



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for OneAccess-Telstra Business SIP with Avaya IP Office Release 11 SIP Trunking - Issue 1.0**

### **Abstract**

These Application Notes illustrate a sample configuration of OneAccess-Telstra Business SIP (Australia) with Avaya IP Office Release 11 using SIP trunks.

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

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

OneAccess-Telstra 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 OneAccess test lab.

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# 1. Introduction

These Application Notes illustrate a sample configuration for OneAccess-Telstra Business SIP (Australia) with Avaya IP Office Release 11 using SIP trunks.

The enterprise SIP trunking service available from OneAccess-Telstra Business SIP is one of many SIP-based Voice over IP (VoIP) services offered to enterprises in Australia for a variety of voice communications needs. The OneAccess-Telstra Business SIP allows enterprises in Australia to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

## 2. General Test Approach and Test Results

The general test approach was to make calls from/to the Avaya IP Office through the OneAccess NTU using OneAccess-Telstra Business SIP. The configuration (shown in **Figure 1**) was used to exercise the features and functionality tests 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and OneAccess-Telstra did not include use of any specific encryption features as requested by OneAccess-Telstra.

### 2.1 Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and outbound call flows between Avaya IP Office and the OneAccess-Telstra Business SIP.

The testing covered functionality required for compliance as a solution supported on the OneAccess-Telstra Business SIP. Calls were made to and from the PSTN across OneAccess-Telstra Business SIP. The following standard features were tested as part of this effort:

- Inbound PSTN calls to various phone types including H.323, SIP, digital and analog telephone at the enterprise. All inbound calls from PSTN are routed to the enterprise across the SIP trunk from the service provider.
- Outbound PSTN calls from various phone types including H.323, SIP, digital and analog telephone at the enterprise. All outbound calls to PSTN are routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya Communicator for Windows.
- Dialing plans including local, long distance, international, outbound toll-free, emergency calls.
- Calling Party Name presentation and Calling Party Name restriction.
- Codecs G.729A, G.711A and G.711MU.
- Fax using pass-through mode.
- DTMF tone transmissions as out-of-band RTP events as per RFC2833.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, forward and conference.
- Off-net call forward with Diversion method.
- Mobile twinning.
- Response to OPTIONS heartbeat and Registration.
- Response to incomplete call attempts and trunk errors.

## 2.2 Test Results

Interoperability testing of OneAccess-Telstra Business SIP was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **CLI restriction** - CLI restriction is not supported on outbound calls from OneAccess-Telstra Business SIP using Broadsoft platform. It is possible to use the carrier short code of \*67, in conjunction with the Avaya IP Office ARS form to achieve the feature. Please see **Section 5.8 Configure caller identity restriction on outbound call** for the detailed configuration.
- **Faxing** – OneAccess-Telstra Business SIP service only supports FAX G.711 pass-through mode. G.711 fax pass-through was successfully tested during the compliance test.
- **Direct Media** – Direct Media must be turned off for SIP line on IP Office to Telstra otherwise one way speech path may occur when changing media path mid call.
- **Blind transfer** – Telstra IP Telephone (TIPT) blind transfer Avaya IP Office phone to another Avaya IP Office phone results in no voice and call is disconnected. Carrier side disconnects the call with error "Switching Equipment Congestion". This issue needs to be investigated from carrier side.

