Avaya Solution & Interoperability Test Lab

Application Notes for Configuring TDS SIP Trunk Service with Avaya IP Office using UDP/RTP - Issue 0.1

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider TDS and Avaya IP Office Release 10.0.

TDS SIP Trunk Service (TDS) provides PSTN access via a SIP trunk between the enterprise and the TDS network 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 the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

TDS is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.
1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between TDS and Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists Avaya IP Office 500 V2 Release 10.0, Avaya embedded Voicemail, Avaya IP Office Application Server (with WebRTC and one-X Portal services enabled), Avaya Communicator for Windows (SIP mode), Avaya Communicator for Web, Avaya H.323, Avaya SIP, digital and analog endpoints. The enterprise solution connects to the TDS network.

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office connecting to TDS.

This configuration (shown in Figure 1) was used to exercise the features and functionality tests listed in Section 2.1. Note: NAT devices added between Avaya IP Office and the TDS network should be transparent to the SIP signaling.

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

2.1. Interoperability Compliance Testing

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

- Incoming PSTN calls to various phone types. Phone types included Avaya H.323, Avaya SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included Avaya H.323, Avaya SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls from/to the Avaya Communicator for Windows (SIP mode).
- Inbound and outbound PSTN calls from/to the Avaya Communicator for Web with basic telephony transfer feature.
• Inbound and outbound long hold time call stability
• Various call types including: local; long distance; international call; outbound toll-free; outbound to assisted operator, 411 and 911 services during the compliance testing
• SIP transport using UDP as supported
• Codec G.711MU
• Caller number/ID presentation
• Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls
• DTMF transmission using RFC 2833
• Voicemail navigation for inbound and outbound calls
• Telephony features such as hold and resume, transfer, and conference
• Fax G.711 pass-through mode
• Use Diversion header in off-net call forwarding
• Use SIP re-Invite or SIP Refer in off-net call transfer
• Twinning to mobile phones on inbound calls

Item not supported by TDS Lab Environment include the following:
• Inbound toll free call
• Fax T.38
• TLS/SRTP

2.2. Test Results
Interoperability testing of TDS was completed with successful results for all test cases with the exception of the limitation described below:

• Blind Call Transfer using Avaya 1140E SIP phone did not complete until transferee picked up the call - The expected behavior of the SIP phone was that after transferring, the phone should display “Transfer successful”. In this case, the user pressed “Trnsfr” button, answered “No” to the question of “Consult with party?” which implied the blind transfer, the transferee phone was ringing and the SIP phone should be released and displaying “Transfer successful”. Instead, the SIP phone was still displaying “Transferring” and did not released until the transferee phone answered the call. This is very minor known limitation on Avaya 1140E SIP phone. There was no user impact. Transfer was still completed with 2-way audio.

2.3. Support
For technical support on the Avaya products described in these Application Notes visit: http://support.avaya.com

For technical support on TDS SIP Trunking, contact TDS at https://tdsbusiness.com/products/voip-phone/trunking.html
3. Reference Configuration

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

The Avaya components used to create the simulated customer site included:

- Avaya IP Office 500 V2
- Avaya embedded Voicemail for IP Office
- Avaya Application Server (Enabled WebRTC and one-X Portal services)
- Avaya 9600 Series IP Deskphones (H.323)
- Avaya 11x0 Series IP Deskphones (SIP)
- Avaya 1408 Digital phone
- Avaya Analog phone
- Avaya Communicator for Windows (SIP)
- Avaya Communicator for Web

Located at the enterprise site is an Avaya IP Office 500 V2 with the MOD DGTL STA16 expansion module which provides connections for 16 digital stations to the PSTN, and the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The voicemail service is embedded on Avaya IP Office. The LAN2 port of Avaya IP Office is connected to the public IP network. Endpoints include Avaya 9600 Series IP Telephone (with H.323 firmware), Avaya 1100 Series IP Telephone (with SIP firmware), Avaya 1408D Digital Telephone, Avaya Analog Telephone, and Avaya Communicator for Windows.

A separate Windows 10 Enterprise PC runs Avaya IP Office Manager to configure and administer Avaya IP Office system.

Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user’s phones will also ring and can be answered at configured mobile phones.
For the purposes of the compliance test, Avaya IP Office users dialed a short code of 9 + N digits to send digits across the SIP trunk to TDS. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to TDS. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, TDS sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

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

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

<table>
<thead>
<tr>
<th>Avaya Telephony Components</th>
<th>Release</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya IP Office solution</td>
<td></td>
</tr>
<tr>
<td>- Avaya IP Office 500V2</td>
<td>10.0.0.3.0 build 5</td>
</tr>
<tr>
<td>- Embedded Voicemail</td>
<td>10.0.0.3.0 build 5</td>
</tr>
<tr>
<td>- Avaya Web RTC Gateway</td>
<td>10.0.0.3.0 build 10</td>
</tr>
<tr>
<td>- Avaya one-X Portal</td>
<td>10.0.0.3.0 build 9</td>
</tr>
<tr>
<td>- Avaya IP Office Manager</td>
<td>10.0.0.3.0 build 5</td>
</tr>
<tr>
<td>- Avaya IP Office Analogue PHONE 8</td>
<td>10.0.0.3.0 build 5</td>
</tr>
<tr>
<td>- Avaya IP Office VCM64/PRI U</td>
<td>10.0.0.3.0 build 5</td>
</tr>
<tr>
<td>- Avaya IP Office DIG DCPx16 V2</td>
<td>10.0.0.3.0 build 5</td>
</tr>
<tr>
<td>Avaya 1140E IP Deskphone (SIP)</td>
<td>04.04.23</td>
</tr>
<tr>
<td>Avaya 9641G IP Deskphone</td>
<td>6.6.3.02_V474</td>
</tr>
<tr>
<td>Avaya 9621G IP Deskphone</td>
<td>6.6.3.02_V474</td>
</tr>
<tr>
<td>Avaya Communicator for Windows (SIP)</td>
<td>2.1.3.237</td>
</tr>
<tr>
<td>Avaya Communicator for Web</td>
<td>1.0.16.1718</td>
</tr>
<tr>
<td>Avaya 1408D Digital Deskphone</td>
<td>R46</td>
</tr>
<tr>
<td>Avaya Analog Deskphone</td>
<td>N/A</td>
</tr>
<tr>
<td>HP Officejet 4500 (fax)</td>
<td>N/A</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>TDS Components</th>
<th>Release</th>
</tr>
</thead>
<tbody>
<tr>
<td>Perimeta Metaswitch SBC</td>
<td>3.9.40Su27</td>
</tr>
</tbody>
</table>

**Note:** Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.
5. Configure Avaya IP Office Solution

This section describes the Avaya IP Office solution configuration necessary to support connectivity to the TDS. It is assumed that the initial installation and provisioning of the Avaya IP Office 500 V2 has been previously completed and therefore is not covered in these Application Notes. For information on these installation tasks refer to Additional References Section 9.

This section describes the Avaya IP Office configuration to support connectivity to TDS system. 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 click OK button. Log in using appropriate credentials.

![Avaya IP Office Selection](image)

*Figure 2 – Avaya IP Office Selection*
5.1. Licensing

The configuration and features described in these Application Notes require the Avaya IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

To verify that there is a SIP Trunk Channels license with sufficient capacity, select **IPOffice_1 → License** on the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm that there is a valid license with sufficient “Instances” (trunk channels) in the Details pane.

![Avaya IP Office License](image)

**Figure 3 – Avaya IP Office License**
5.2. System Tab

Navigate to System (1) under the IPOffice_1 on the left pane and select the System tab in the Details pane. The Name field can be used to enter a descriptive name for the system. In the reference configuration, IPOffice_1 was used as the name in IP Office.

![Avaya IP Office System Configuration](image)

**Figure 4 - Avaya IP Office System Configuration**
5.3. LAN2 Settings

In the sample configuration, LAN2 is used to connect the enterprise network to TDS network.

To configure the LAN2 settings on the IP Office, complete the following steps. Navigate to IPOffice_1 → System (1) in the Navigation and Group Panes and then navigate to the LAN2 → LAN Settings tab in the Details Pane. Set the IP Address field to the IP address assigned to the Avaya IP Office LAN2 port. Set the IP Mask field to the mask used on the public network. All other parameters should be set according to customer requirements. Click OK to submit the change.

