". It contains the following fields:

- User name: 7039399245
- Authentication Name: 7039399245
- Contact: 7039399245
- Password: *****
- Expiry (mins): 60 (with a spinner box)
- Registration required: ☒

At the bottom right of the dialog are "OK" and "Cancel" buttons.

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab; click the **Add** button and the New Channel area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- ◆ Set **Local URI**, **Contact** and **Display Name** to **Use 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 5.6**.
- ◆ Set **PAI** to **None**.
- ◆ Associate this line with an incoming line group 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. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- ◆ Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- ◆ Set **Registration** to **1:7039399245**. This is the account credentials previously configured on the line's SIP Credentials tab.

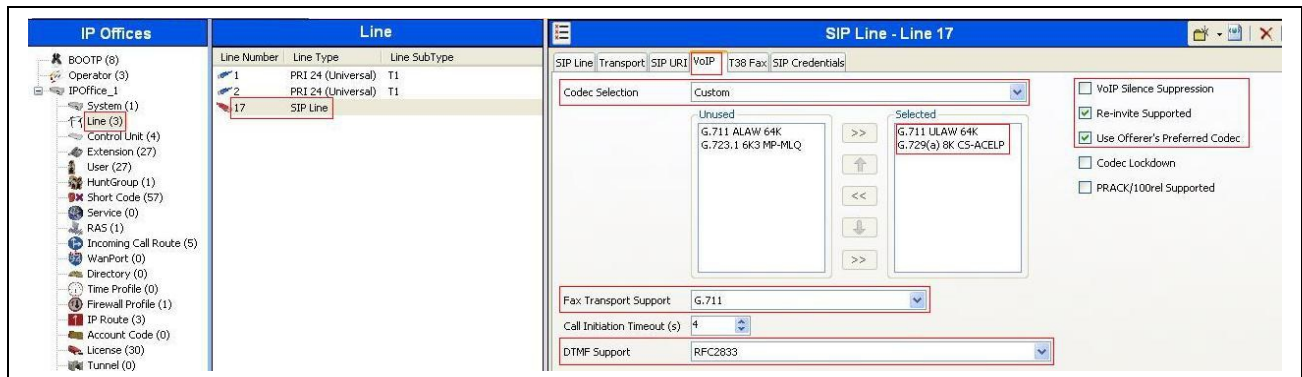
The screenshot shows the 'SIP Line - Line 17' configuration window. The 'SIP URI' tab is selected. Below the tabs is a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The table contains one entry with Channel 1, Groups 17 17, Via 1..., Local URI N..., Contact 1: 7039..., and Max Calls 10. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is an 'Edit Channel' dialog box. The dialog box contains the following fields and values:

Via	192.168.1.200
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	None
Registration	1: 7039399245
Incoming Group	17
Outgoing Group	17
Max Calls per Channel	10

At the bottom right of the dialog box 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:

- ✦ The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. Selecting **G.711 ULAW 64K** and **G.729(a) 8K CS –ACELP** codecs causes Avaya IP Office to include these codecs, which are supported by the Megapath SIP Trunking service, in the Session Description Protocol (SDP) offer, in that order.
- ✦ Set **Fax Transport Support** to **G.711** from the pull-down menu.
- ✦ Set the **DTMF Support** field to **RFC2833** from the pull-down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- ✦ Uncheck the **VoIP Silence Suppression** box.
- ✦ Check the **Re-invite Supported** box.
- ✦ Check **Use Offerer's Preferred Codec**.
- ✦ Default values may be used for all other parameters.



5.5. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered “6N;” short code used in the test configuration.

- ✦ In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **6N;**, this short code will be invoked when the user dials 6 followed by any number.
- ✦ Set **Feature** to **Dial**. This is the action that the short code will perform.
- ✦ Set **Telephone Number** to **N”@megapathvoice.com”**. 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 host part following the “@” is the domain of the service provider network.
- ✦ Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound call.

◆ Set Locale to United States (US English).