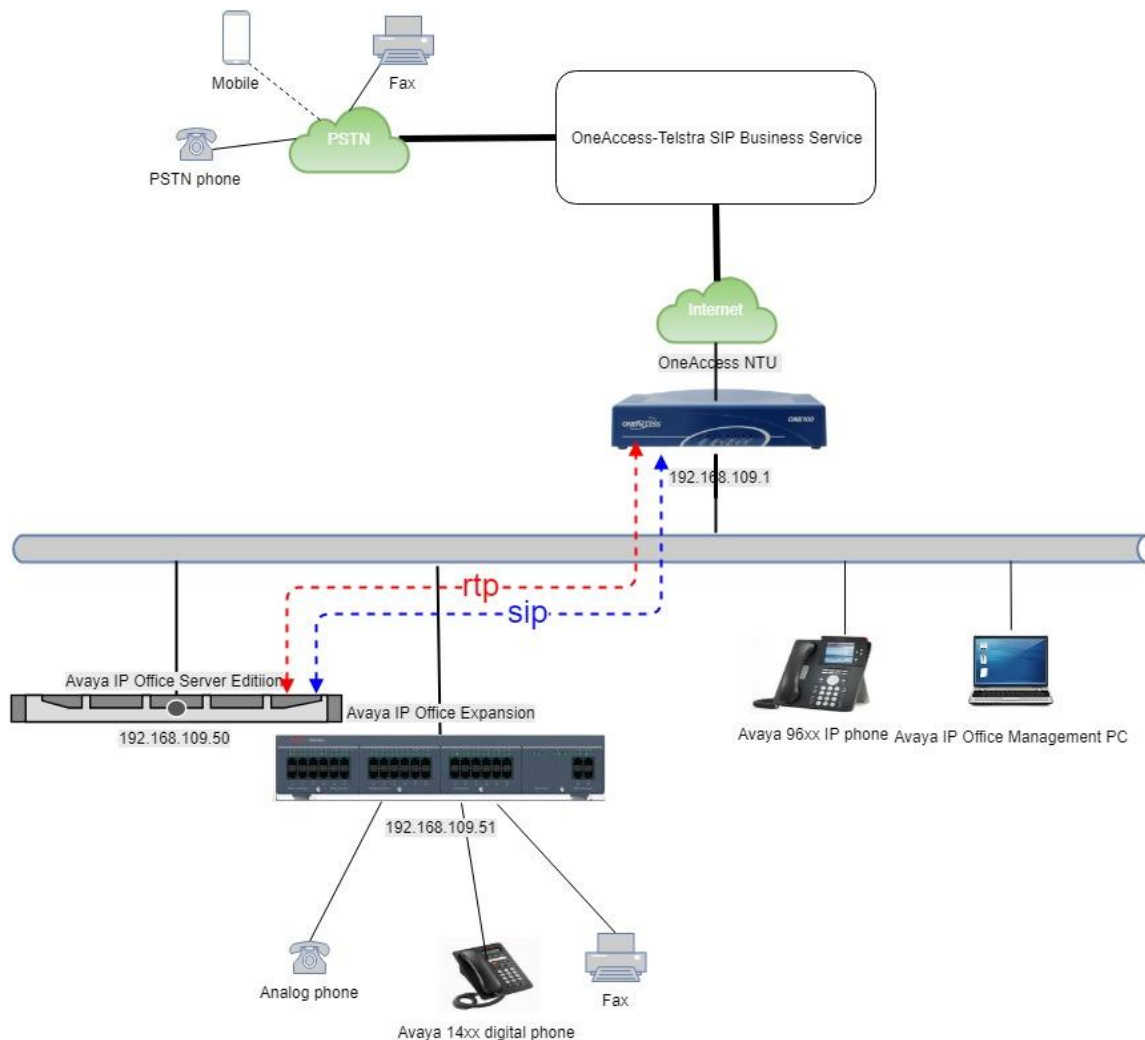
## 2.3 Support

For technical support for OneAccess-Telstra SIP Business service, contact Telstra Support at <https://www.telstra.com.au/support> or call 1800-199-458.

### 3. Reference Configuration

The reference configuration used in these Application Notes is shown in the diagram below and consists of several components.

- Avaya IP Office Server Edition running on VMware ESXi
- Avaya IP Office 500 V2.
- Avaya IP phones are represented with Avaya 9600 Series IP Telephones running H.323 software, Avaya 1600 Series IP Telephones running H.323 firmware, and Avaya 1100 Series IP Telephones running SIP firmware.
- Avaya Communicator for Windows
- Avaya 2400 Series Digital Telephones.
- OneAccess-Telstra Business SIP provided one trunk group and DID range for this testing is 0285xxx4xx (10 digits). Enterprise network is connected to Telstra network via OneAccess SIP Network Termination Unit (NTU).



**Figure 1: Network Components as Tested**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
<b>Avaya</b>	
Avaya IP Office Server Edition	11.0.0.0.0 build 849
Avaya IP Office 500 V2 Expansion System	11.0.0.0.0 build 849
Avaya 9600 Series IP Deskphones – H.323	6.6.5
Avaya 2400 Series Digital Deskphones	R6
Avaya 1600 Series IP Deskphones, H.323	1.3.11
Avaya 1100 Series IP Deskphones, SIP	4.4.8
Avaya Communicator for Windows	2.1.4.0
Analog Telephones	N/A
Fax Machine	N/A
<b>Service Provider</b>	
OneAccess-Telstra Business SIP	N/A

## 5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to OneAccess SIP NTU. 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 (not shown).

### 5.1 LAN1 Settings

In the sample configuration, the LAN1 port was used to connect to OneAccess SIP NTU. To access the LAN1 settings, first navigate to **System (1) > 000C292B2458** in the **Navigation** and **Group** panes and then navigate to the **LAN1 > LAN Settings** tab in the **Details** pane. Set the **DHCP Mode** to **Server**, then 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 network. Other parameters are set as default values.

The screenshot shows the Avaya IP Office Manager application with the 'LAN Settings' tab selected. The 'IP Address' field is set to '192 . 168 . 109 . 50' and the 'IP Mask' field is set to '255 . 255 . 255 . 0'. A red box highlights these two fields, with a blue callout box pointing to them that says 'DEFINE CALL SERVER IP ADDRESS'. Below these fields, the 'Number Of DHCP IP Addresses' is set to '154'. The 'DHCP Mode' section has three radio buttons: 'Server' (selected), 'Client', and 'Disabled'. A red box highlights the 'Server' radio button, with a blue callout box pointing to it that says 'DHCP MODE SERVER OR DISABLED, DEPENDING IF DHCP SERVER IS REQUIRED FOR SIP/IP ENDPOINTS'. An 'Advanced' button is visible to the right of the DHCP Mode section.

In 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 OneAccess SIP NTU. The **SIP Registrar Enable** box is checked to allow Avaya IP Office SIP phones usage. The **SIP Domain Name** is set to desired IP Office SIP domain or IP address. The **Layer 4 Protocol** use **UDP/TCP** with port **5060**. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. The **Enable RTCP Monitoring on Port 5005** is checked. 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 displays the VoIP configuration tab with the following settings:

- H323 Gatekeeper Enable**: ☒
  - Auto-create Extn**: ☐
    - Auto-create User**: ☐
      - H323 Remote Extn Enable**: ☐
  - H.323 Signalling over TLS**: Disabled
    - Remote Call Signalling Port**: 1720
- SIP Trunks Enable**: ☒ (Highlighted with a red box and a blue callout that says "CHECK SIP TRUNKS ENABLE")
- SIP Registrar Enable**: ☒
  - Auto-create Extn/User**: ☐
    - SIP Remote Extn Enable**: ☐
  - SIP Domain Name**: 192.168.109.50
  - SIP Registrar FQDN**: 192.168.109.50
- Layer 4 Protocol**:
  - ☒ **UDP**: **UDP Port** 5060, **Remote UDP Port** 5060
  - ☒ **TCP**: **TCP Port** 5060, **Remote TCP Port** 5060
  - ☐ **TLS**: **TLS Port** 5061, **Remote TLS Port** 5061
- Challenge Expiry Time (secs)**: 10
- RTP**:
  - Port Number Range**:
    - Minimum**: 40750, **Maximum**: 50750
  - Port Number Range (NAT)**:
    - Minimum**: 40750, **Maximum**: 50750
  - Enable RTCP Monitoring on Port 5005**: ☒
    - RTCP collector IP address for phones**: 0 . 0 . 0 . 0 . 0
  - Keepalives**:
    - Scope**: Disabled, **Periodic timeout**: 0
    - Initial keepalives**: Disabled



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. The parameter was set to **Open Internet**. All other parameters should be set according to customer requirements.