![Figure 5 - Avaya IP Office LAN2 Settings](image)
The VoIP tab as shown in the screenshot on the next page was configured with following settings:

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Deskphones/Softphones using the H.323 protocol to register
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to TDS system
- Check the **SIP Registrar Enable** to allow Avaya IP Deskphones/Softphones to register using the SIP protocol.
- Input **SIP Domain Name** as 10.10.98.14
- The **Layer 4 Protocol** uses UDP with UDP Port as 5060
- Verify **Keepalives** to select **Scope** as RTP-RTCP with **Periodic timeout 60** and select **Initial keepalives** as **Enabled**
- All other parameters should be set according to customer requirements
- Click **OK** to submit the changes
Figure 6 - Avaya IP Office LAN2 VoIP
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- **Select the Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in Figure 1, so the parameter was set to **Open Internet**. With this configuration, STUN Server will not be used.
- **Set the Binding Refresh Time (sec)** to **60**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTION messages to service provider to keep the connection alive.
- **Set Public IP Address** to the IP address of the Avaya IP Office LAN2 port.
- **Set Public Port for UDP** as **5060** (For SIP Trunk connecting to TDS).
- **All other parameters should be set according to customer requirements**.
- **Click OK** to submit the changes.

**Figure 7 - Avaya IP Office LAN2 Network Topology**
5.4. System Telephony Settings

Navigate to IPOffice_1 ➔ System (1) in the Navigation and Group Panes (not shown) and then navigate to the Telephony ➔ Telephony tab in the Details Pane. Choose the Companding Law typical for the enterprise location. For North America, U-Law is used. Uncheck the Inhibit Off-Switch Forward/Transfer box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. Set Hold Timeout (sec) to a valid number. Set Default Name Priority to Favor Trunk. Defaults were used for all other settings. Click OK to submit the changes.

![Figure 8 - Avaya IP Office Telephony](image-url)
5.5. System VoIP Settings

Navigate to IPOffice_1 → System (1) in the Navigation and Group Panes and then navigate to the VoIP tab in the Details Pane. Leave the RFC2833 Default Payload as default of 101. Select codec G.711 ULAW 64K which TDS supports. Click OK to submit the changes.

Figure 9 - Avaya IP Office VoIP
5.6. Administer SIP Line
A SIP Line is needed to establish the SIP connection between Avaya IP Office and TDS system. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager to create a SIP Line. Follow the steps in Section 5.6.1 to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:
- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the Use Network Topology Info field on the Transport tab

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in Section 5.6.2.

Also, the following SIP Line settings are not supported on Basic Edition:
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required
- SIP Advanced Engineering

Alternatively, a SIP Line can be created manually. To do so, right-click Line in the Navigation Pane and select New → SIP Line. Then, follow the steps outlined in Section 5.6.2. For the compliance test, SIP Line 17 was used as trunk for both outgoing and incoming calls.
5.6.1. Create SIP Line from Template

1. Create a new folder in the computer where Avaya IP Office Manager is installed (e.g., C:\TDS\Template). Copy the template file (TDS-IPO10.xml) to this folder.
2. Create the SIP Trunk from the template: Right-click on Line in the Navigation Pane, then navigate to New from Template → Open from file.

![Create SIP Line from Template Diagram]

**Figure 10 – Create SIP Line from Template**
3. Select the **Template Files (*.xml)** and select the copied template from step 1 at IP Office template directory `C:\TDS\Template\`. Click **Open** button to create a SIP line from template.

![Image of creating SIP line from IP Office template directory]

**Figure 11 – Create SIP Line from IP Office Template Directory**
4. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in Section 5.6.2.

Figure 12 – Create SIP Line from Template successfully
5.6.2. Create SIP Line Manually

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).

