



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 8.1 with Completel UCM Service – Issue 1.0

Abstract

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

The Completel UCM Service provides PSTN access via a SIP trunk connected to the Completel Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or digital trunks. Completel 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 Completel UCM Service and Avaya IP Office. Completel UCM provides PSTN access via a SIP trunk connected to the Completel 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 Completel UCM. 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 Completel UCM. 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 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

Note: T.38 fax is normally tested but this is not currently functioning on IP Office due to the codec change from the network being in an UPDATE message rather than a re-INVITE. Fax will only work using G.711.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for Completel UCM with the following observations:

- When incoming test calls were not answered, it was five minutes before IP Office sent a “503 Service unavailable”. This can be reduced depending on customer requirements.
- When outgoing test calls were not answered, it was only one minute before the network sent a “BYE”. This is shorter than is usually observed.
- Test calls to busy destinations were answered and an announcement played. The more usual failure for a busy call is 486 “Busy Here” which prevents any unwanted charging of the caller
- DTMF test calls did not function correctly to all destinations. These were reliable only to test phones at the Service Provider’s site. The advice from the Service provider is that RFC 2833 is not implemented on all interconnects.
- Toll Free access was not available for incoming calls and was not tested.
- T.38 was not correctly negotiated on outbound fax test calls, these only worked when G.711 was the codec selected on call set-up. The fault is due to the incorrect handling of the SIP UPDATE sent by the network to change the codec to T.38. This requires a fix on IP Office that is not due for implementation in the short term.

2.3. Support

For technical support on Completel products please visit the website at www.Completel.net or contact an authorized Completel representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to Completel UCM Service. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with SIP firmware), an Avaya 2420 Digital Telephone, Avaya Analogue Telephone and 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 public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been changed to a private format and all phone numbers have been obscured beyond the city code.