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows the hierarchy: BOOTP (8), Operator (3), IPOffice_1, System (1), Line (3), Control Unit (4), Extension (27), User (27), HuntGroup (1), Short Code (58), Service (0), RAS (1), Incoming Call Route (5), WanPort (0), and Directory (0). The 'Short Code' table is shown with columns for Code, Telephone Number, and Feature. The '6N;: Dial' configuration is highlighted in the table. The configuration details for '6N;: Dial' are shown on the right, including Code (6N;), Feature (Dial), Telephone Number (N"@megapathvoice.com"), Line Group ID (17), Locale (United States (US English)), and Force Account Code (unchecked).

Code	Telephone Number	Feature
*40	1	Relay Off
*53*N#	N	Call Pickup Members
*57*N#	N	Forward Busy Number
*70*N#	N	Dial Physical Extn by Num
*71*N#	N	Dial Physical Extn by Id
9000	"MAINTENANCE"	Relay On
*91N;	N".1"	Record Message
*92N;	N".2"	Record Message
*DSSN	";[0]151/ERR - "N	Display Msg
*SDN	";[0]151/ERR - "N	Display Msg
*SKN	";[0]151/ERR - "N	Display Msg
6N;	N"@megapathvoice.com"	Dial
7N	N	Dial

The simple "6N;" short codes illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 7N is illustrated for access to ARS. When the Avaya IP Office user dials 7 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID 50: Main**, configurable via ARS. See **Section 5.8** for example ARS route configuration for **50: Main** as well as a backup route.

The screenshot displays the Avaya IP Office configuration interface for the '7N: Dial*' short code. The configuration details are shown on the right, including Code (7N), Feature (Dial), Telephone Number (N), Line Group ID (50: Main), Locale (United States (US English)), and Force Account Code (unchecked). The 'Line Group ID' field is highlighted with a red box.

5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **H323 9246**. 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. They also 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 5.4**). The example below shows the settings for user **H323 9246**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Megapath. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.

The screenshot displays the Avaya SIP configuration interface. On the left, the 'IP Offices' pane shows a tree structure with 'User (27)' selected. The center pane shows a list of users with 'H323 9246' highlighted. The right pane shows the configuration for 'H323 9246 : 9246' with the 'SIP' tab selected. The 'SIP' tab contains the following fields:

Field	Value
SIP Name	7039399246
SIP Display Name (Alias)	H323 9246
Contact	7039399246

Below the fields is an 'Anonymous' checkbox, which is currently unchecked.

One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for User **H323 9246**. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **616139675205**. Other options can be set according to customer requirements.

H323 9246: 9246*

Voice Recording Button Programming Menu Programming **Mobility** Phone Manager Options Hunt Group Membership Announcements

☐ Internal Twinning

Twinned Handset: <None>

Maximum Number of Calls: 1

☐ Twin Bridge Appearances

☐ Twin Coverage Appearances

☐ Twin Line Appearances

☒ **Mobility Features**

☒ **Mobile Twinning**

Twinned Mobile Number (including dial access code): 616139675205

Twinning Time Profile: <None>

Mobile Dial Delay (secs): 2

Mobile Answer Guard (secs): 0

☐ Hunt group calls eligible for mobile twinning

☐ Forwarded calls eligible for mobile twinning

☐ Twin When Logged Out

☐ one-X Mobile Client

☐ Mobile Call Control

☐ Mobile Callback

5.7. 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. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group 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 5.4**.
- ◆ Set the **Incoming Number** to the incoming number on which this route should match.
- ◆ Set **Locale** to **United States (US English)**.
- ◆ Default values can be used for all other fields.

The screenshot displays the 'Incoming Call Route' configuration window. The left pane shows the 'IP Offices' tree with 'Incoming Call Route (5)' selected. The center pane shows a table of existing routes. The right pane shows the configuration for a new route with the title '17 7039399246'.

Line Group ID	Incoming Number	Destination
0		DialIn
17	7039399246	9246 H323 9246
17	2063882468	2468 SIP2468
17	2063882469	2469 Digital 2469
17	7039399245	9245 FAX 9245

Configuration fields for the new route:

- Bearer Capability: Any Voice
- Line Group ID: 17
- Incoming Number: 7039399246
- Incoming Sub Address: (empty)
- Incoming CLI: (empty)
- Locale: United States (US English)
- Priority: 1 - Low
- Tag: (empty)
- Hold Music Source: System Source

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 703-939-9246 on line 17 are routed to extension **9246 H323 9246**.