On the SIP Line tab in the Details Pane, configure the parameters as shown below:

- Select available Line Number, 17 in this case
- Set ITSP Domain Name to TDS SIP signaling server IP address. This field is used to specify the default host part of the SIP URI in the To and R-URI fields for outgoing calls
- Set Local Domain Name to IP address of Avaya IP Office LAN2 port. This field is used to specify the default host part of the SIP URI in the From field for outgoing calls

  **Note:** For the user making the call, the user part of the From SIP URI is determined by the settings of the SIP URI channel record being used to route the call (see SIP URI → Local URI). For the destination of the call, the user part of the To and R-URI fields are determined by dial short codes of the form 9N;N“@192.168.63.146” where N is the user part of the SIP URI and “@192.168.63.146” can be used to override the host part of the To and R-URI

- Check the In Service and Check OOS boxes
- Set URI Type to SIP
- For Session Timers, set Refresh Method to Re-invite with Timer (sec) to 1200
- Set Name Priority to Favor Trunk. As described in Section 5.4, the Default Name Priority parameter may retain the default Favor Trunk setting, or can be configured to Favor Directory. As shown below, the default Favor Trunk setting was used in the reference configuration

  - **Note:** TDS supports both re-Invite and SIP Refer in off-net transfer call during the compliance testing

- Default values may be used for all other parameters
- Click OK to commit then press Ctrl + S to save

![Figure 13 – SIP Line Configuration](image-url)
On the **Transport** tab in the Details Pane, configure the parameters as shown below:

- The **ITSP Proxy Address** was set to the IP address of TDS Signaling Server: **192.168.63.146**. This is the SIP Proxy IP address used for outgoing SIP calls.
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060**.
- The **Use Network Topology Info** parameter was set to **LAN 2**. This associates the SIP Line 17 with the parameters in the **IPOffice_1 → System (1) → LAN2 → Network Topology** tab. The **Listen Port** was set to **5060**.
- The **Calls Route via Registrar** was unchecked as TDS did not support the dynamic Registration on the SIP Trunk.
- Other parameters retain default values.
- Click **OK** to commit then press Ctrl + S to save.

![Figure 14 – SIP Line Transport Configuration](image)

The SIP URI entry must be created to match any DID number assigned to an Avaya IP Office user and Avaya IP Office will route the calls on this SIP line. Select the **SIP URI** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit...** button. In the example screen on next page, the previously configured entries are edited.

A SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.8**.
- Set **Identity** to **Auto** and **Header** to **P Asserted ID**
- For **Forwarding And Twinning**, set **Send Caller ID** to **Diversion Header**

**Note:** When using twinning feature, the calling party number displayed on the twinned phone is controlled by **Send Caller ID** parameter.

- Set **Diversion Header** to **None**
- Set **Registration** to **0**: `<None>`
- Associate this line with an incoming line group in the **Incoming Group** field and an outgoing line group in the **Outgoing Group** field. This line group number will be used in...
defining incoming and outgoing call routes for this line. For the compliance test, a new line group 17 was defined that only contains this line (line 17)

- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern
- Click **OK** to submit the changes

![SIP Line SIP URI Configuration](image)

**Figure 15 – SIP Line SIP URI Configuration**

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. The **G.711 ULAW** codec is selected. Avaya IP Office supports the codec, which is sent to the TDS. Check the **Re-invite Supported** box
- Set Fax Transport Support to G.711 from the pull-down menu. Note: TDS supports only G.711 pass-through mode during the compliance testing
- Set the DTMF Support to RFC2833 from the pull-down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833
- Default values may be used for all other parameters
- Click OK to submit the changes

Figure 16 – SIP Line VoIP Configuration
5.7. Outgoing Call Routing – Short Code

The following section describes the Short Code for outgoing calls to TDS.

To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered “9N;” short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, 9N;, this short code will be invoked when the user dials 9 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to N“@192.168.63.146”. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value N represents the number dialed by the user. The host part following the “@” is the TDS signaling server IP address
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the SIP URI tab on the SIP Line in Section 5.6.2. This short code will use this line group when placing the outbound call
- Set the **Locale** to United States (US English)
- Default values may be used for all other parameters
- Click **OK** to submit the changes

![Diagram of Short Code 9N](image)

**Figure 17 – Short Code 9N**
The feature of incoming calls from mobility extension to idle-appearance FNE (Feature Name Extension) is hosted by Avaya IP Office. The Short Code **FNE00** was configured with following parameters:

- For **Code** field, enter FNE feature code as **FNE00** for dial tone
- Set **Feature** to **FNE Service**
- Set **Telephone Number** to 00
- Set **Line Group ID** to 0
- Set the **Locale** to **United States (US English)**
- Default values may be used for other parameters
- Click **OK** to submit the changes

![Figure 18 – Short Code FNE](image-url)
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 select User in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is 2318. Select the SIP tab in the Details pane.