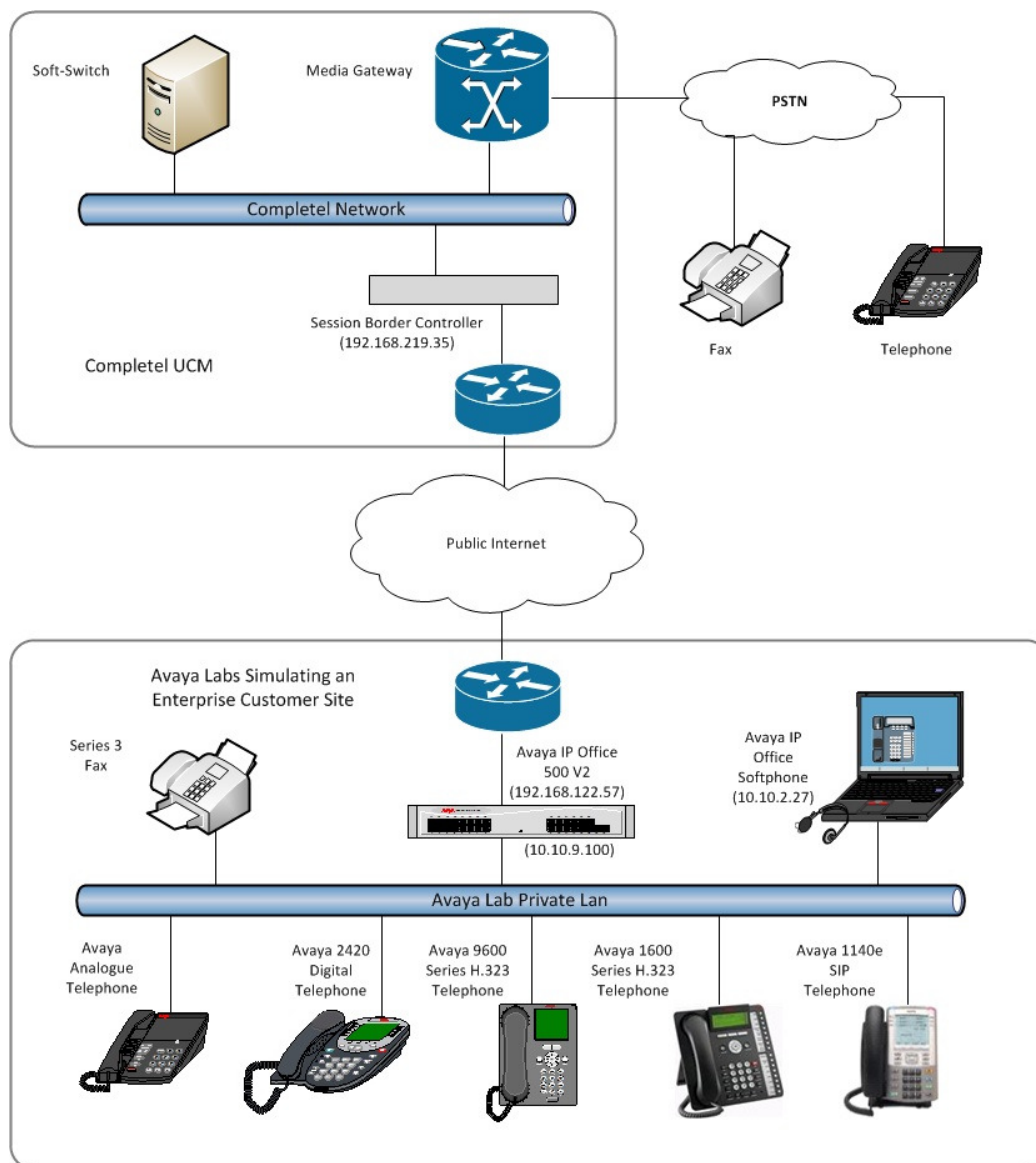


Figure 1: Completel UCM to Avaya IP Office Topology

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 1140e SIP Telephone	04.03.09.00
Avaya 1603 Phone (H.323)	1.3100
Avaya 1608 Phone (H.323)	1.3100
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analogue Phone	N/A
Avaya Softphone	3.2.3.48 6700
Completel	
ACME Net-Net 4500 SBC	SCX6.2.0 MR-5 GA (Build 777)
Cirpack Softswitch	v4.2J14

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to Completel UCM. 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** (not shown) to launch the application. Navigate to **File → Open Configuration** (not shown), 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 Completel.

SIP Trunk Channels	
Licence Key	xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
Licence Type	SIP Trunk Channels
Licence Status	Valid
Instances	255
Expiry Date	Never

5.2. LAN2 Settings

In the sample configuration, the LAN2 port was used to connect the Avaya IP Office to the external internet. To access the LAN2 settings, first navigate to **System → GSSCP_IPO2** in the Navigation Pane where GSSCP_IPO2 is the name of the IP Office. Navigate to **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. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).

GSSCP_IPO2	
System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP	
LAN Settings VoIP Network Topology SIP Registrar	
IP Address	192 . 168 . 122 . 57
IP Mask	255 . 255 . 255 . 128
Primary Trans. IP Address	192 . 168 . 122 . 7
Firewall Profile	<None>
RIP Mode	None
<input type="checkbox"/> Enable NAT	
Number Of DHCP IP Addresses	200
DHCP Mode: <input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dialin <input checked="" type="radio"/> Disabled	
<button>Advanced</button>	

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).

The screenshot displays the Avaya IP Office configuration interface for the **GSSCP_IPO2** system. The left pane shows a tree view of the system components, including **IP Offices**, **System (1)**, **GSSCP_IPO2**, **Line (9)**, **Control Unit (4)**, **Extension (32)**, **User (31)**, **HuntGroup (1)**, **Short Code (59)**, **Service (0)**, **RAS (1)**, **Incoming Call Route (5)**, **WanPort (0)**, **Directory (0)**, **Time Profile (0)**, **Firewall Profile (1)**, **IP Route (5)**, **Account Code (0)**, **Licence (76)**, **Tunnel (0)**, **User Rights (8)**, **ARS (1)**, and **RAS Location Request (0)**.

