



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 8.1 with Etisalat SIP Trunk service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the Etisalat SIP Trunk Service and Avaya IP Office.

The Etisalat SIP Trunk Service provides PSTN access via a SIP trunk connected to the Etisalat Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or digital trunks. Etisalat are 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 the Etisalat SIP Trunk Service and Avaya IP Office. Etisalat SIP Trunk provides PSTN access via a SIP trunk connected to the Etisalat network as an alternative to legacy Analogue or Digital trunks. This approach generally results in lower cost for customers.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to Etisalat SIP Trunk. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

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

Avaya IP Office was connected to Etisalat SIP Trunk. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types including H.323, Digital and Analogue 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 including H.323, Digital, and Analogue 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 an IP Office Softphone client
- Various call types including: local, long distance, international, toll free (outbound) and directory assistance
- Codecs G.711A, G.711MU and G.729A
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- T.38 fax

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for Etisalat SIP Trunk with the following observations:

- When testing called party hang-up on incoming calls, the signalling was as expected, but the PSTN phone did not appear to clear. This was proved to be a network characteristic as PSTN to PSTN calls behaved in a similar manner.
- When testing calling party hang-up on outgoing calls, the signalling was as expected, but again the PSTN phone did not appear to clear.
- The handling of DTMF according to RFC 2833 did not work for incoming calls from all originations, though it was effectively tested with an origination provided by Etisalat during test.
- Toll Free access was not available for incoming calls and was not tested.
- Outbound international calls could not be tested as international access was not available from the Etisalat Lab environment, though outbound national calls were tested successfully.
- No test call was made to Emergency Services as no test call was booked with the Emergency Services Operator.
- Incoming calling number restriction (CLIR) could not be tested as there was no facility for restricting CLI from the network.
- Outgoing calling number restriction (CLIR) could not be tested as the test access was screened using the From header. Calls with a From header containing “anonymous” in the user portion of the URI failed.

2.3. Support

For technical support on Etisalat products please visit the website at www.etisalat.ae or contact an authorized Etisalat representative on 800-9111 For Enterprise Customers & 800-5800 For SMB customers.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to Etisalat SIP Trunk Service. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 9630 IP Telephone (with H.323 firmware), an Avaya 9621 IP Telephone (with H.323 firmware), an Avaya 1408 Digital Telephone, an Analogue Telephone, and a fax machine. The site also has a Windows XP PC running Avaya IP Office Manager to configure the Avaya IP Office as well as an IP Office Softphone client for mobility testing. For security purposes, any PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, all phone numbers have been obscured beyond the city code.