The screenshot displays the 'Incoming Call Route' configuration window with the 'Destinations' tab selected. The left pane and center pane are the same as in the previous screenshot. The right pane shows the 'Destinations' configuration for the route '17 7039399246'.

TimeProfile	Destination	Fallback Extension
Default Value	9246 H323 9246	

5.8. ARS and Alternate Routing

While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is shown here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, ARS can be used rather than the simple **6N**; short code approach documented in **Section 5.5**. With ARS, a secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish these call behaviors.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named **Main**. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route, and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure including items like BOOTP, Operator, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, License, Tunnel, User Rights, Auto Attendant, and ARS (2). The 'ARS' group is selected, and the 'Main' route is highlighted in the 'ARS' pane. The main configuration area is titled 'Main' and contains the following fields and sections:

- ARS Route Id:** 50
- Route Name:** Main
- Dial Delay Time:** System Default (4)
- Secondary Dial tone:** ☒ SystemTone
- Check User Call Barring:** ☒
- In Service:** ☒ (linked to **Out of Service Route:** 51: backup)
- Time Profile:** <None> (linked to **Out of Hours Route:** <None>)
- Code Table:**

Code	Telephone Number	Feature	Line Group ID
11	911	Dial Emergency	0
911	911	Dial Emergency	0
0N;	0N	Dial 3K1	0
1N;	1N'megaphavoice.com'	Dial 3K1	17
XN;	N	Dial 3K1	0
XXXXXXXXXXN	N	Dial 3K1	0
- Alternate Route Priority Level:** 3
- Alternate Route Wait Time:** 30
- Alternate Route:** 51: backup

Buttons for 'Add...', 'Remove', and 'Edit...' are located to the right of the code table. At the bottom right are 'OK', 'Cancel', and 'Help' buttons.

Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., **7N** in **Section 5.5**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 7-1-613-967-5205, the call would be directed to Line Group 17, the SIP Line configured and described in these Application Notes. If Line Group 17 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

The following screen shows an example ARS configuration for the route named **backup** with ARS Route ID 51. Continuing the example, if the user dialed 7-1-613-967-5205, and the call could not be routed via the primary route **50: Main** described above, the call will be delivered to this **backup** route. Per the configuration shown below, the call will be delivered to Line Group 1, using an analog trunk connecting the Avaya IP Office to the PSTN as a backup connection. In this case, the originally dialed number (sans the short code 7) will be dialed as is through the analog/PRI trunk to the PSTN.

Code	Telephone Number	Feature	Line Group ID
11	911	Dial Emergency	0
911	911	Dial Emergency	0
1N	1N	Dial	0

In the testing associated with the configuration, calls were successfully delivered to SIP Line 17 via the primary ARS route **50: Main** or to the analog/PRI trunk via the backup ARS route shown above. When the primary route experiences a network outage, Avaya IP Office successfully routed the call via the backup route.

5.9. Privacy/Anonymous Calls

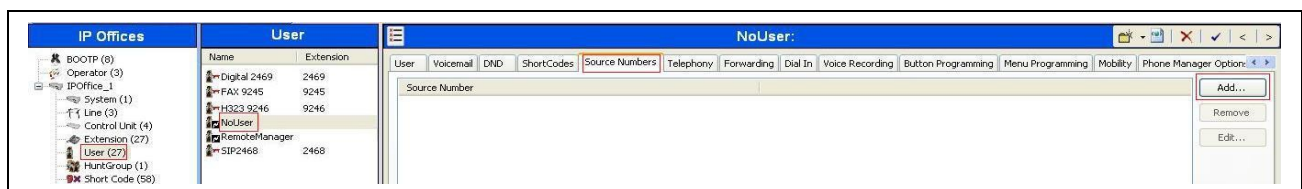
For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “restricted” and “anonymous” respectively and enable privacy:id. On the other hand, by default Avaya IP Office will send the P-Preferred-Identity (PPI) or can be configured to use the P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. For the compliance test, Megapath does not use either PPI or PAI for the purposes of privacy.