The right pane shows the **VoIP** tab selected under the **LAN Settings** section. The **VoIP** sub-tab is active, showing the following settings:

- ☒ 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**
 - DSCP(Hex): B8, DSCP Mask (Hex): FC, SIG DSCP (Hex): 88
 - DSCP: 46, DSCP Mask: 63, SIG DSCP: 34
- 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, **300** seconds or 5 minutes is shown as an example. Alternatively, the periodic sending of OPTIONS can be specified in the User settings, see **Section 5.10** for more details. On completion, click the **OK** button (not shown).

The screenshot shows the 'GSSCP_IPO2' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields:

- STUN Server IP Address: 0 . 0 . 0 . 0
- Firewall/NAT Type: Open Internet
- Binding Refresh Time (seconds): 300
- Public IP Address: 0 . 0 . 0 . 0
- STUN Port: 3478
- Public Port: 570
- Run STUN: [Button]
- Cancel: [Button]
- 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, **ALAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the Service Provider across the SIP trunk. On completion, click the **OK** button (not shown).

The screenshot shows the 'GSSCP_IPO2' configuration window with the 'Telephony' tab selected. The 'Analogue Extensions' section contains the following fields:

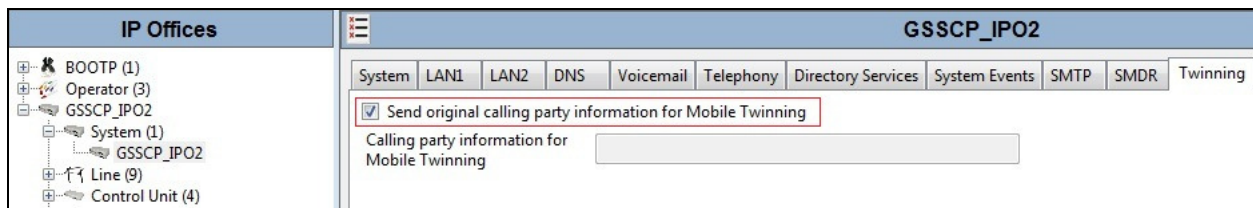
- Default Outside Call Sequence: Normal
- Default Inside Call Sequence: Ring Type 1
- Default Ring Back Sequence: Ring Type 2
- Restrict Analogue Extension Ringer Voltage: ☐
- Dial Delay Time (secs): 4
- Dial Delay Count: 0
- Default No Answer Time (secs): 15
- Hold Timeout (secs): 0
- Park Timeout (secs): 300
- Ring Delay (secs): 5
- Call Priority Promotion Time (secs): Disabled
- Default Currency: GBP
- Default Name Priority: Favour Trunk

The 'Companding Law' section contains the following fields:

- Switch: ☒ A-Law
- Line: ☒ A-Law Line
- DSS Status: ☐
- Auto Hold: ☒
- Dial By Name: ☒
- Show Account Code: ☒
- Inhibit Off-Switch Forward/Transfer: ☐
- Restrict Network Interconnect: ☐
- Drop External Only Impromptu Conference: ☐
- Visually Differentiate External Call: ☐
- Unsupervised Analog Trunk Disconnect Handling: ☐
- High Quality Conferencing: ☒