The values entered for the SIP Name and Contact fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line. The example below shows the settings for user 2318. The SIP Name and Contact are set to one of the DID numbers assigned to the enterprise provided by TDS. The SIP Display Name (Alias) parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the Anonymous box may be checked to withhold the user’s information from the network.

![User Configuration](image)

Figure 19 – User Configuration
One of the H.323 IP Deskphones at the enterprise site uses the Mobile Twinning feature. The following screen shows the Mobility tab for User 2318. The Mobility Features and Mobile Twinning boxes are checked. The Twinned Mobile Number field is configured with the number to dial to reach the twinned mobile telephone, in this case 91613XXX5281. Check Mobile Call Control to allow incoming calls from mobility extension to access FNE00 (defined in Section 5.7). Other options can be set according to customer requirements.

![Figure 20 – Mobility Configuration for User](image-url)

Figure 20 – Mobility Configuration for User
5.9. Incoming Call Route

An Incoming Call Route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by service provider. To create an incoming call route, select Incoming Call Route in the left Navigation Pane, then right-click in the center Group Pane and select New (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 Group 17 defined on the SIP URI tab on the SIP Line in Section 5.6.2
- Set the Incoming Number to the incoming DID number on which this route should match.
- Default values can be used for all other fields

![Incoming Call Route Configuration](image1)

**Figure 21 – Incoming Call Route Configuration**

On the Destination tab, select the destination extension from the pull-down menu of the Destination field. In this example, incoming calls to 6085552318 on line 17 are routed to Destination 2318 as below screenshot:

![Incoming Call Route for Destination 2318](image2)

**Figure 22 – Incoming Call Route for Destination 2318**
For Feature Name Extension Service testing purpose, the incoming calls to DID number 6085558191 were configured to access FNE00. The Destination was appropriately defined as FNE00 as below screenshot:

![Incoming Call Route for Destination FNE](image)

**Figure 23 – Incoming Call Route for Destination FNE**

For Voice Mail testing purpose, the incoming calls to DID number 6085558192 were configured to access VoiceMail. The Destination was appropriately defined as VoiceMail as below screenshot:

![Incoming Call Route for Destination Voice Mail](image)

**Figure 24 – Incoming Call Route for Destination Voice Mail**

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.
6. TDS SIP Trunk Configuration

TDS is responsible for the configuration of TDS SIP Trunk Service. The customer must provide the IP address used to reach the Avaya IP Office LAN2 port at the enterprise. TDS will provide the customer necessary information to configure the SIP connection between Avaya IP Office and TDS. The provided information from TDS includes:

- IP address and port number used for signaling or media servers through any security
- DID numbers
- TDS SIP Trunk Specification (If applicable)
7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP Line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** for each channel (The below screen shot showed 2 active calls at present time).

![Figure 25 – SIP Trunk status](image)

**Figure 25 – SIP Trunk status**
- Use the Avaya IP Office System Status application to verify that no alarms are active on the SIP line. Launch the application from Start → Programs → IP Office → System Status on the PC where Avaya IP Office Manager was installed. Select Alarm → Trunks to verify that no alarms are active on the SIP line.

![IP Office System Status](image)

**Figure 26 – SIP Trunk alarm**

- Verify that a phone connected to the PSTN can successfully place a call to Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Use a network sniffing tool e.g., Wireshark to monitor the SIP signaling between the enterprise and TDS. The sniffer traces are captured at the LAN2 port interface of the Avaya IP Office.
8. Conclusion

TDS passed compliance testing excepting the limitation in Section 2.2. These Application Notes describe the procedures required to configure the SIP connections between Avaya IP Office and the TDS system as shown in Figure 1.

9. Additional References


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

Product documentation for TDS SIP Trunk may be found at: https://tdsbusiness.com/products/voip-phone/trunking.html
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