The screenshot shows a configuration window with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, and Dial. The 'Network Topology' tab is selected. Under 'Network Topology Discovery', the 'Firewall/NAT Type' is set to 'Open Internet', which is highlighted with a red box. A blue arrow points from a callout box labeled 'SELECT OPEN INTERNET' to this dropdown. Other visible fields include 'STUN Server Address' (0.0.0.0), 'STUN Port' (3478), 'Binding Refresh Time (seconds)' (0), 'Public IP Address' (0.0.0.0), and 'Public Port' (UDP, TCP, TLS all set to 0). 'Run STUN' and 'Cancel' buttons are also present.

## 5.2 System Telephony Settings

Navigate to **System (1) > 000C292B2458** in the **Navigation** and **Group** panes and then navigate to the **Telephony > Telephony** tab in the **Details** pane. Choose the **Companding Law** typical for the enterprise location. For Australia, **A-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 **Dial Delay Count** to **15** so IP Office will allow up to 15 digit dialing. Set **Dial Delay Time (sec)** to desired number.

The screenshot shows the 'Telephony' configuration page for a system. The 'Telephony' tab is selected. The 'Companding Law' section is highlighted with a red box, showing 'Switch' and 'Line' settings, both set to 'A-Law'. The 'Dial Delay Time (sec)' is set to 4 and 'Dial Delay Count' is set to 15, both highlighted with red boxes. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked and highlighted with a red box. Other settings include 'Default Outside Call Sequence' (Normal), 'Default Inside Call Sequence' (Ring Type 1), 'Default Ring Back Sequence' (Ring Type 2), 'Restrict Analogue Extension Ringer Voltage' (unchecked), 'DSS Status' (unchecked), 'Auto Hold' (checked), 'Dial By Name' (checked), 'Show Account Code' (checked), 'Restrict Network Interconnect' (unchecked), and 'Include location specific information' (unchecked).

## 5.3 System Codec Settings

Navigate to **System (1) > 000C292B2458** in the **Navigation** and **Group** panes and then navigate to the **Codecs** tab in the **Details** pane. Choose the **RFC2833 Default Payload** as IP Office default of **101**. Select codecs **G.729(a) 8K CS-ACELP**, **G.711 ALAW 64K** and **G.711 ULAW 64K**.

The screenshot shows the 'Codecs' configuration page for a system. The 'VoIP' tab is selected. The 'RFC2833 Default Payload' is set to 101. The 'Available Codecs' list includes G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, and G.729(a) 8K CS-ACELP. The 'Default Codec Selection' section shows 'Unused' and 'Selected' lists. The 'Selected' list is highlighted with a red box and contains G.729(a) 8K CS-ACELP, G.711 ALAW 64K, and G.711 ULAW 64K. A blue box labeled 'DEFINE CODECS TO BE USED' points to the 'Selected' list.

## 5.4 Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and OneAccess-Telstra Business SIP. 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** (not shown) and enter the desired number for **Line number** (here **2** was chosen). On the **SIP Line** tab in the **Details** pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the enterprise domain (or IP address) so that IP Office uses this domain as the host portion of the SIP URI in SIP headers such as the From header.
- Set **Local Domain Name** to the same domain (or IP address) set in **LAN1**.
- Check the **In Service** box.
- Set **URI Type** to SIP.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Under **Session Timers**:
  - **Refresh Method**: Select **Update**.
  - **Timer (sec)**: Enter **90**.
- Set **Country Code** to **61** (Country Code of Australia).
- Set **National Prefix** to **0**.
- Set **Incoming Supervised REFER** to **Never**.
- Set **Outgoing Supervised REFER** to **Never**.

The screenshot displays the 'SIP Line' configuration window with the following fields and values:

- Line Number:** 2
- ITSP Domain Name:** 192.168.109.1
- Local Domain Name:** 192.168.109.50
- URI Type:** SIP URI
- Location:** Cloud
- Prefix:** 0
- National Prefix:** (empty)
- International Prefix:** (empty)
- Country Code:** 61
- Name Priority:** System Default
- Description:** (empty)
- In Service:** ☒ (CHECK IN SERVICE)
- Check OOS:** ☒
- Session Timers:**
  - Refresh Method:** Update
  - Timer (seconds):** 90
- Redirect and Transfer:**
  - Incoming Supervised REFER:** Never
  - Outgoing Supervised REFER:** Never
  - Send 302 Moved Temporarily:** ☐
  - Outgoing Blind REFER:** ☐

Annotations in the image include red boxes around the ITSP Domain Name, Local Domain Name, and Prefix fields, and blue arrows pointing to the 'CHECK IN SERVICE' checkbox, the 'DEFINE ITSP DOMAIN NAME AND LOCAL DOMAIN NAME' label, and the 'DEFINE SYSTEM PREFIXES' label.