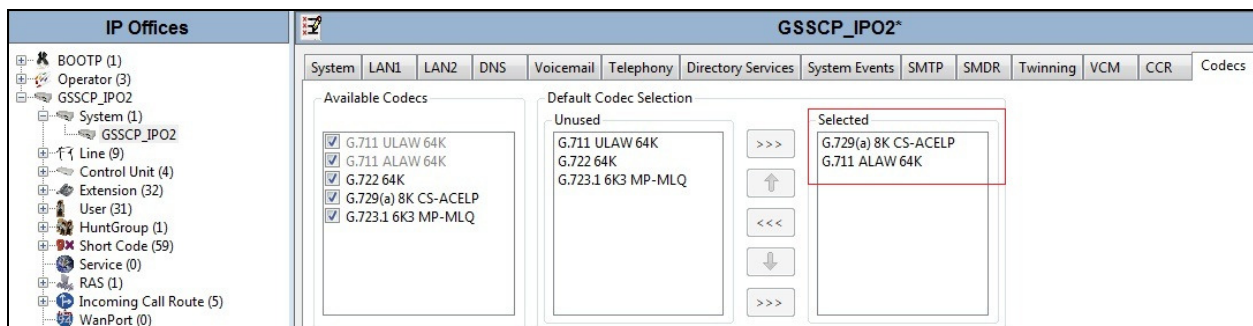
5.4. System Twinning Settings

Navigate to the **Twining** tab, check the box labeled **Send original calling party information for Mobile Twinning**. With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header (not the associated desk phone number) for calls that originate from an internal extension. For inbound PSTN calls to a twinned enabled phone, Avaya IP Office will continue to send the associated host phone number in the SIP From header (used for the caller display). 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).



5.5. Codec Settings

Navigate to the **Codecs** tab 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.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** were used. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).



5.6. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Completel UCM 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 Completel. In test this was **trusted.voip.completel.fr**.
- Set **Send Caller ID** to **None** as it is only required if the box labeled **Send original calling party information for Mobile Twinning** is unchecked 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, the 'IP Offices' navigation pane shows a tree structure with 'Line (9)' selected. The main pane is titled 'SIP Line - Line 18' and contains the following configuration fields:

- Line Number:** 18
- ITSP Domain Name:** trusted.voip.completel.fr
- In Service:** ☒
- Prefix:** (empty)
- National Prefix:** 0
- Country Code:** 47
- International Prefix:** 00
- Send Caller ID:** None
- Association Method:** By Source IP address
- REFER Support:** ☒
 - Incoming:** Never
 - Outgoing:** Never
- UPDATE Supported:** Auto
- Use Tel URI:** ☐
- Check OOS:** ☐
- Call Routing Method:** Request URI
- Originator number for forwarded and twinning calls:** (empty)
- Name Priority:** System Default
- Caller ID from From header:** ☒
- Send From In Clear:** ☐
- User-Agent and Server Headers:** (empty)

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

- Set **ITSP Proxy Address** to the IP address of the Completel SIP proxy
- 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 'SIP Line - Line 18' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '192.168.219.35'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', and 'Use Network Topology Info' is set to 'None'. 'Listen Port' is also '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty. The left pane shows a tree view of the system hierarchy.

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 Line - Line 18' configuration window with the 'SIP URI' tab selected. The table has columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. On the right, there are three buttons: 'Add...', 'Remove', and 'Edit...'. The 'Add...' button is highlighted with a red box.

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 *, this setting allows all calls on this line. If the number set in the SIP tab of the User as shown in **Section 5.8**.
- Set **Contact**, **Display Name** and **PAI** to **Use Internal Data**
- For **Registration**, only **0: <None>** is available at this point. Registration details are defined under the **SIP Credentials** tab and, once defined, are available in this drop down menu.
- 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 **18** was defined that was associated to a single line (line 18).
- 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.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	18 18	<... >	*	0: <Non...				10

Edit Channel

Via: <None>

Local URI: *

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 18

Outgoing Group: 18

Max Calls per Channel: 10

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 Completel this was **G.729(a) 8K CS-ACELP** followed by **G.711 ALAW 64K**
- 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
- 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 18*' configuration window with the 'VoIP' tab selected. The 'Codec Selection' section has a dropdown menu set to 'Custom'. Below it, there are two boxes: 'Unused' and 'Selected'. The 'Unused' box contains 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'. The 'Selected' box contains 'G.729(a) 8K CS-ACELP' and 'G.711 ALAW 64K'. Between the boxes are arrows for moving codecs: '>>>' (right), '<<<' (left), and up/down arrows. To the right of the 'Selected' box are checkboxes for 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (checked). Below the codec selection, there is a 'Fax Transport Support' dropdown set to 'T38', a 'Call Initiation Timeout (s)' field set to '4', and a 'DTMF Support' dropdown set to 'RFC2833'.