5.10. SIP OPTIONS

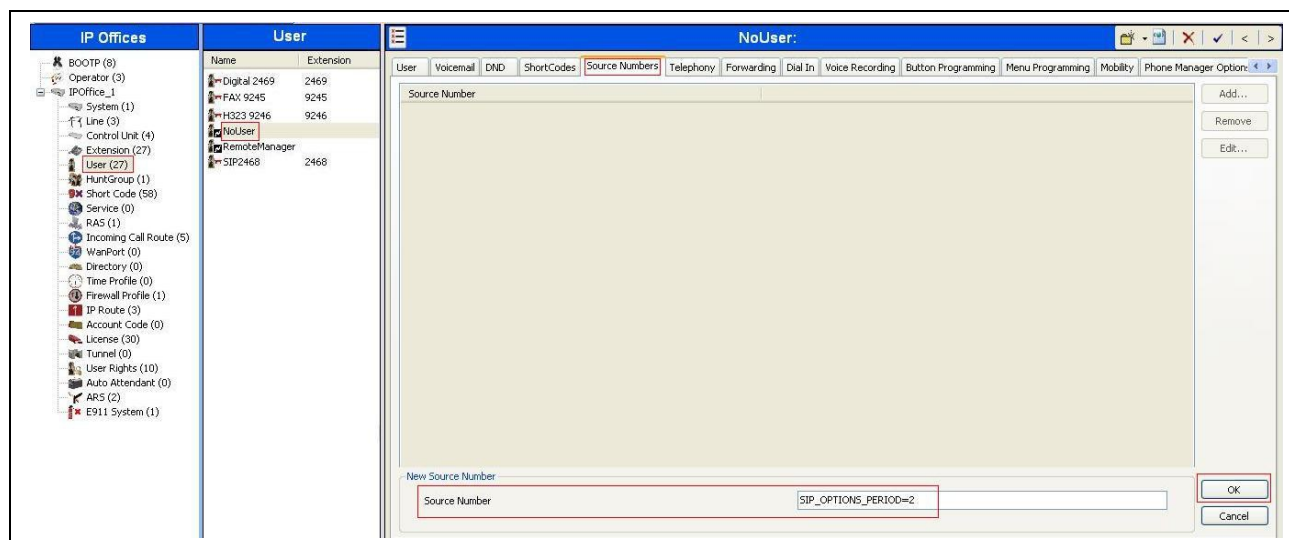
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD** 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**.

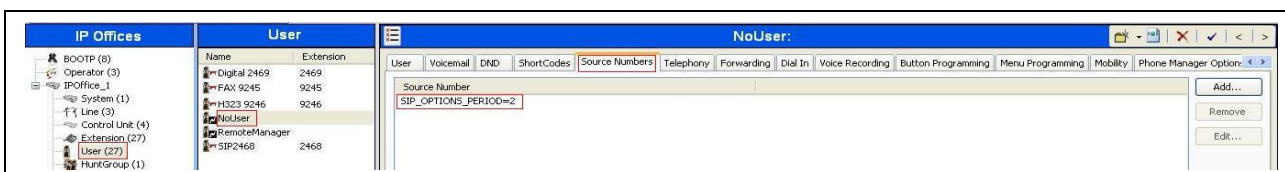
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an **OPTIONS** period of 1 minute was desired. The **Binding Refresh Time** was set to **60** seconds (1 minute) in **Section 5.1**. The **SIP_OPTIONS_PERIOD** was set to **2** minutes. Avaya IP Office chose the **OPTIONS** period as the smaller of these two values (1 minute).



5.11. 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.