Select the **Transport** tab:

- The **ITSP Proxy Address** is set to the IP address of OneAccess SIP NTU. As shown in screenshot below, this IP address is 192.168.109.1.
- In the **Network Configuration** area, **UDP** is selected as the Layer 4 Protocol, and the **Send Port** is set to **5062**, Listen Port is set to **5060**. The **Use Network Topology Info** parameter is set to **None**. Other parameters retain default values in the screen below.
- **Define Explicit DNS Server** as IP address of OneAccess-Telstra's router. As shown in the screenshot below, this is 92.168.109.1

SIP Line **Transport** Call Details VoIP SIP Credentials SIP Advanced Engineering

ITSP Proxy Address  ITSP PROXY IP ADDRESS

Network Configuration

Layer 4 Protocol  Send Port  SET SEND PORT 5062

Use Network Topology Info  Listen Port

Explicit DNS Server(s)  0 . 0 . 0 . 0 DEFINE EXPLICIT DNS OF OneAccess-Telstra's ROUTER

Calls Route via Registrar ☒

Separate Registrar

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user.

From **SIP Line** configuration, select the **Call Details** tab then click the **Add** button and the **New URI** window will appear,

- Define the **Incoming Group ID** and **Outgoing Group ID**
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern, as per your license entitlement.
- Set **Local URI**, **Contact**, and **Display Name** to **Use Internal Data**.
- Define the credentials used with SIP URI entry
- Define the rest settings as shown in the screen to present pilot number on outbound calls.

**DEFINE LINE GROUP ID USED FOR INCOMING AND OUTGOING CALLS**

**DEFINE MAX SIP TRUNK SESSIONS AS PER CURRENT SIP TRUNK CHANNELS LICENSE**

**DEFINE CREDENTIALS USED**

**DEFINE THESE SETTINGS TO ENABLE USER>SIP TAB AND GROUP>SIP TAB AND TO PRESENT EXTENSION LEVEL CALLER ID ON OUTBOUND CALLS.**

	Display	Content	Field meaning		
			Outgoing Calls	Forwarding/Twinning	Incoming Calls
Local URI	Use Internal Data	Use Internal Data	Caller	Explicit	Called
Contact	Use Internal Data	Use Internal Data	Caller	Explicit	Called
P Asserted ID	<input checked="" type="checkbox"/> Use Internal Data	Use Internal Data	Caller	Caller	Called
P Preferred ID	<input type="checkbox"/> None	None	None	None	None
Diversion Header	<input checked="" type="checkbox"/> Auto	Auto	None	Caller	None
Remote Party ID	<input type="checkbox"/> None	None	None	None	None

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.729(a) 8K CS-ACELP** , **G.711 ALAW 64K** and **G.711 ULAW 64K** codecs causes Avaya IP Office to include these codecs.
- Check the **Re-invite Supported** box.
- Uncheck **Codec Lockdown** box.
- Uncheck **Allow Direct Media Path** box.
- Set **Fax Transport Support** to **G.711** from the pull-down menu.
- Set the **DTMF Support** to **RFC2833/RFC4733** from the pull-down menu.
- Default values may be used for all other parameters.

The screenshot displays the Avaya IP Office configuration interface for the VoIP tab. The interface includes a navigation bar at the top with tabs for SIP Line, Transport, Call Details, VoIP, SIP Credentials, SIP Advanced, and Engineering. The VoIP tab is selected. The main configuration area is divided into several sections. The 'Codec Selection' section features a dropdown menu currently set to 'System Default'. Below this, there are two columns of codec lists: 'Unused' and 'Selected'. The 'Selected' list contains 'G.729(a) 8K CS-ACELP', 'G.711 ALAW 64K', and 'G.711 ULAW 64K'. The 'Fax Transport Support' section has a dropdown menu set to 'G.711'. The 'DTMF Support' section has a dropdown menu set to 'RFC2833/RFC4733'. The 'Media Security' section has a dropdown menu set to 'Disabled'. On the right side, there is a 'CHECK REINVITE SUPPORTED' section with several checkboxes: 'Local Hold Music' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), and 'PRACK/100rel Supported' (unchecked). Blue callout boxes with arrows provide instructions: 'DEFINE CODECS USED' points to the codec selection area, 'CHECK REINVITE SUPPORTED' points to the Re-invite Supported checkbox, 'UNCHECK DIRECT MEDIA PATH' points to the Allow Direct Media Path checkbox, 'SET FAX TRANSPORT SUPPORT TO G.711' points to the Fax Transport Support dropdown, and 'SET DTMF TO RFC2833/RFC4733' points to the DTMF Support dropdown. Red boxes highlight the 'System Default' dropdown, the 'Re-invite Supported' checkbox, the 'G.711' selection, and the 'RFC2833/RFC4733' selection.

Select **SIP Credentials** tab, configure the credentials as provided by service provider.

- Enter the **User Name**, **Authentication Name** and **Contact** numbers and the associated password.
- Set **Expiry** time to **10** minutes
- Check the **Registration required**

The screenshot shows the 'Edit SIP Credentials' form. A red box highlights the 'User name', 'Authentication Name', and 'Contact' fields, which all contain the value '285 4'. A blue callout box points to these fields with the text 'PILOT NUMBER OF SERVICE IN ALL FIELDS. REMOVE LEADING ZERO FROM FNN.'. Another red box highlights the 'Expiry (mins)' field, which is set to '10'. A blue callout box points to this field with the text 'SET TO 10 MINUTES'. A third red box highlights the 'Registration required' checkbox, which is checked. A blue callout box points to this checkbox with the text 'CHECK REGISTRATION REQUIRED'.

Select **SIP Advanced** tab:

- Check **Indicate HOLD** box.