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 18*' 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, the 'Scan Line Fix-up' and 'TFOP Enhancement' checkboxes are checked. The 'Disable T30 ECM', 'Disable EFlags For First DIS', 'Disable T30 MR Compression', and 'NSF Override' checkboxes are unchecked. The 'Country Code' and 'Vendor Code' are both set to '0'.

Select the **SIP Credentials** tab to configure the authentication parameters for Completel UCM. 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 18' configuration window with the 'SIP Credentials' tab selected. A table lists the credentials:

Index	UserName	Authentication Name	Contact	Expiry (mins)	Register
1	0276510990	0276510990	trusted.voip.comptel.fr	60	True

Below the table, the 'Edit SIP Credentials' section shows the following fields:

- User name: 0276510990
- Authentication Name: 0276510990
- Contact: trusted.voip.comptel.fr
- Password: *****
- Expiry (mins): 60
- Registration required: ☒

Buttons for 'Add...', 'Remove', 'Edit...', 'OK', and 'Cancel' are visible.

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 **90N;** which will be invoked when the user dials 9 followed by a public number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **0N** which will insert the public number with leading 0 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 system components, including short codes like *92N;, *DSSN, *SDN, *SKN, 118N;, 1802, and 90N;. The main area on the right is titled '90N;; Dial' and contains the 'Short Code' configuration form. The form fields are as follows:

Short Code	
Code	90N;
Feature	Dial
Telephone Number	0N
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>

Short codes are also used for routing of national calls and Operator calls. An example for Operator calls, in this case directory enquiries, is shown below.

- The example of an Operator call shows **118** for the first three digits with no leading digit 9. This allows 118 numbers to be dialled without dialling 9 for an outside line
- Set **Telephone Number** to **118** and any subsequent dialled digits. In test, a 6 digit number was dialed for directory enquiries.
- Set other parameters as shown in the first example.

IP Offices	118N;: Dial												
<ul style="list-style-type: none">*92N;*DSSN*SDN*SKN118N;180290N;Service (0)RAS (1)Incoming Call Route (5)WanPort (0)Directory (0)Time Profile (0)	<div>Short Code</div> <table><tr><td>Code</td><td>118N;</td></tr><tr><td>Feature</td><td>Dial</td></tr><tr><td>Telephone Number</td><td>118N</td></tr><tr><td>Line Group ID</td><td>18</td></tr><tr><td>Locale</td><td></td></tr><tr><td>Force Account Code</td><td><input type="checkbox"/></td></tr></table>	Code	118N;	Feature	Dial	Telephone Number	118N	Line Group ID	18	Locale		Force Account Code	<input type="checkbox"/>
Code	118N;												
Feature	Dial												
Telephone Number	118N												
Line Group ID	18												
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 **Teleworker** User from the Profile drop down menu
- Check the **Enable Softphone** box

The screenshot shows the IP Office configuration interface. On the left is a navigation tree with 'IP Offices' expanded, showing a hierarchy: System (1) > Line (9) > Control Unit (4) > Extension (32) > User (31). Under 'User', 'NoUser' is selected. The main panel is titled 'Ext89010: 89010' and contains a 'User' tab. The 'User' tab has several fields: Name (Ext89010), Password (masked with asterisks), Confirm Password (masked with asterisks), Full Name (Ext89010), Extension (89010), Email Address (empty), Locale (dropdown), Priority (5), System Phone Rights (None), and Profile (Teleworker User). Below these fields are several checkboxes: Receptionist (unchecked), Enable Softphone (checked), Enable one-X Portal Services (checked), Enable one-X TeleCommuter (checked), Enable Remote Worker (checked), Enable Flare (unchecked), and Ex Directory (unchecked). There is also a 'Flare Mode' dropdown set to 'Standalone'.