6. EdgeMarc SIP ALG Configuration

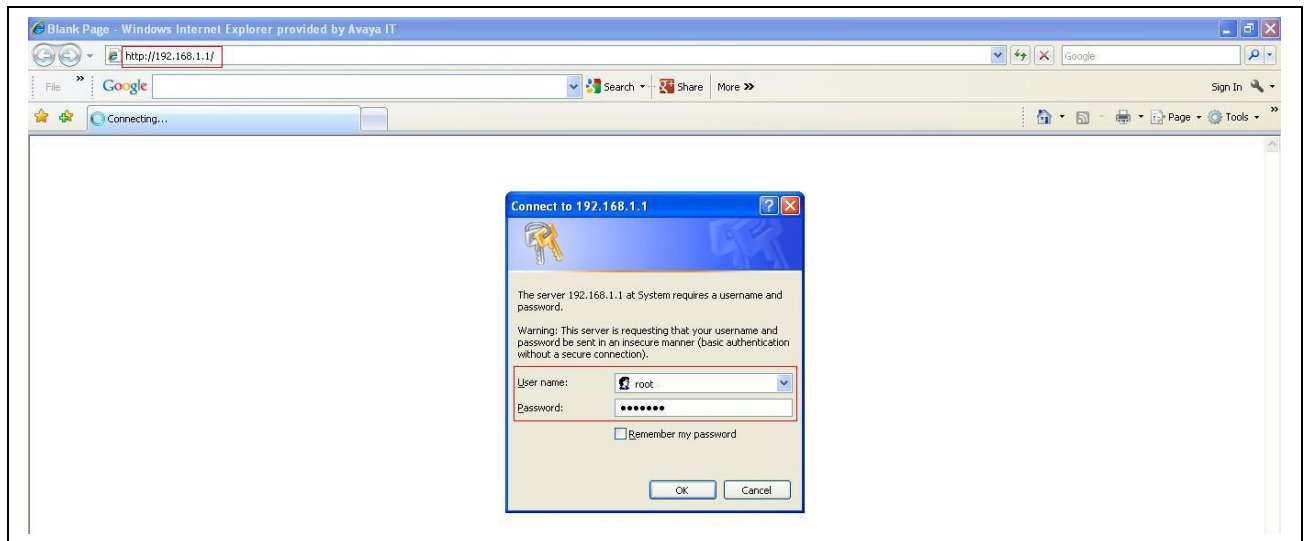
The EdgeMarc SIP ALG gateway is configured to manage all SIP signaling and provides voice quality management. All data traffic also traverses the EdgeMarc SIP ALG. It is part of the Megapath SIP trunk service and Megapath will provide it to the customer when the customer orders the Megapath SIP trunk service. Megapath manages it and the end-customer does not manage it.

6.1. EdgeMarc SIP ALG Login

The EdgeMarc SIP ALG was configured with a local LAN address of 192.168.1.1 and a subnet mask of 255.255.255.0. A personal computer is configured with Ethernet IP address assigned to any address other than 192.168.1.1 in the same subnet mask, for example 192.168.1.2

Launch a web browser on personal computer and enter the following URL: <http://192.168.1.1> and hit enter.

The following login window should appear:



Enter **User name** and **Password** field.

Click OK and the system page should be appeared next.

6.2. EdgeMarc SIP ALG Network Configuration

From the **Configuration Menu**, select **Network** menu option.

Under **Network**, input the public and private networks as followings:

- ♦ **LAN Interface Settings:**
 - **IP Address:** 192.168.1.1.
 - **Subnet Mask:** 255.255.255.0.
- ♦ **WAN Interface Settings: Static IP.**
 - **IP Address:** 10.10.98.14 (Provide this IP Address to service provider to set up the connectivity).
 - **Subnet Mask:** 255.255.255.192.
- ♦ **Network Settings:**
 - **Default Gateway:** 10.10.98.1.

Submit the changes.

Configuration Menu

- ♦ **Network**
 - Subinterfaces
 - VLAN Configuration
 - WAN VLAN Configuration
- ♦ DHCP Relay
- ♦ DHCP Server
- ♦ Firewall
- ♦ NAT
- ♦ PPTP Server
- ♦ Survivability
- ♦ PRI/Net Configuration
- ♦ Test UA
- ♦ Traffic Shaper
- ♦ VoIP ALG
- ♦ VPN
- ♦ WAN Link
- ♦ Redundancy
- ♦ System
 - Clients List
 - Dynamic DNS
 - File Download
 - File Server

6.3. EdgeMarc SIP ALG SIP Settings

From the **Configuration Menu**, select **VoIP ALG** menu option → **SIP** option.