The screenshot shows the 'SIP Advanced' configuration tab. In the 'Media' section, the 'Indicate HOLD' checkbox is checked. A blue callout box points to this checkbox with the text 'CHECK INDICATE HOLD'. The 'Call Control' section contains various settings for call timeouts and responses.

## 5.5 ARS table

In the left **Navigation**, right-click on **ARS** and select **New** to create a new ARS form and place in service.

The screenshot displays the ARS configuration interface. At the top, the 'ARS' tab is selected. The form includes the following fields:

- ARS Route Id: 51
- Route Name: Showpilot
- Dial Delay Time: System Default (5)
- Description: (empty)
- Secondary Dial tone: ☐ (unchecked)
- SystemTone: (dropdown menu)
- Check User Call Barring: ☒ (checked)

The 'In Service' checkbox is checked, highlighted by a blue callout box labeled 'CHECK IN SERVICE'. Below this, the 'Time Profile' is set to '<None>'. The 'Out of Service Route' and 'Out of Hours Route' are both set to '<None>'. A table below these fields contains the following data:

Code	Telephone Number	Feature	Line Group ID
N;	N"@192.168.109.1"	Dial	20

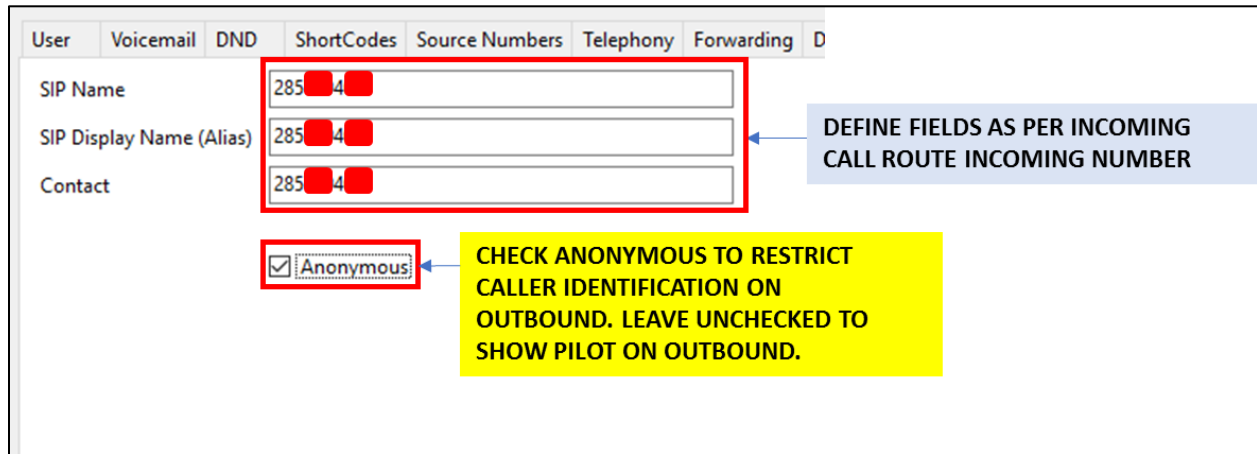
A blue callout box points to the 'Line Group ID' column, stating: 'DEFINE SHORT CODE TO DIAL OUT WITH CORRECT LINE GROUP ID AS CONFIGURED IN SIP URI TAB.' To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table, the 'Alternate Route Priority Level' is set to 3, and the 'Alternate Route Wait Time' is set to 30. The 'Alternate Route' dropdown is set to '<None>'. Arrows indicate the flow of configuration from the 'In Service' status to the 'Time Profile' and then to the 'Alternate Route' settings.



## 5.6 User

Any user that is used to make outbound calls to OneAccess-Telstra Business SIP must be configured with one of the DID numbers assigned. From the **User** in the left **Navigation**, select a user in the user list and navigate to SIP tab of that user

- Enter one of the DID numbers to **SIP Name**, **SIP Display Name (Alias)** and **Contact**.
- Check **Anonymous** to restrict caller identification on outbound call



User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	D
SIP Name							
285 [red square] 4 [red square]							
SIP Display Name (Alias)							
285 [red square] 4 [red square]							
Contact							
285 [red square] 4 [red square]							
<input checked="" type="checkbox"/> Anonymous							

DEFINE FIELDS AS PER INCOMING CALL ROUTE INCOMING NUMBER

CHECK ANONYMOUS TO RESTRICT CALLER IDENTIFICATION ON OUTBOUND. LEAVE UNCHECKED TO SHOW PILOT ON OUTBOUND.

## 5.7 Incoming Call Route

An incoming call route maps an inbound DID 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 5.4**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left. In this sample configuration, assigned DID numbers starting with 028 have been masked as 285xxx4xx due to security reasons.

Standard		Voice Recording	Destinations
Bearer Capacity	Any Voice		
Line Group ID	19		
Incoming Number	285xxx4xx		
Incoming Sub Address			
Incoming CLI			
Locale			
Priority	1 - Low		
Tag			
Hold Music Source	System Source		
Ring Tone Override	None		

Annotations:

- SET ANY VOICE FOR BEARER CAPABILITY
- DEFINE INCOMING GROUP ID AS CONFIGURED IN SIP URI TAB
- DEFINE INCOMING NUMBER. TESTING REQUIRED DROPPING THE LEADING ZERO FROM THE FNN.

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button.