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 'Call Settings' configuration page for extension 89010. The 'Call Waiting On' checkbox is checked and highlighted with a red box. Other settings include 'Outside Call Sequence', 'Inside Call Sequence', 'Ringback Sequence', 'No Answer Time (secs)', 'Wrap-up Time (secs)', 'Transfer Return Time (secs)', and 'Call Cost Mark-Up'.

Next Select the **SIP** tab 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 Completel.

In the example below, one of the DDI numbers in the test range is used, though only the first five and final digits are shown. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. On completion, click the **OK** button (not shown).

The screenshot shows the 'SIP' configuration page for extension 89010. The 'SIP Name', 'SIP Display Name (Alias)', and 'Contact' fields are highlighted with a red box. The 'Anonymous' checkbox is also visible.

Note: The **Contact** field must be in E.164 format for the caller ID on the called phone to display properly. Also note that the **Anonymous** box can be checked if The Calling Line Identity is to be Restricted (CLIR).

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** (not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** 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 configuration interface for an incoming call route. The left pane, titled 'IP Offices', contains a tree view with the following structure: BOOTP (1), Operator (3), GSSCP_IPO2, System (1), Line (9), Control Unit (4), Extension (32), User (31), HuntGroup (1), Short Code (59), Service (0), RAS (1), Incoming Call Route (5), and WanPort (0). Under 'Incoming Call Route (5)', there are five entries: 18 02765 nnn0, 18 02765 nnn1, 18 02765 nnn2, 18 02765 nnn3, and 18 02765 nnn5. The right pane, titled '18 02765 nnn0', has three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active, showing the following fields: Bearer Capability (Any Voice), Line Group ID (18), Incoming Number (02765 nnn0), Incoming Sub Address, Incoming CLI, Locale, Priority (1 - Low), Tag, and Hold Music Source (System Source).

Note: A number of digits of the DDI have been obscured. Number format is national.

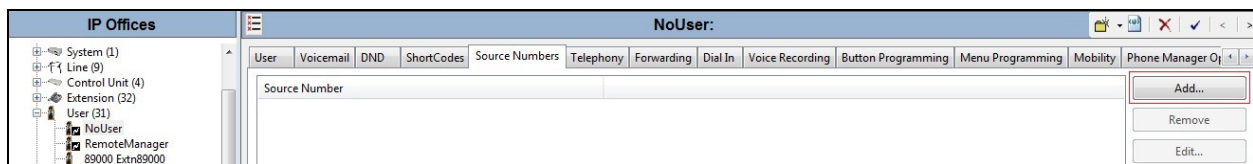
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 18 are routed to extension 89022.

The screenshot shows the 'Destinations' tab of the configuration window. It contains a table with the following columns: TimeProfile, Destination, and Fallback Extension. The table has one row with the following values: Default Value, 89022 Extn89022, and an empty field.

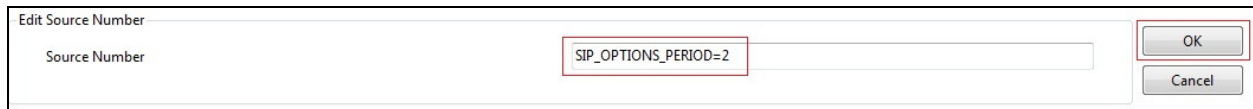
5.10. SIP Options

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 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, the **Network Topology** information was not used as there was no requirement for NAT and a STUN server. As a result, the **SIP_OPTIONS_PERIOD** was used to define the rate at which OPTIONS messages were sent.