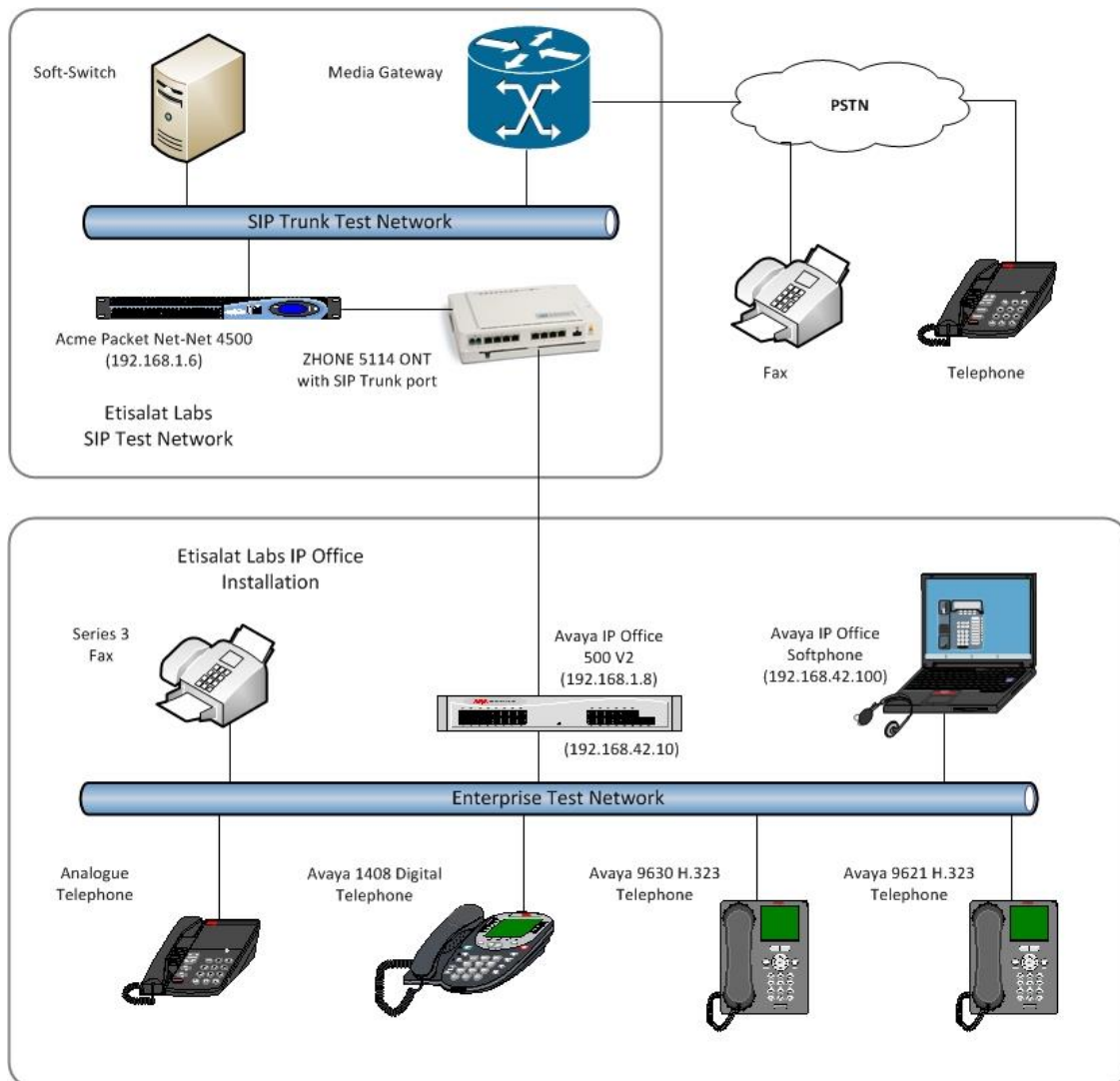


Figure 1: Etisalat SIP Trunk to Avaya IP Office Topology

4. Equipment and Software Validated

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

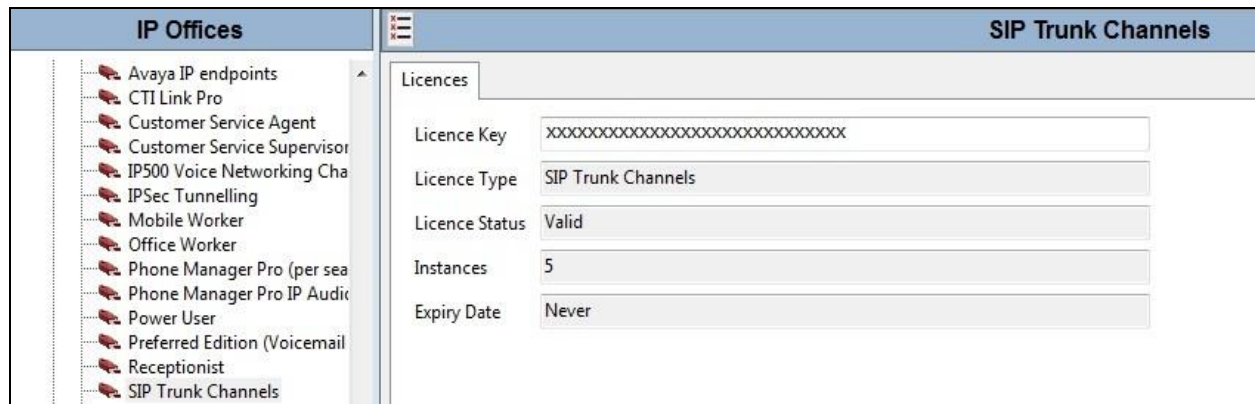
Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	Avaya IP Office R8.1(67)
Avaya 9630 Phone (H.323)	3.1
Avaya 9621 Phone (H.323)	6.2
Analogue Phone	N/A
Avaya Softphone	3.2.3.48 6700
Etisalat	
Softswitch : Huawei Softx3000	R10
SBC : AcmePacket 4500	SCX6.4.0

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to Etisalat SIP Trunk. 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) is assumed to already be in place.

5.1. Verify System Capacity

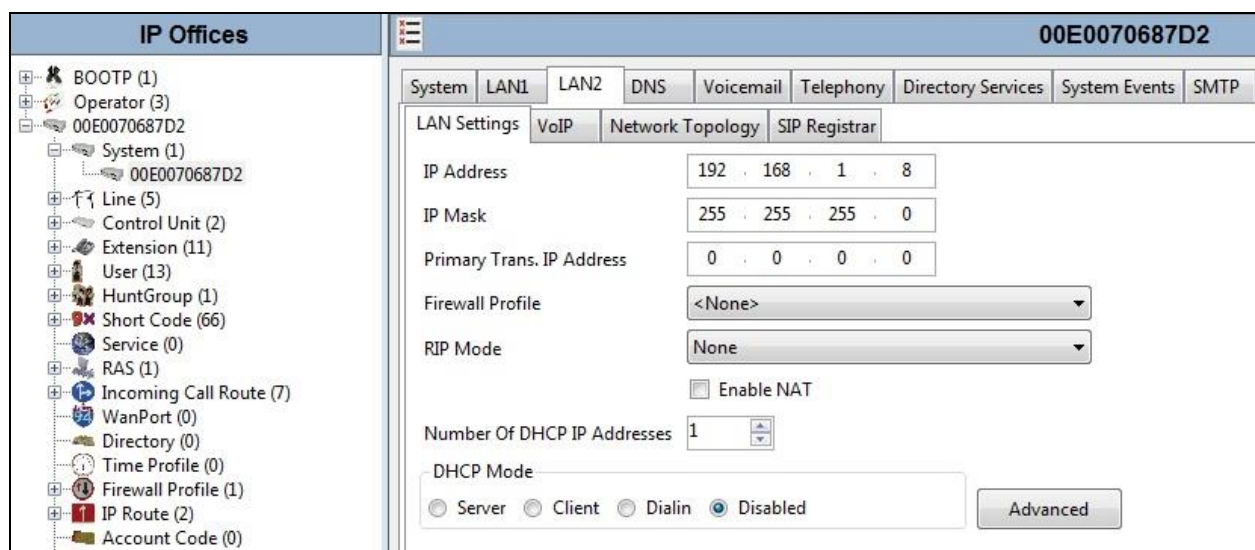
Navigate to **License → SIP Trunk Channels** in the Navigation Pane. In the Details Pane verify that the **License Status** is Valid and that the number of **Instances** is sufficient to support the number of SIP trunk channels provisioned by Etisalat.



SIP Trunk Channels	
Licences	
Licence Key	xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
Licence Type	SIP Trunk Channels
Licence Status	Valid
Instances	5
Expiry Date	Never

5.2. LAN2 Settings

In the sample configuration, the LAN2 port was used to connect the Avaya IP Office to the network SBC. To access the LAN2 settings, first navigate to **System → <IP Office Name>** in the Navigation Pane where IP Office Name is the name of the IP Office being configure, in this case **00E0070687D2**. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public interface of the IP Office. **Primary Trans. IP Address** is the next hop, usually the default gateway address, though in this case as there is a direct connection to the network SBC a default gateway address is not required. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).



00E0070687D2								
System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP
LAN Settings								
IP Address	192 . 168 . 1 . 8							
IP Mask	255 . 255 . 255 . 0							
Primary Trans. IP Address	0 . 0 . 0 . 0							
Firewall Profile	<None>							
RIP Mode	None							
<input checked="" type="checkbox"/> Enable NAT								
Number Of DHCP IP Addresses	1							
DHCP Mode								
<input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dialin <input checked="" type="radio"/> Disabled								
Advanced								

On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The IP Office Softphone uses SIP. If Softphone along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. 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.