Standard		Voice Recording	Destinations
TimeProfile	Destination	Fallback Extension	
Default Value	404		

Annotation:

- SELECT THE DESTINATION EXTENSION FROM THE PULL-DOWN MENU

Repeat above steps to map multiple DID numbers to multiple users / extensions. As shown in below screenshot, multiple DID numbers are mapped to multiple users / extensions, and they are sharing same line group ID

	Line Group ID	Incoming Number	Destination
<b>DEFINE INCOMING GROUP ID AS CONFIGURED IN SIP URI TAB</b>	19	285 4	404
	19	285 4	405
	19	285 4	780 9608G
	19	285 4	770 9608
	19	285 4	760 9611
	19	285 4	200 Main
		<b>DEFINE INCOMING NUMBER. TESTING REQUIRED DROPPING THE LEADING ZERO FROM THE FNN.</b>	

## 5.8 Configure Caller Identity Restriction on Outbound Call

Select **Line** in the **Navigation** pane. From the **Line**, select an existing SIP line and navigate to **Call Details** tab on the right pane. Click **Add** to create a new **SIP URI**.

- Define the **Line Group ID** that are used for incoming call and outgoing calls.
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern, as per your license entitlement.
- Define the **Credentials** that is used for the line.
- Set **Local URI**, **Contact**, and **P Asserted ID** to a pilot number; and set **Diversion Header** to **Auto**.

SIP Line - 3 | Call Details | SIP URI

New URI

Incoming Group: 20 Max Sessions: 256

Outgoing Group: 20

Credentials: 1: 285 4

DEFINE LINE GROUP ID USED FOR INCOMING AND OUTGOING CALLS

DEFINE MAX SIP TRUNK SESSIONS AS PER CURRENT SIP TRUNK CHANNELS LICENSE

DEFINE CREDENTIALS USED

	Display	Content	Field meaning		
			Outgoing Calls	Forwarding/Twinning	Incoming Calls
Local URI	285 4	285 4	Caller	Explicit	Explicit
Contact	285 4	285 4	Caller	Explicit	Explicit
P Asserted ID	<input checked="" type="checkbox"/> 285 4	285 4	Caller	Caller	Called
P Preferred ID	<input type="checkbox"/> None	None	None	None	None
Diversion Header	<input checked="" type="checkbox"/> Auto	Auto	None	Caller	None
Remote Party ID	<input type="checkbox"/> None	None	None	None	None

DEFINE THESE SETTINGS TO PRESENT PILOT NUMBER ON OUTBOUND CALLS.

Select **SIP Advanced** tab and configure the following parameters as follows:

- Check **Send From In Clear**.
- Check **Indicate HOLD**.

The screenshot displays the 'SIP Advanced' configuration tab. The 'Addressing' section includes 'Association Method' (By Source IP address), 'Call Routing Method' (Request URI), and 'Suppress DNS SRV Lookups' (unchecked). The 'Identity' section lists various options, with 'Send From In Clear' checked and highlighted by a red box. A yellow callout box points to this checkbox with the text: 'CHECK SEND FROM IN CLEAR IF YOU WISH TO RESTRICT CALLER ID ON OUTBOUND'. The 'Media' section includes options like 'Allow Empty INVITE', 'Send Empty re-INVITE', 'Allow To Tag Change', 'P-Early-Media Support', 'Send SilenceSup=Off', 'Force Early Direct Media', 'Media Connection Preservation', and 'Indicate HOLD'. The 'Indicate HOLD' checkbox is checked and highlighted by a red box. A blue callout box points to this checkbox with the text: 'CHECK INDICATE HOLD'. The 'Call Control' section includes 'Call Initiation Timeout (s)' (4), 'Call Queuing Timeout (m)' (5), 'Service Busy Response' (486 - Busy Here), 'on No User Responding Send' (408-Request Timeout), 'Action on CAC Location Limit' (Allow Voicemail), 'Suppress Q.850 Reason Header' (unchecked), 'Emulate NOTIFY for REFER' (unchecked), and 'No REFER if using Diversion' (unchecked).

Go to the **ARS** form and perform the following steps:

- Create a new **ARS** form and place in service.
- Define a short code to dial out with the group id as configured in **SIP URI** tab.

The screenshot shows the ARS configuration form with the following fields and values:

- ARS Route Id: 51
- Route Name: Showpilot
- Dial Delay Time: System Default (5)
- Description: (empty)
- Secondary Dial tone: ☐ (unchecked)
- System Tone: (dropdown menu)
- Check User Call Barring: ☒ (checked)
- In Service: ☒ (checked, highlighted with a red box and a blue arrow pointing to a "CHECK IN SERVICE" callout)
- Out of Service Route: <None>
- Time Profile: <None>
- Out of Hours Route: <None>

A table is displayed below the In Service field:

Code	Telephone Number	Feature	Line Group ID
N;	N"@192.168.109.1"	Dial	20

The table row is highlighted with a red box, and a blue arrow points to it from a callout box that says: "DEFINE SHORT CODE TO DIAL OUT WITH CORRECT LINE GROUP ID AS CONFIGURED IN SIP URI TAB."

Below the table, there are two more fields:

- Alternate Route Priority Level: 3
- Alternate Route Wait Time: 30
- Alternate Route: <None>

- Modify the main **ARS** form to be out of service.
- Define the out of service route to route to the second **ARS** form.

ARS Route Id: 50

Route Name: Main

Dial Delay Time: System Default (5)

Description:

Secondary Dial tone: ☐ SystemTone

Check User Call Barring: ☒

**CHECK IN SERVICE AND DEFINE OUT OF SERVICE ROUTE FORM AS BUILT IN PREVIOUS PICTURE** → ☒ Out of Service Route: 51: Showpilot

Time Profile: <None> Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
N;	N"@192.168.109.1"	Dial	19

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

Alternate Route Priority Level: 3

Alternate Route Wait Time: 30

Alternate Route: <None>

Go to **User** configuration, select **SIP** tab and configure the following fields:

- Define the **SIP Name**, **SIP Display Name (Alias)** and **Contact**.
- Check **Anonymous**.