Under **SIP Settings**, input the parameters as followings:

- ♦ **SIP Server Address:** **lab-2-siptrunk-a.voice.speakeasy.net** (This is fully-qualified domain name or IP address of the SIP gateway or session border controller provided by service provider).
- ♦ **SIP Server Port:** **5060**.
- ♦ Check **Use Custom Domain**.
- ♦ **SIP Server Domain:** **megapathvoice.com**.

Submit the changes.

Configuration Menu

- Network
- DHCP Relay
- DHCP Server
- Firewall
- NAT
- PPTP Server
- Survivability
- PRI/Net
- Configuration
- Test UA
- Traffic Shaper
- VoIP ALG
 - MGCP
 - SIP
 - Trunking
- VPN
- WAN Link
- Redundancy
- System
 - Clients List
 - Dynamic DNS
 - File Download
 - File Server

SIP Settings

Help

SIP protocol settings.

The SIP Server settings specify the address and port that all client traffic shall be forwarded to.

SIP Server Address:	lab-2-siptrunk-a.voice.s
SIP Server Port:	5060
Use Custom Domain:	<input checked="" type="checkbox"/>
SIP Server Domain:	megapathvoice.com

List of SIP Servers:

Enable Multi-homed Outbound Proxy Mode: ☐

Enable Transparent Proxy Mode: ☐

Limit Outbound to listed Proxies / SIP Servers: ☒

Limit Inbound to listed Proxies / SIP Servers: ☐

Stale Timer

The stale timer, if set, is used to automatically delete SIP clients that have not registered within the given time period.

Stale client time (m):

Registration Rate-Pacing parameters are available on the [Survivability page](#).

6.4. EdgeMarc SIP ALG SIP Trunking Configuration

From the **Configuration Menu**, select **VoIP ALG** menu option → **SIP** → **Trunking**.

In the **Add a trunking device** menu box:

- ♦ Set the **Action** to **Add new trunking device**.
- ♦ Input a recognizable **Name** for the trunking device: **Avaya IP Office**.
- ♦ Input the **IP Address** of the Avaya IP Office server: **192.168.1.200**.
- ♦ Input the **SIP Port** number of the Avaya IP Office server: **5060**.

Select the **Commit** button to create the SIP trunking device.

Configuration Menu

- ♦ [Network](#)
- ♦ [DHCP Relay](#)
- ♦ [DHCP Server](#)
- ♦ [Firewall](#)
- ♦ [NAT](#)
- ♦ [PPTP Server](#)
- ♦ [Survivability](#)
- ♦ [PRI/Net](#)
- ♦ [Configuration](#)
- ♦ [Test UA](#)
- ♦ [Traffic Shaper](#)
- ♦ [VoIP ALG](#)
 - [MGCP](#)
 - [SIP](#)
 - [Trunking](#)
- ♦ [VPN](#)
- ♦ [WAN Link](#)
- ♦ [Redundancy](#)
- ♦ [System](#)
 - [Clients List](#)
 - [Dynamic DNS](#)
 - [File Download](#)
 - [File Server](#)
 - [HTTPS Certificate](#)
 - [Network Information](#)

SIP Trunking

Configuration of SIP trunking devices.

SIP Trunking devices

A SIP trunking device can be a PSTN gateway, or similar device, that does not issue REGISTER messages. Calls will be forwarded to the device based on the dial-plan rules below.

If VLANs are enabled, the SIP trunking device needs to be in the same VLAN as defined in the VoIP ALG page.

SIP Trunking Devices			
Select: All None Action: Delete			
	Address	Port	Name
<input type="checkbox"/>	192.168.1.200	5060	Avaya IP Office

Add a trunking device

Action: [Add new trunking device](#)

Name:

Address:

Port:

[Commit](#) [Reset](#)

In the **Add a rule** menu box:

- ◆ Set the **Action** to **Add new rule**.
- ◆ Set **Type** to **Inbound**.
- ◆ Input **Pattern-match (if not default)**: **703**.
- ◆ Set **Trunking device** to **Avaya IP Office (192.168.1.200:5060)**.