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 signalling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signalling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

IP Offices

- BOOTP (1)
- Operator (3)
- 00E0070687D2
 - System (1)
 - Line (5)
 - Control Unit (2)
 - Extension (11)
 - User (13)
 - HuntGroup (1)
 - Short Code (66)
 - Service (0)
 - RAS (1)
 - Incoming Call Route (7)
 - WanPort (0)
 - Directory (0)
 - Time Profile (0)
 - Firewall Profile (1)
 - IP Route (2)
 - Account Code (0)
 - Licence (24)
 - Tunnel (0)
 - User Rights (8)
 - Auto Attendant (0)
 - ARS (1)

00E0070687D2

System | LAN1 | LAN2 | DNS | Voicemail | Telephony | Directory Services | System Events | SMTP | SMDR | Twinning | VCM | CCR

LAN Settings | VoIP | Network Topology | SIP Registrar

☒ 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

B8 DSCP(Hex) FC DSCP Mask (Hex) 88 SIG DSCP (Hex) 88

46 DSCP 63 DSCP Mask 34 SIG DSCP

DHCP Settings

Primary Site Specific Option Number (SSON) 176

Secondary Site Specific Option Number (SSON) 242

On the **Network Topology** tab in the Details Pane enter the information required if NAT is to be used. During test, NAT was not required and there was no requirement for a STUN server. To disable this facility, 0.0.0.0 is entered in the **STUN Server IP Address** and **Public IP Address** fields. If NAT is to be used, this tab can also be used to set the **Binding Refresh Time** for the periodic sending of OPTIONS. Alternatively, the periodic sending of OPTIONS can be specified in the User settings, see **Section 5.10** for more details. During test, OPTIONS were not sent from IP Office. On completion, click the **OK** button (not shown).

The screenshot shows the IP Office configuration interface for system 00E0070687D2. The left pane shows a tree view of the system configuration. The right pane has tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, and Twinning. The Network Topology tab is selected, showing the Network Topology Discovery section. The STUN Server IP Address is set to 0.0.0.0, the STUN Port is 3478, the Firewall/NAT Type is Open Internet, the Binding Refresh Time (seconds) is 0, the Public IP Address is 0.0.0.0, and the Public Port is 0. There are Run STUN and Cancel buttons, and a checkbox for Run STUN on startup.

5.3. System Telephony Settings

Navigate to the **Telephony** → **Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Europe, Middle East and Africa, **ALAW** is commonly 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. On completion, click the **OK** button (not shown).

The screenshot shows the IP Office configuration interface for system 00E0070687D2. The left pane shows a tree view of the system configuration. The right pane has tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, CCR, and Codecs. The Telephony tab is selected, showing the Analogue Extensions section. The Default Outside Call Sequence is Normal, the Default Inside Call Sequence is Ring Type 1, the Default Ring Back Sequence is Ring Type 2, and the Restrict Analogue Extension Ringer Voltage checkbox is unchecked. The Dial Delay Time (secs) is 4, the Dial Delay Count is 0, the Default No Answer Time (secs) is 15, the Hold Timeout (secs) is 0, the Park Timeout (secs) is 300, the Ring Delay (secs) is 5, the Call Priority Promotion Time (secs) is Disabled, the Default Currency is EUR, and the Default Name Priority is Favour Trunk. The Companding Law section shows the Switch set to A-Law and the Line set to A-Law Line. The Inhibit Off-Switch Forward/Transfer checkbox is unchecked. Other options include DSS Status, Auto Hold, Dial By Name, Show Account Code, Restrict Network Interconnect, Drop External Only Impromptu Conference, Visually Differentiate External Call, Unsupervised Analog Trunk Disconnect Handling, and High Quality Conferencing.

5.4. System Twinning Settings

Navigate to the **Twinning** tab, if the original calling party information is to be sent to the twinned phone in the SIP From header, check the box labeled **Send original calling party information for Mobile Twinning**. During test, the network was screening calls based on the numbers in the From header. To avoid calls failing screening, the box was not checked and a default number for the IP Office was entered in the **Calling party information for Mobile Twinning** field. This setting only affects twinning and does not impact the messaging of other redirected calls such as forwarded calls. If this box is checked, it will also override any setting of the **Send Caller ID** parameter on the SIP line (**Section 5.6**). On completion, click the **OK** button (not shown).

The screenshot shows the 'IP Offices' tree on the left with 'Line (5)' selected. The main pane displays the '00E0070687D2' configuration. The 'Twinning' tab is active, showing the 'Send original calling party information for Mobile Twinning' checkbox (unchecked) and the 'Calling party information for Mobile Twinning' text field containing '0251nnnn0'.

5.5. Codec Settings

Navigate to the **Codecs** tab (not shown) on the Details Pane. Check the Available Codecs boxes as required. Note that **G.711 ULAW 64K** and **G.711 ALAW 64K** are greyed out and always available. Once available codecs are selected, they can be used or unused by using the horizontal arrows as required. Note that in test, **G.711 ULAW 64K**, **G.711 ALAW 64K** and **G.729(a) 8K** were used. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).

The screenshot shows the 'IP Offices' tree on the left with 'Line (5)' selected. The main pane displays the '00E0070687D2' configuration. The 'Codecs' tab is active, showing three panels: 'Available Codecs' (with checkboxes for G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ), 'Default Codec Selection' (with 'Unused' and 'Selected' lists and navigation arrows), and 'Selected' (showing G.711 ULAW 64K, G.711 ALAW 64K, and G.729(a) 8K CS-ACELP).

5.6. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Etisalat SIP Trunk service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New→SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the parameters below to connect to the SIP Trunking service.

- Set **ITSP Domain Name** field to the domain name used by Etisalat. In test this was **51nnnn9.etisalat** (as the user portion contains part of the DDI number, some digits have been overwritten).
- Set **Send Caller ID** to that required if the box labeled **Send original calling party information for Mobile Twinning** is unchecked in **Section 5.4**. During test this was set to **None**, and the number used for Mobile Twinning was specified in the **Calling party information for Mobile Twinning** field in **Section 5.4**.
- Ensure the **In Service** box is checked.
- Default values may be used for all other parameters.

On completion, click the **OK** button (not shown).

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with various components like BOOTP, Operator, System, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, Licence, Tunnel, User Rights, Auto Attendant, and ARS. The 'Line (5)' component is selected, and 'Line 17' is highlighted. The main pane shows the 'SIP Line - Line 17' configuration page. The 'SIP Line' tab is active, showing fields for Line Number (17), ITSP Domain Name (51nnnn9.etisalat), Prefix, National Prefix (0), Country Code, International Prefix (00), In Service (checked), Use Tel URI, Check OOS, Call Routing Method (Request URI), Originator number for forwarded and twinning calls, Name Priority (System Default), Caller ID from From header, Send From In Clear, User-Agent and Server Headers, Send Caller ID (None), Association Method (By Source IP address), REFER Support (checked), Incoming (Auto), Outgoing (Auto), and UPDATE Supported (Auto).

Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address of the network SBC
- Set **Use Network Topology Info** to **None** if NAT is not to be used as was the case during test
- Set **Layer 4 Protocol** to **UDP**
- Set **Send Port** and **Listen Port** to **5060**

On completion, click the OK button (not shown).