User Voicemail DND ShortCodes Source Numbers Telephony Forwarding D

SIP Name: 285 4

SIP Display Name (Alias): 285 4

Contact: 285 4

**DEFINE FIELDS AS PER INCOMING CALL ROUTE INCOMING NUMBER**

☒ **Anonymous**

**CHECK ANONYMOUS TO RESTRICT CALLER IDENTIFICATION ON OUTBOUND. LEAVE UNCHECKED TO SHOW PILOT NUMBER ON OUTBOUND.**

Go to **Group** configuration, select **SIP** tab and perform these steps:

- Define the **SIP Name**, **SIP Display Name (Alias)** and **Contact**.
- Check **Anonymous**.

The screenshot shows the 'SIP' configuration tab within a 'Group' configuration window. The interface includes tabs for 'Group', 'Queuing', 'Overflow', 'Fallback', 'Voicemail', 'Voice Recording', and 'Annou'. Three text input fields are visible: 'SIP Name', 'SIP Display Name (Alias)', and 'Contact'. Each field contains the text '285' followed by a red square icon and the number '4'. A red rectangular box highlights these three fields. A blue callout box with an arrow pointing to the fields contains the text: 'DEFINE FIELDS AS PER INCOMING CALL ROUTE INCOMING NUMBER'. Below the input fields is a checkbox labeled 'Anonymous', which is checked. A red rectangular box highlights this checkbox. A yellow callout box with an arrow pointing to the checkbox contains the text: 'CHECK ANONYMOUS TO RESTRICT CALLER IDENTIFICATION ON OUTBOUND. LEAVE UNCHECKED TO SHOW PILOT NUMBER ON OUTBOUND.'



## 5.9 Configuring Expansion System to Allow Fax Support

In **Line** tab, configure the **IP Office Line** as in screenshot below on IP Office Server Edition.

The screenshot shows the 'IP Office Line - Line 1' configuration page in the 'Line' tab. The 'Short Codes' and 'VoIP Settings' sub-tabs are visible. The 'Line Number' is set to 1. The 'Transport Type' is 'WebSocket Server'. The 'Networking Level' is 'SCN'. The 'Security' is 'Medium'. The 'Telephone Number' is empty. The 'Prefix' is empty. The 'Outgoing Group ID' is '99001'. The 'Number of Channels' is '250'. The 'Outgoing Channels' is '250'. The 'Gateway' section shows the 'Address' as '192 . 168 . 109 . 51', 'Location' as 'Cloud', 'Password' as '.....', and 'Confirm Password' as '.....'. The 'SCN Resiliency Options' section has three checkboxes: 'Supports Resiliency' (unchecked), 'Backs up my IP Phones' (unchecked), 'Backs up my Hunt Groups' (unchecked), and 'Backs up my IP Dect Phones' (unchecked). The 'Description' field is empty.

Select **VoIP Settings** tab, select Codec list and configure the following fields:

- Uncheck **Allow Direct Media Path**.
- Select **G.711** for **Fax Transport Support**.

The screenshot shows the 'IP Office Line - Line 1' configuration page in the 'VoIP Settings' tab. The 'Codec Selection' dropdown is set to 'System Default'. The 'Fax Transport Support' dropdown is set to 'G.711'. The 'Call Initiation Timeout (s)' is '4'. The 'Media Security' is 'Same as System (Disabled)'. The 'Out Of Band DTMF' checkbox is checked. The 'Allow Direct Media Path' checkbox is unchecked. The 'Selected' codec list includes 'G.729(a) 8K CS-ACELP', 'G.711 ALAW 64K', and 'G.711 ULAW 64K'. The 'Unused' codec list includes 'G.722 64K'. Annotations include: 'DEFINE SYSTEM CODECS' pointing to the 'Codec Selection' dropdown, 'UNCHECK ALLOW DIRECT MEDIA PATH' pointing to the 'Allow Direct Media Path' checkbox, and 'SELECT G.711 TO ENABLE FAX TRANSPORT SUPPORT' pointing to the 'Fax Transport Support' dropdown.

Create a new **ARS** form on IP Office Expansion System and perform the following steps:

- Check **In Services**.
- Add **Short Code** with **Line Group ID** of **99999** as shown in screenshot below.

**ARS**

ARS Route Id: 50

Route Name: Main

Dial Delay Time: System Default (1)

Description:

☒ In Service

Time Profile: <None>

Secondary Dial tone: ☐ SystemTone

Check User Call Barring: ☒

Out of Service Route: <None>

Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
?	.	Dial	99999

Add... Remove Edit...

Alternate Route Priority Level: 3

Alternate Route Wait Time: 30

Alternate Route: <None>

**Place in service**

**LINE GROUP ID OF IP OFFICE LINE IN EXPANSION SERVER**

Configure user in IP Office Expansion, select **SIP** tab and enter one of the DID numbers to **SIP Name**, **SIP Display Name (Alias)** and **Contact**.