Select the **Commit** button to create the rule.

Add a rule

Action:

Add new rule ▼

Type:

Inbound ▼

Call Party:

Called ▼

Default rule:

☐

Priority (inbound & redirect only):

☐

Pattern-match (if not default):

703

Strip digits:

0

Add string:

Use B2BUA:

☐

Use SIP proxy as secondary target:

☐

Trunking device:

Avaya IP Office (192.168.1.200:5060) ▼

Commit

Reset

The following screen shot is the list of rules for this testing:

Rules

Rules are used to forward and/or modify incoming and outgoing calls. There are 3 types of rules:

- Inbound: from server to trunking device
- Outbound: from trunking device to server
- Redirect: from local phone to trunking device (w/o routing to server)

Outbound rules can match against and/or modify either the calling or called number. Inbound and redirect rules operate on the called number only. Stripped and added digits always apply to the left-most digits of the DID.

Dial Rules									
Select: All None									Action: Delete
	Type	Party	PRI	Pattern - match	Strip	Add	b2bua	ss	Trunking device
<input type="checkbox"/>	Inbound			703					Avaya IP Office (192.168.1.200:5060)
<input type="checkbox"/>	Inbound			206					Avaya IP Office (192.168.1.200:5060)
<input type="checkbox"/>	Inbound			312					Avaya IP Office (192.168.1.200:5060)
<input type="checkbox"/>	Inbound			408					Avaya IP Office (192.168.1.200:5060)

7. Megapath SIP Trunking Configuration

Megapath is responsible for the configuration of Megapath SIP Trunking service. The customer must provide the EdgeMarc WAN interface IP address (**Session 6.2**) used to reach Megapath SIP Trunking service. Megapath will provide the customer the necessary information to configure the SIP connection between Avaya IP Office and Megapath. The provided information from Megapath includes:

- ◆ Fully Qualified Domain Name, IP address and port number of Acme Packet Session Border Controller used for signaling or media through any security.
- ◆ DID numbers.
- ◆ Megapath SIP trunking Specification.

8. Verification Steps

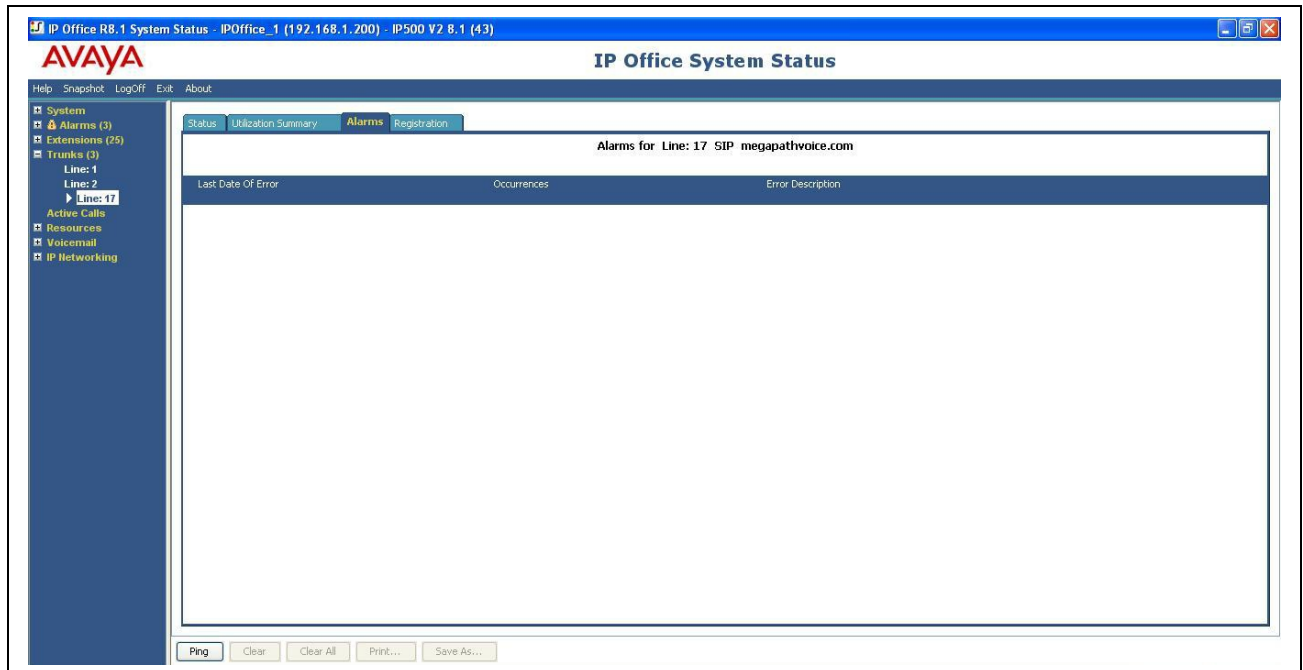
The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** for each channel (The below screen shot showed an active call at present time).