The screenshot shows the Avaya IP Office configuration interface. On the left is a tree view of the system hierarchy under 'IP Offices', including BOOTP (1), Operator (3), System (1), Line (5), Control Unit (2), Extension (11), User (13), HuntGroup (1), Short Code (66), and Service (0). The main pane is titled 'SIP Line - Line 17' and has tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'Transport' tab is selected. It contains the following fields: ITSP Proxy Address (192.168.1.6), Network Configuration section with Layer 4 Protocol (UDP), Send Port (5060), Use Network Topology Info (None), and Listen Port (5060). Below this are Explicit DNS Server(s) (0.0.0.0), Calls Route via Registrar (checked), and a Separate Registrar field.

After the SIP line parameters are defined, the SIP URIs 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.

The screenshot shows the 'SIP URI' tab for 'SIP Line - Line 17'. It features a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The table is currently empty. To the right of the table are three buttons: 'Add...', 'Remove', and 'Edit...'.

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** to **Use Internal Data**, this setting ensures that the number set in the SIP tab of the User as shown in **Section 5.8** is sent out in the From header. During test, this was required to ensure that the calls did not fail screening.
- Set **Contact**, **Display Name** and **PAI** to **Use Internal Data**
- For **Registration**, provided that registration details have been entered in the **SIP Credentials** tab, the appropriate registration details can be selected in the drop down menu. If the entry has not yet been made, select **0: <None>**.
- 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 **17** was defined that was associated to a single line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

On completion, click the **OK** button.

SIP Line - Line 17

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

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
---------	--------	-----	-----------	---------	--------------	-----	------------	-----------

Buttons: Add..., Remove, Edit...

Edit Channel

Via: <None>

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 1: 0251nnnn9

Incoming Group: 17

Outgoing Group: 17

Max Calls per Channel: 10

Buttons: OK, Cancel

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

- Select **Custom** in the **Codec Selection** drop down menu to specify the preferred codecs
- Highlight codecs in the **Unused** box that are to be used on this line and click on the right arrows to move them to the **Selected** box
- Highlight codecs in the **Selected** box that are not to be used and click on the left arrows to move them to the **Unused** box
- Highlight codecs in the **Selected** box and use the up and down arrows to change the priority order of the offered codecs, for testing with Etisalat this was **G.711 ALAW 64K** followed by **G.729(a) 8K CS-ACELP**
- Select **T38** in the **Fax Transport Support** drop down menu to allow T.38 fax operation
- Select **RFC2833** in the **DTMF Support** drop down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833. Note that during test, out of band transmission of DTMF using the **Info** message was tested, though at the time there was no method of passing DTMF provided by the network apart from sending tones in-band.
- Uncheck the **VoIP Silence Suppression** box
- Check the **Re-invite Supported** box, to allow for codec re-negotiation in cases where the target of the incoming call or transfer does not support the codec originally negotiated on the trunk
- Check the **PRACK/100rel Supported** box to allow for reliable responses to provisional call set-up messages such as 183 Session progress and 180 Ringing.
- Default values may be used for all other parameters
- On completion, click the **OK** button (not shown).

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The 'Codec Selection' section has a 'Custom' dropdown. Below it, the 'Unused' box contains 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'. The 'Selected' box contains 'G.711 ALAW 64K' and 'G.729(a) 8K CS-ACELP'. Arrows between the boxes allow moving codecs, and up/down arrows within the 'Selected' box allow reordering. On the right, checkboxes for 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (checked) are shown. At the bottom, 'Fax Transport Support' is set to 'T38', 'Call Initiation Timeout (s)' is set to '4', and 'DTMF Support' is set to 'Info'.

Select the **T.38 Fax** tab to set the T.38 parameters for the line. Un-check the Use Default Values box (not shown) and select **0** from the **T38 Fax Version** drop down menu. Set the **Max Bit Rate (bps)** to **14400**. All other field may retain their default values. On completion, click the **OK** button (not shown).

The screenshot shows the 'SIP Line - Line 17*' configuration window with the 'T38 Fax' tab selected. The 'T38 Fax Version' dropdown is set to '0'. The 'Transport' dropdown is set to 'UDPTL'. The 'Redundancy' section has 'Low Speed' and 'High Speed' both set to '0'. The 'TCF Method' dropdown is set to 'Trans TCF'. The 'Max Bit Rate (bps)' dropdown is set to '14400'. The 'EFlag Start Timer (msecs)' is set to '2600', 'EFlag Stop Timer (msecs)' is set to '2300', and 'Tx Network Timeout (secs)' is set to '150'. On the right, several checkboxes are visible: 'Scan Line Fix-up' (checked), 'TFOP Enhancement' (checked), 'Disable T30 ECM' (unchecked), 'Disable EFlags For First DIS' (unchecked), 'Disable T30 MR Compression' (unchecked), and 'NSF Override' (unchecked). Below these are 'Country Code' and 'Vendor Code' both set to '0'.

Select the **SIP Credentials** tab to configure the authentication parameters for Etisalat SIP Trunk. The authentication takes place on the SIP registration. Define the login details in the **User name** and **Authentication Name** fields. Define the domain in the **Contact** field and the password in the **Password** field. Check the **Registration required** box

The screenshot shows the 'SIP Line - Line 17*' configuration window with the 'SIP Credentials' tab selected. At the top, there is a table with columns: Index, UserNa..., Authentication Name, Contact, Expiry (mins), and Register. Below the table are buttons for 'Add...', 'Remove', and 'Edit...'. At the bottom, the 'Edit SIP Credentials' section contains the following fields: 'User name' (0251nnnn9), 'Authentication Name' (51nnnn9.etisalat), 'Contact' (0251nnnn9), 'Password' (masked with asterisks), 'Expiry (mins)' (6), and 'Registration required' (checked). 'OK' and 'Cancel' buttons are at the bottom right.

Note: It is advisable at this stage to save the configuration as described in **Section 5.11** to make the Line Group ID defined in **Section 5.6** available.

5.7. Short Codes

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **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
- The example shows **9N;** which 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** which will insert the dialled number into the Request URI and To headers in the outgoing SIP INVITE message
- Set the **Line Group Id** to the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.6**

On completion, click the **OK** button (not shown).

The screenshot displays the Avaya configuration interface. On the left, the 'IP Offices' navigation pane lists various office configurations, including *53*N#, *55, *57*N#, *70, *9000*, *91N;, *92N;, *DSSN, *SDN, *SKN, and 9N;. The main pane shows the 'Short Code' configuration for the '9N;; Dial' short code. The configuration fields are as follows:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	17
Locale	
Force Account Code	<input type="checkbox"/>

5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required. The example below shows the changes required to use IP Office Softphone which was used in test. Softphone replaced Phone Manager at IP Office 8.0.