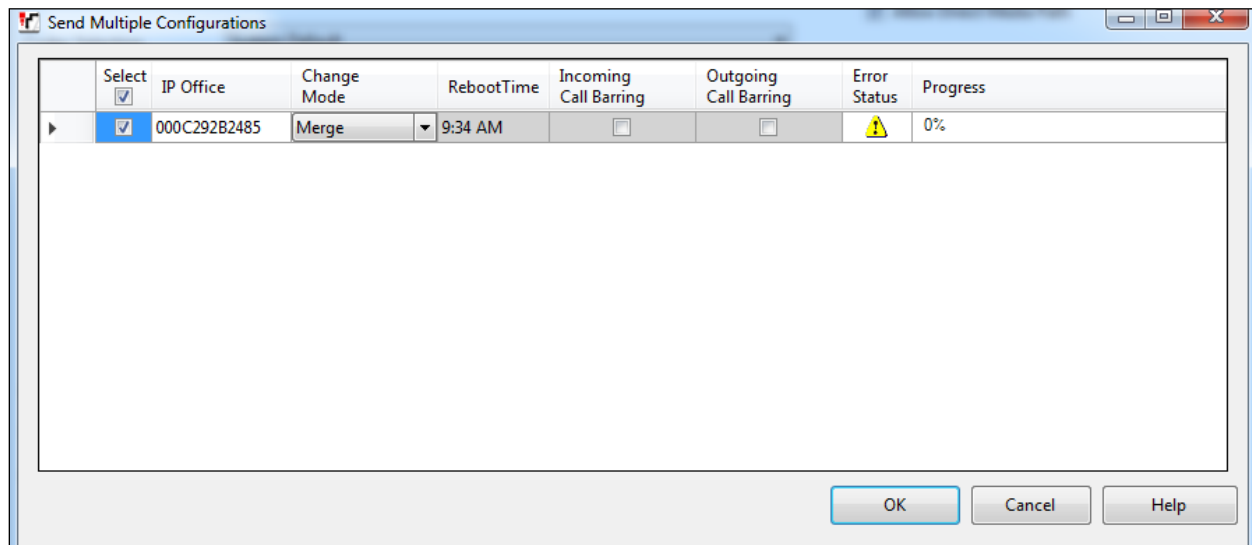
The screenshot shows the configuration interface for a user in IP Office Expansion. The 'SIP' tab is selected. The fields for 'SIP Name', 'SIP Display Name (Alias)', and 'Contact' are highlighted with a red box and contain the number '285-4'. A blue callout box points to these fields with the text 'DEFINE FIELDS AS PER INCOMING CALL ROUTE INCOMING NUMBER'. Below the fields is an 'Anonymous' checkbox.

User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In
SIP Name				285-4			
SIP Display Name (Alias)				285-4			
Contact				285-4			

☐ Anonymous

## 5.10 Save Configuration

Navigate to **File > Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections. The screen below is displayed indicating the system configuration has been changed and needs to be saved. **Merge, Immediate, When Free** or **Timed** is shown under the **Change Mode** column, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to save the configuration.



## 6. Verification Steps

The following steps may be used to verify the configuration on Avaya IP Office and OneAccess-Telstra Business SIP Trunk Service.

### 6.1 Avaya IP Office

On the PC that has IP Office Manager installed, navigate to **Start > All Programs > IP Office > System Status**. A login window appears, login with proper credentials. Click on **Trunks > Line: 2** (the SIP line configured on IP Office for SIP trunking) and verify that **Line Service State** is **In Service** with all settings as administered.

The screenshot shows the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - 000C292B2485 (10.1.20.14) - IP Office Linux PC 10.1.0.1.0 build 3". The main window has a menu bar with "Help", "Snapshot", "LogOff", "Exit", and "About". A left-hand navigation pane lists various system components: "System", "Alarms (5)", "Extensions (0)", "Trunks (2)", "Line: 1", "Line: 2" (selected), "Active Calls", "Resources", "Voicemail", "IP Networking", and "Locations". The main content area is titled "IP Office System Status" and contains tabs for "Status", "Utilization Summary", and "Alarms". The "Status" tab is active, displaying the "SIP Trunk Summary" for Line 2. The summary includes the following details:

- Line Service State: In Service
- Peer Domain Name: 192.168.109.1
- Resolved Address: 10.1.20.9
- Line Number: 2
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: G711 A, G711 Mu, G729 A
- Enable Faststart: Off
- Silence Suppression: Off
- Media Stream: RTP
- Layer 4 Protocol: TLS
- SIP Trunk Channel Licenses: 10
- SIP Trunk Channel Licenses in Use: 0
- SIP Device Features: UPDATE (Incoming and Outgoing)

A green circular progress indicator shows 0% utilization. Below the summary is a table with columns: "Cha...", "U..", "Call", "Curr...", "Time", "Remote C...", "Con...", "Caller", "Other", "Dire...", "Rou...", "Rec...", "Rec...", "Tra...", "Tra...". The first row shows "1" in the "Cha..." column and "Idle" in the "Call" column. A "Round Trip Delay" button is visible in the bottom right of the table area. At the bottom of the window, there are buttons for "Trace", "Trace All", "Pause", "Ping", "Call Details", "Graceful Shutdown", "Force Out of Service", "Print...", and "Save As...". The status bar at the bottom right shows the time "1:08:33 AM" and the status "Online".

### 6.2 Telephony Services

1. Place inbound/outbound calls, answer the calls, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnects properly.
2. Verify basic call functions such as hold, transfer, and conference.
3. Verify the use of DTMF signaling by placing call to some public Interactive Voice Response (IVR) system, and navigating the menu using phone keypad.

## 7. Conclusion

As illustrated in these Application Notes, Avaya IP Office Release 11 can be configured to interoperate successfully with OneAccess-Telstra Business SIP. This solution allows enterprise users access to the PSTN using the OneAccess-Telstra Business SIP. Please refer to **Section 2.2** for observations.

## 8. Additional References

This section references the documentation relevant to these Application Notes. Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Deploying IP Office Server Edition Solution*, Release 11.
- [2] *Deploying IP Office IP500 V2*, Release 11.
- [3] *Administering Avaya IP Office with Manager*, Release 11.

Product documentation for OneAccess-Telstra Business SIP is available from Telstra.

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