The screenshot displays the Avaya IP Office System Status application window. The title bar indicates the application is running on IP Office R8.1, connected to IPOffice_1 (192.168.1.200) and IP500 V2 8.1 (43). The main window is titled "IP Office System Status" and features a navigation pane on the left with options like System, Alarms (9), Extensions (25), Trunks (3), Line: 1, Line: 2, Line: 47, Active Calls, Resources, Voicemail, and IP Networking. The main content area shows the "Status" tab, which includes a "SIP Trunk Summary" section. This section displays various parameters such as Peer Domain Name (megapathvoice.com), Resolved Address (192.168.1.1), Line Number (17), Number of Administered Channels (10), Number of Channels in Use (1), Administered Compression (G711 Mu, G729 A), Silence Suppression (Off), SIP Trunk Channel Licenses (Unlimited), and SIP Trunk Channel Licenses in Use (1). A green progress indicator shows 0% usage. Below this summary is a table with 15 columns: Channel Number, URI, Call Ref, Current State, Time in State, Remote Media Address, Codec, Connection Type, Caller ID or Dialed Digits, Other Party on Call, Direction of Call, Round Trip Delay, Receive Jitter, Receive Packet Loss, Transmit Jitter, and Transmit Packet Loss. The table lists 10 channels, with the first channel (Channel 1) showing an "Outgoing Al..." state and a "Time in State" of "00:00:18". The remaining channels (2-10) are in an "Idle" state with a "Time in State" of "2 days 22:5...". At the bottom of the window, there are buttons for Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As...

Channel Number	URI	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet Loss...	Transmit Jitter	Transmit Packet Loss...
1		0	3	Outgoing Al...	00:00:18	192.168.1.1			Extn 9246, H323 9246	Outgoing					
2				Idle	2 days 22:5...										
3				Idle	2 days 22:5...										
4				Idle	2 days 22:5...										
5				Idle	2 days 22:5...										
6				Idle	2 days 22:5...										
7				Idle	2 days 22:5...										
8				Idle	2 days 22:5...										
9				Idle	2 days 22:5...										
10				Idle	2 days 22:5...										

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



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

9. Conclusion

Megapath SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the SIP connection between Avaya IP Office and the Megapath SIP Trunking service via EdgeMarc SIP ALG as shown in **Figure 1**.

10. Additional References

- [1] IP Office 8.1 Installation, Document number 15-601042 Issue 26j, 19 Sep 2012
- [2] IP Office 8.1 Manager 10.1, Document number 15-601011 Issue 29o, 03 Aug 2012
- [3] IP Office 8.1 Administering Voicemail Pro, Document number 15-601063 Issue 27b, 05 June 2012
- [4] Megapath SIP Trunking Specification – SMB Platform Interoperability for IP PBXs, Document number VSEST2090831, 16 Sep 2011

Product documentation for Avaya products may be found at: <http://support.avaya.com>. Additional IP Office documentation can be found at: <http://marketingtools.avaya.com/knowledgebase/>

Product documentation for Megapath SIP Trunking may be found at: <http://www.megapath.com/voice/sip-trunking/>

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