- Change the **Name** of the User if required, this will be used for login to the IP Office Softphone
- Select an appropriate User from the **Profile** drop down menu, in test **Power User** was used as this includes the ability to use Softphone
- Check the **Enable Softphone** box

The screenshot displays the IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'User (13)' expanded, listing various users including '3000 Sumedh'. The main area on the right shows the configuration for 'Sumedh: 3000' with tabs for User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The 'User' tab is active, showing fields for Name (Sumedh), Password (****), Confirm Password (****), Full Name, Extension (3000), Email Address, Locale, Priority (5), System Phone Rights (None), and Profile (Power User). Below these are checkboxes for Receptionist, Enable Softphone (checked), Enable one-X Portal Services, Enable one-X TeleCommuter, Enable Remote Worker, Enable Flare, and Ex Directory. The Flare Mode is set to Simultaneous.

IP Office Softphone uses SIP for signalling and hence required setting of the **SIP Registrar Enable** as described in **Section 5.2**. Call forwarding and transfer make use of the SIP REFER message. To handle SIP REFER on IP Office, the Call waiting function is used.

To turn on Call Waiting, navigate to **Telephony**→**Call Settings**. Check the **Call Waiting On** box.

The screenshot shows the 'Sumedh: 3000' web interface. The 'Telephony' tab is selected, and the 'Call Settings' sub-tab is active. The 'Call Waiting On' checkbox is checked, and the 'Answer Call Waiting On Hold' checkbox is also checked. Other settings include 'Outside Call Sequence' (Default Ring), 'Inside Call Sequence' (Default Ring), 'Ringback Sequence' (Default Ring), 'No Answer Time (secs)' (System Default (15)), 'Wrap-up Time (secs)' (2), 'Transfer Return Time (secs)' (Off), and 'Call Cost Mark-Up' (100).

Next Select the **SIP** tab (not shown) in the Details Pane. To reach the **SIP** tab click the right arrow on the right hand side of the Details Pane until it becomes visible. 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. These 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 5.6**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from Etisalat.

In the example below, one of the DDI numbers in the test range is used, though only digits beyond the area code are provided. This matches the number as received in the Request URI for incoming calls. For outgoing calls, this number is made up to the full national number in the Etisalat network. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name, though in test the DDI number was used. On completion, click the **OK** button (not shown).

The screenshot shows the 'Sumedh: 3000' web interface with the 'SIP' tab selected. The 'SIP Name', 'SIP Display Name (Alias)', and 'Contact' fields are all set to '51nnnn0'. The 'Anonymous' checkbox is unchecked.

Note: The digits that would make it possible to identify the DDI numbers have been overwritten.

5.9. Incoming Call Routing

An incoming call route maps an inbound DDI number on a specific line to an internal extension. 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 5.6**
- Set the **Incoming Number** to the incoming number that this route should match on.
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 left pane shows the 'IP Offices' hierarchy, with 'Incoming Call Route (7)' expanded, showing routes 1 through 4. The right pane contains the following fields:

Field	Value
Bearer Capacity	Any Voice
Line Group ID	17
Incoming Number	51nnnn0
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

Note: A number of digits of the DDI have been obscured. Received number format includes only the digits beyond the area code.

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 17 are routed to extension 3000.

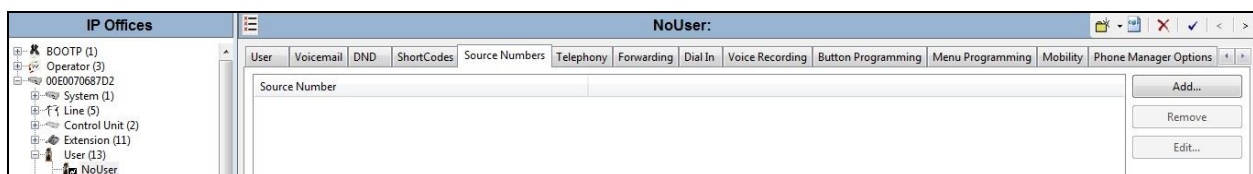
The screenshot shows the 'Incoming Call Route' configuration window with the 'Destinations' tab selected. The window displays a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	3000 Sumedh	

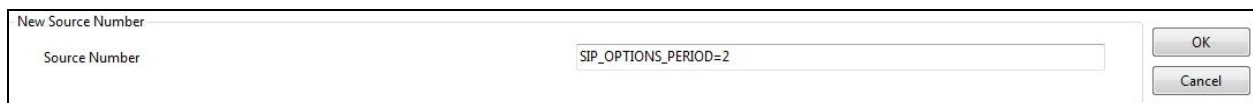
5.10. SIP Options

Avaya IP Office has the facility to send SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the lower value of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **NoUser** user. During test, OPTIONS was not sent from IP Office. If OPTIONS is required and the **Network Topology** information is not used, use **SIP_OPTIONS_PERIOD** to define the rate at which OPTIONS are sent.

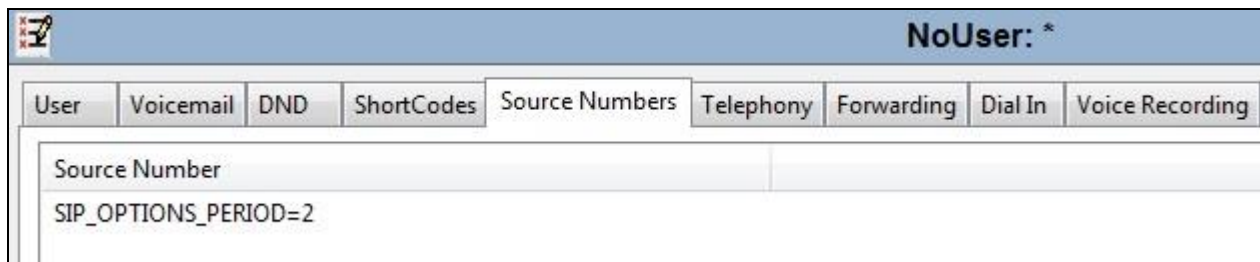
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button



At the bottom of the subsequent 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 2 minutes was used.