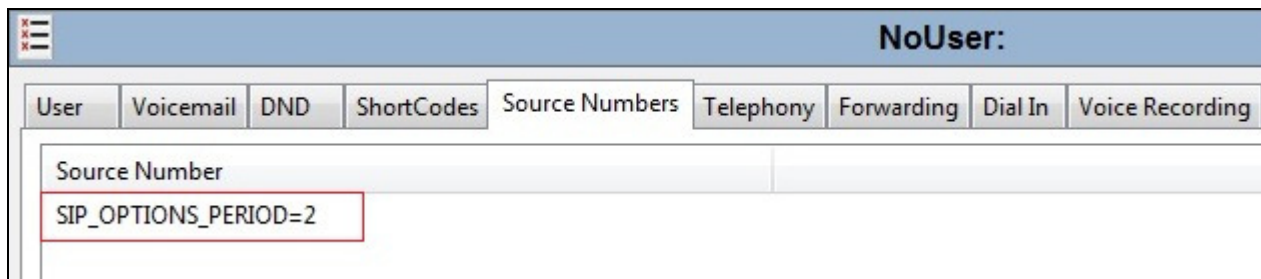
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button



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** (not shown) in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

6. Completel UCM Configuration

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

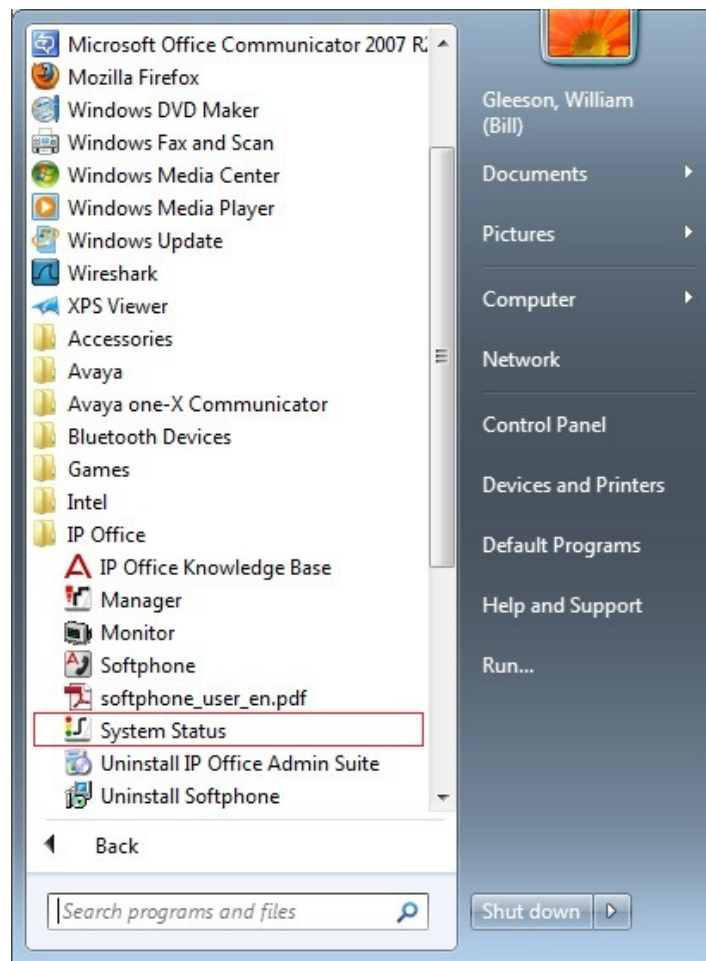
- IP address of UCM SIP proxy
- Network SIP Domain
- Supported codecs
- DDI numbers
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

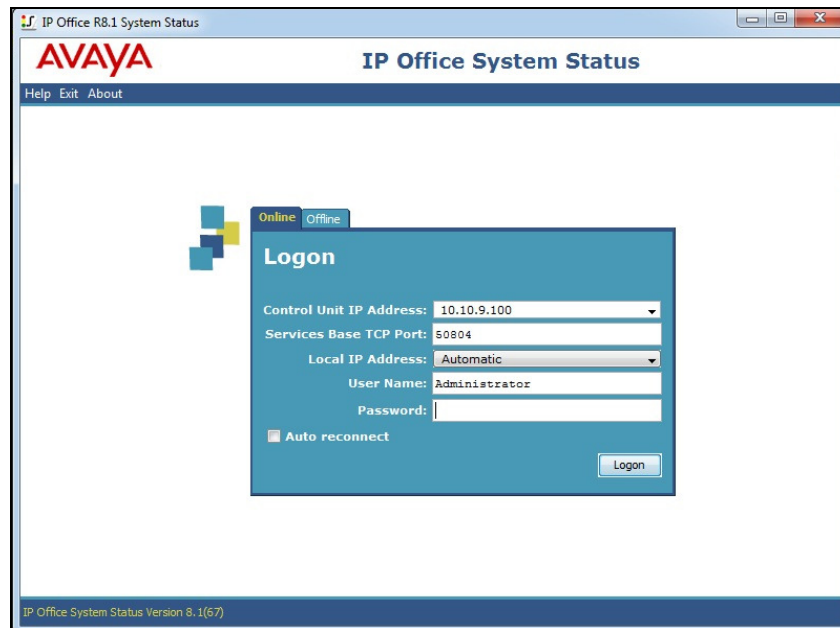
This section includes steps that can be used to verify that the configuration has been done correctly.