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. Etisalat SIP Trunk Configuration

Etisalat is responsible for the configuration of the SIP Trunk service. The customer will need to provide the public IP address used to reach the Avaya IP Office at the enterprise. Etisalat will provide the customer the necessary information to configure the SIP connection to the SIP Trunking service including:

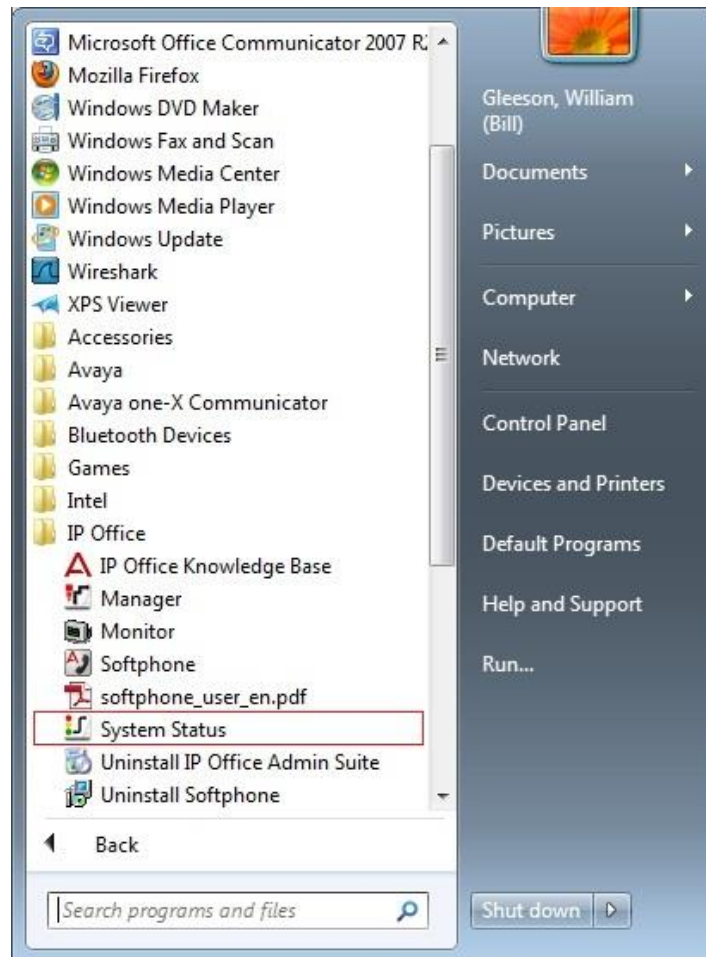
- IP address of the SIP Trunk SIP proxy
- Network SIP Domain
- Supported codecs
- DDI numbers
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

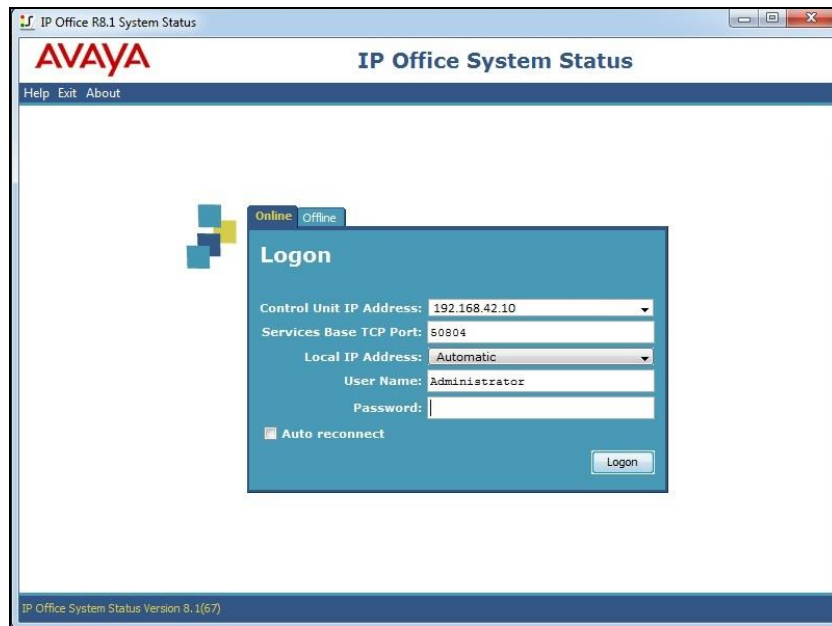
This section includes steps that can be used to verify that the configuration has been done correctly.

7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. Opening the application will vary depending on the Operating System used, the following example shows a Windows 7 Laptop PC. Click the Start button and select **All Programs**→ **IP Office** → **System Status**.



Log in to IP Office System Status at the prompt using the **Control Unit IP Address** for the IP office. The **User Name** and **Password** are the same as those used for IP Office Manager.



From the left hand menu expand **Trunks** and choose the SIP trunk (**17** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational. IP address has been changed.

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			Idle	00:12:04											
2			Idle	02:06:00											
3			Idle	02:57:00											
4			Idle	3 days 22:...											
5			Idle	3 days 22:...											
6			Idle	3 days 22:...											
7			Idle	3 days 22:...											
8			Idle	3 days 22:...											
9			Idle	3 days 22:...											
10			Idle	3 days 22:...											

8. Conclusion

The Etisalat SIP Trunk service compliance testing was completed with the only outstanding issue preventing certification being the lack of support for DTMF. A number of additional observations were made; these are listed in **Section 2.2**.

9. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>.

- [1] *IP Office 8.1 KnowledgeBase Technical Documentation CD*, 17th December 2012.
- [2] *IP Office 8.1 Installing IP500/IP500 V2*, Document number15-601042, 22nd August 2013.
- [3] *IP Office R8.1 FPI Manager 10.1*, Document number15-601011, 30th August 2013.
- [4] *IP Office 8.1 Using System Status*, Document number15-601758, 24th May 2013
- [5] *IP Office Softphone Installation*, Document number 100164693, 12th June 2012
- [6] *IP Office SIP Extension Installation*, 3rd October 2011

Appendix A – Call Flows

Registration

```
78965843ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
REGISTER sip:5106999.etisalat SIP/2.0
Via: SIP/2.0/UDP
192.168.1.8:5060;rport;branch=z9hG4bK90762813c6c2d88a7416d9797d5d48c6
From: <sip:025106999@5106999.etisalat>;tag=413c360984ff9ebf
To: <sip:025106999@5106999.etisalat>
Call-ID: 2cd1493a3bc8f98ca205ed72040ea9e3
CSeq: 115445249 REGISTER
Contact: "Unknown" <sip:025106999@192.168.1.8:5060;transport=udp>
Expires: 360
Authorization: Digest
username="5106999.etisalat", realm="Huawei", nonce="e44421bb65cf31c5605aaf0cf82ba06a", response="9c4
cd738bbe775a45bb6bfafef40d00c", uri="sip:5106999.etisalat"
Max-Forwards: 70
User-Agent: IP Office 8.1 (67)
Supported: timer,100rel
Content-Length: 0

78965880ms SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.1.8:5060;received=192.168.1.8;branch=z9hG4bK90762813c6c2d88a7416d9797d5d48c6;rport=5060
From: <sip:025106999@5106999.etisalat>;tag=413c360984ff9ebf
To: <sip:025106999@5106999.etisalat>;tag=ra7qrsuk
Call-ID: 2cd1493a3bc8f98ca205ed72040ea9e3
CSeq: 115445249 REGISTER
Expires: 360
Contact: <sip:025106999@192.168.1.8:5060;transport=udp>;expires=360
Server: Huawei SoftX3000 V300R010
Content-Length: 0
```

Incoming call

```
INVITE sip:5106904@5106999.etisalat;user=phone SIP/2.0
Via: SIP/2.0/UDP 192.168.1.6:5060;branch=z9hG4bK97g97m0008t031vvc4f1.1
Call-ID: yh766u93y36ptv9quuhu679ky6rv8vsvr@SoftX3000
From: <sip:026323466@10.129.32.24;user=phone>;tag=sahss96k-CC-22
To: <sip:5106904@5106999.etisalat;user=phone>
CSeq: 1 INVITE
Contact: <sip:026323466@192.168.1.6:5060;transport=udp>
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,PRACK,SUBSCRIBE,NOTIFY,UPDATE,MESSAGE,REFER
User-Agent: Huawei SoftX3000 V300R010
Supported: 100rel
Max-Forwards: 69
Content-Length: 230
Content-Type: application/sdp
Route: <sip:5106904@192.168.1.8:5060;user=phone;lr>

v=0
o=HuaweiSoftX3000 6747844 6747844 IN IP4 192.168.1.6
s=Sip Call
c=IN IP4 192.168.1.6
t=0 0
m=audio 37576 RTP/AVP 8 0 18
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
2300561ms CMCallEvt: 0.1056.0 -1 BaseEP: NEW CMEndpoint f5279fa0 TOTAL NOW=1 CALL_LIST=0
2300563ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
```

SIP/2.0 100 Trying
 Via: SIP/2.0/UDP 192.168.1.6:5060;branch=z9hG4bK97g97m0008t031vvc4f1.1
 From: <sip:026323466@10.129.32.24;user=phone>;tag=sahss96k-CC-22
 To: <sip:5106904@5106999.etisalat;user=phone>;tag=6delc56cd84f054e
 Call-ID: yh766u93y36ptv9quuhu679ky6rv8vsr@SoftX3000
 CSeq: 1 INVITE
 Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
 Supported: timer,100rel
 Server: IP Office 8.1 (67)
 Content-Length: 0

2300580ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
 SIP/2.0 180 Ringing
 Via: SIP/2.0/UDP 192.168.1.6:5060;branch=z9hG4bK97g97m0008t031vvc4f1.1
 From: <sip:026323466@10.129.32.24;user=phone>;tag=sahss96k-CC-22
 To: <sip:5106904@5106999.etisalat;user=phone>;tag=6delc56cd84f054e
 Call-ID: yh766u93y36ptv9quuhu679ky6rv8vsr@SoftX3000
 CSeq: 1 INVITE
 Contact: "5106904" <sip:5106904@192.168.1.8:5060;transport=udp>
 Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
 Supported: timer,100rel
 Server: IP Office 8.1 (67)
 Content-Length: 0

2305691ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
 SIP/2.0 200 OK
 Via: SIP/2.0/UDP 192.168.1.6:5060;branch=z9hG4bK97g97m0008t031vvc4f1.1
 From: <sip:026323466@10.129.32.24;user=phone>;tag=sahss96k-CC-22
 To: <sip:5106904@5106999.etisalat;user=phone>;tag=6delc56cd84f054e
 Call-ID: yh766u93y36ptv9quuhu679ky6rv8vsr@SoftX3000
 CSeq: 1 INVITE
 Contact: "5106904" <sip:5106904@192.168.1.8:5060;transport=udp>
 Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
 Supported: timer,100rel
 Server: IP Office 8.1 (67)
 Content-Type: application/sdp
 Content-Length: 169

v=0
 o=UserA 1786988355 1285439439 IN IP4 192.168.1.8
 s=Session SDP
 c=IN IP4 192.168.1.8
 t=0 0
 m=audio 49152 RTP/AVP 18
 a=rtpmap:18 G729/8000
 a=fmtp:18 annexb=no

2306149ms SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
 ACK sip:5106904@192.168.1.8:5060;transport=udp SIP/2.0
 Via: SIP/2.0/UDP 192.168.1.6:5060;branch=z9hG4bKfhhtri30986h01vjc4q0.1
 Call-ID: yh766u93y36ptv9quuhu679ky6rv8vsr@SoftX3000
 From: <sip:026323466@10.129.32.24;user=phone>;tag=sahss96k-CC-22
 To: <sip:5106904@5106999.etisalat;user=phone>;tag=6delc56cd84f054e
 CSeq: 1 ACK
 Max-Forwards: 69
 Content-Length: 0

2727849ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
 BYE sip:026323466@192.168.1.6:5060;transport=udp SIP/2.0
 Via: SIP/2.0/UDP
 192.168.1.8:5060;rport;branch=z9hG4bKe94f9dbb87e18a40996bb88a75c58c64
 From: "5106904"
 <sip:5106904@5106999.etisalat;user=phone>;tag=6delc56cd84f054e
 To: <sip:026323466@10.129.32.24;user=phone>;tag=sahss96k-CC-22
 Call-ID: yh766u93y36ptv9quuhu679ky6rv8vsr@SoftX3000
 CSeq: 3 BYE
 Contact: "5106904" <sip:5106904@192.168.1.8:5060;transport=udp>
 Max-Forwards: 70

```

Authorization: Digest
username="5106999.etisalat",realm="Huawei",nonce="66698a50a602214e193be00e79cde595",response="b99
6aca70dcd1bb71585f3944d663192",uri="sip:026323466@192.168.1.6:5060;transport=udp"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Supported: timer,100rel
User-Agent: IP Office 8.1 (67)
Content-Length: 0

2727908ms SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.1.8:5060;received=192.168.1.8;branch=z9hG4bKe94f9dbb87e18a40996bb88a75c58c64;rport=5060
From: "5106904"
<sip:5106904@5106999.etisalat;user=phone>;tag=6de1c56cd84f054e
To: <sip:026323466@10.129.32.24;user=phone>;tag=sahss96k-CC-22
Call-ID: yh766u93y36ptv9quuhu679ky6rv8vrsr@SoftX3000
CSeq: 3 BYE
Content-Length: 0