7.1. SIP Trunk status

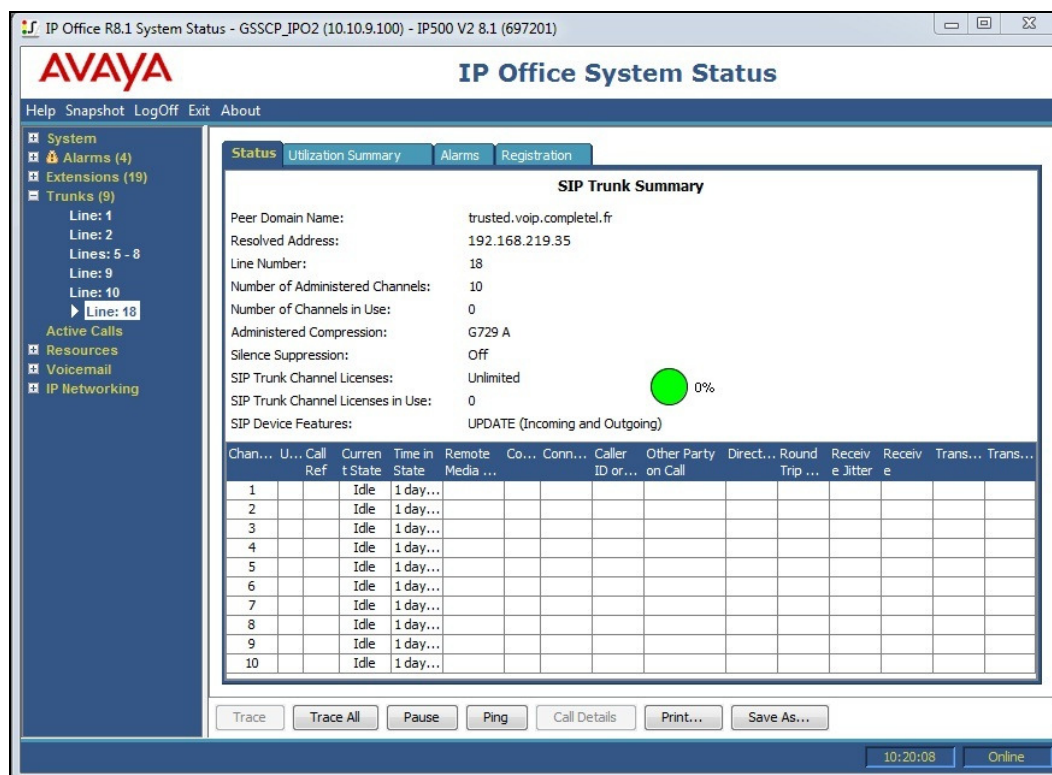
The status of the SIP trunk can be verified by opening the System Status application. A Windows 7 Laptop PC was used for testing and the application was opened by pressing the Start button and selecting **All Programs → IP Office → System Status**.



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 (**18** 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.



8. Conclusion

The Completel UCM service passed compliance testing. Interoperability testing of the sample configuration was completed with successful results for Completel UCM. Refer to **Section 2.2** for test observations.

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

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