```

Outgoing call

```

4693702ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
INVITE sip:027711437@5106999.etisalat SIP/2.0
Via: SIP/2.0/UDP
192.168.1.8:5060;rport=5060;branch=z9hG4bK40b8ab0d5595a7a8f879aabfa62c8bcf
From: "5106901" <sip:5106901@5106999.etisalat>;tag=e1068905ce984530
To: <sip:027711437@5106999.etisalat>
Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1401317681 INVITE
Contact: "5106901" <sip:5106901@192.168.1.8:5060;transport=udp>
Authorization: Digest
username="5106999.etisalat",realm="Huawei",nonce="ff21825507b3fca1fb25cb3d8fb07b67",response="41e
5d66f23cfa251e81d7df643f43fdf",uri="sip:027711437@5106999.etisalat"
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Content-Type: application/sdp
Supported: timer,100rel
User-Agent: IP Office 8.1 (67)
P-Asserted-Identity: "5106901" <sip:5106901@192.168.1.8:5060>
Content-Length: 273

v=0
o=UserA 2699102053 3871155097 IN IP4 192.168.1.8
s=Session SDP
c=IN IP4 192.168.1.8
t=0 0
m=audio 49152 RTP/AVP 8 9 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

4693703ms CD: CALL: 252.1075.0 BState=Idle Cut=0 Music=0.0 Aend="Extn3020(3020)" (0.0)
Bend="Line 17" [Line 17] (0.0) CalledNum=9027711437# () CallingNum=3020 (Extn3020) Internal=0
Time=5621 AState=Dialling

4693710ms SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
192.168.1.8:5060;received=192.168.1.8;branch=z9hG4bK40b8ab0d5595a7a8f879aabfa62c8bcf;rport=5060
From: "5106901" <sip:5106901@5106999.etisalat>;tag=e1068905ce984530
To: <sip:027711437@5106999.etisalat>
Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1401317681 INVITE

4693852ms SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
SIP/2.0 180 Ringing

```

Via: SIP/2.0/UDP
192.168.1.8:5060;received=192.168.1.8;branch=z9hG4bK40b8ab0d5595a7a8f879aabfa62c8bcf;rport=5060
From: "5106901" <sip:5106901@5106999.etisalat>;tag=e1068905ce984530
To: <sip:027711437@5106999.etisalat>;tag=sqk79ru8-CC-36
Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1401317681 INVITE
Contact: <sip:192.168.1.6:5060;transport=udp>
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,PRACK,SUBSCRIBE,NOTIFY,UPDATE,MESSAGE,REFER
Require: 100rel
RSeq: 1
Content-Length: 0

4693861ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
PRACK sip:192.168.1.6:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP
192.168.1.8:5060;rport;branch=z9hG4bK61acc870cec3736a4b35c3026803728d
From: "5106901" <sip:5106901@5106999.etisalat>;tag=e1068905ce984530
To: <sip:027711437@5106999.etisalat>;tag=sqk79ru8-CC-36
Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1401317682 PRACK
Rack: 1 1401317681 INVITE
Max-Forwards: 70
Authorization: Digest
username="5106999.etisalat",realm="Huawei",nonce="ff21825507b3fca1fb25cb3d8fb07b67",response="712
ed4957885794924efec548356c310",uri="sip:192.168.1.6:5060;transport=udp"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Supported: timer,100rel
User-Agent: IP Office 8.1 (67)
Content-Length: 0

4693899ms SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.1.8:5060;received=192.168.1.8;branch=z9hG4bK61acc870cec3736a4b35c3026803728d;rport=5060
From: "5106901" <sip:5106901@5106999.etisalat>;tag=e1068905ce984530
To: <sip:027711437@5106999.etisalat>;tag=sqk79ru8-CC-36
Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1401317682 PRACK
Content-Length: 0

4697646ms SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.1.8:5060;received=192.168.1.8;branch=z9hG4bK40b8ab0d5595a7a8f879aabfa62c8bcf;rport=5060
From: "5106901" <sip:5106901@5106999.etisalat>;tag=e1068905ce984530
To: <sip:027711437@5106999.etisalat>;tag=sqk79ru8-CC-36
Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1401317681 INVITE
Allow:
INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,PRACK,SUBSCRIBE,NOTIFY,UPDATE,MESSAGE,REFER
Contact: <sip:192.168.1.6:5060;transport=udp>
Content-Length: 161
Content-Type: application/sdp

v=0
o=HuaweiSoftX3000 15794863 15794863 IN IP4 192.168.1.6
s=Sip Call
c=IN IP4 192.168.1.6
t=0 0
m=audio 35394 RTP/AVP 8
a=rtpmap:8 PCMA/8000
a=ptime:20

4697651ms SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
ACK sip:192.168.1.6:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP
192.168.1.8:5060;rport;branch=z9hG4bK09aa885c19d55e5494835632d73a791c
From: "5106901" <sip:5106901@5106999.etisalat>;tag=e1068905ce984530
To: <sip:027711437@5106999.etisalat>;tag=sqk79ru8-CC-36

```

Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1401317681 ACK
Max-Forwards: 70
Authorization: Digest
username="5106999.etisalat",realm="Huawei",nonce="ff21825507b3fca1fb25cb3d8fb07b67",response="55a
772f6537667772974312c28fe9576",uri="sip:192.168.1.6:5060;transport=udp"
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
User-Agent: IP Office 8.1 (67)
Content-Length: 0

4708642mS SIP Rx: UDP 192.168.1.6:5060 -> 192.168.1.8:5060
BYE sip:5106901@192.168.1.8:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP
192.168.1.6:5060;branch=z9hG4bK5cicrp308g6gjps9s460sd0000g00.1
Call-ID: e41efd641a5154a584b40b2d9431b0a5
From: <sip:027711437@5106999.etisalat>;tag=sqk79ru8-CC-36
To: "5106901" <sip:025106901@5106999.etisalat>;tag=e1068905ce984530
CSeq: 1 BYE
Reason: Q.850;cause=16;text="normal call clearing"
Max-Forwards: 69
Content-Length: 0

4708645mS SIP Tx: UDP 192.168.1.8:5060 -> 192.168.1.6:5060
SIP/2.0 200 OK
Via: SIP/2.0/UDP
192.168.1.6:5060;branch=z9hG4bK5cicrp308g6gjps9s460sd0000g00.1
From: <sip:027711437@5106999.etisalat>;tag=sqk79ru8-CC-36
To: "5106901" <sip:025106901@5106999.etisalat>;tag=e1068905ce984530
Call-ID: e41efd641a5154a584b40b2d9431b0a5
CSeq: 1 BYE
Supported: timer,100rel
Server: IP Office 8.1 (67)
Content-Length: 0

```

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