Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Sonexis ConferenceManager with Avaya IP Office using a SIP trunk – Issue 1.0

Abstract

These Application Notes describe the procedure for configuring Sonexis ConferenceManager to interoperate with Avaya IP Office using a SIP trunk.

Sonexis ConferenceManager is an in-house audio conferencing bridge that eliminates the costly pay-as-you-go fees of subscription-based services, while setting new standards for security and ease of use. Sonexis ConferenceManager is designed to work within existing voice and data networks — and available with a fully integrated Web conferencing option

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 procedure for configuring Sonexis ConferenceManager (herein referred to as ConferenceManager) to interoperate with Avaya IP Office.

ConferenceManager is an in-house audio conferencing bridge that eliminates the costly pay-as-you-go fees of subscription-based services, while setting new standards for security and ease of use. ConferenceManager is designed to work within existing voice and data networks, and ConferenceManager is available with a fully integrated Web conferencing option.

These Application Notes assume that Avaya IP Office is already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document.

- SIP trunk configuration in IP Office
- Short Code for call route
- Incoming Call Route

2. General Test Approach and Test Results

The general test approach was to place calls to and from ConferenceManager. The main objectives were to verify the following:

- Inbound calls
- Outbound calls
- Hold / Resume
- Call termination (origination/destination)
- Transfer (blind/consult)
- Conference (client initiated/host initiated)
- DTMF
- ANI/DNIS

2.1. Interoperability Compliance Testing

The interoperability compliance testing included features and serviceability tests. The focus of the compliance testing was primarily on verifying the interoperability between ConferenceManager and Avaya IP Office.

2.2. Test Results

The test objectives were verified. For serviceability testing, ConferenceManager operated properly after recovering from failures such as cable disconnects, and resets of ConferenceManager and Avaya IP Office.

2.3. Support

Technical support for the ConferenceManager solution can be obtained by contacting Sonexis:

- URL – CustomerCare@sonexis.com
- Phone – (866) 676-6394

3. Reference Configuration

Figure 1 illustrates the configuration used in these Application Notes. The sample configuration shows an enterprise with Avaya IP Office. Endpoints include an Avaya 1616-I IP Telephone, a 4625SW IP Telephone, and an Avaya 1416 Digital Telephone on IP Office.

Note: An Avaya S8300D Server and an Avaya G450 Media Gateway were included to simulate PSTN calls.

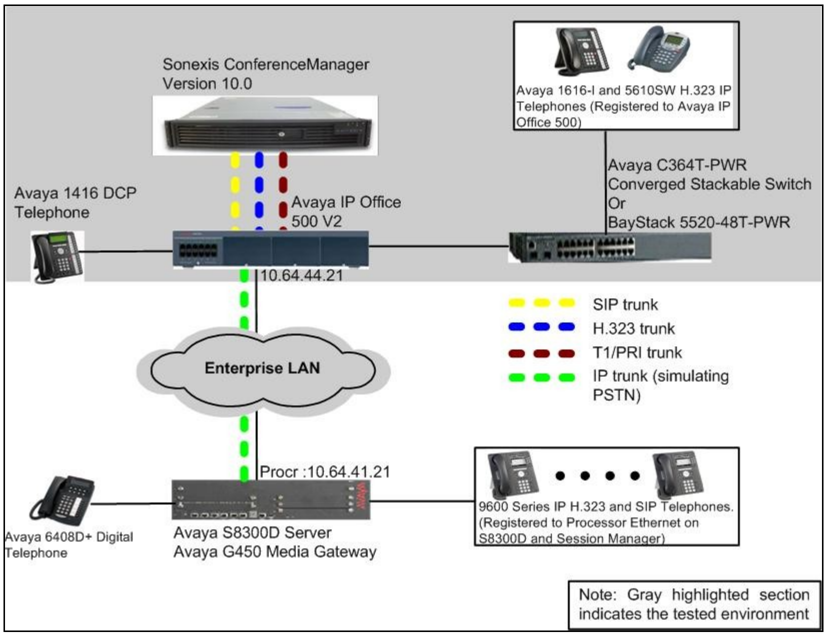


Figure 1: Test Configuration of Sonexis ConferenceManager

4. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment		Software/Firmware
Avaya IP Office 500 V2		7.0(12)
Avaya IP Office Manager on Windows XP Professional 2002 with SP3		9.0(3)
Avaya S8300D Server w/ G450 Media Gateway (used to simulate PSTN calls)		6.0.1
Avaya H.323 IP Telephones on IP Office		
	4625SW (H.323)	2.9.1
	1616-I (H.323)	1.22
Avaya 1416 Digital Telephone		-
Avaya H.323 IP SIP Telephones on Avaya Aura ® Communication Manager (simulating PSTN phones)		
	9620 (SIP)	2.6.4
	9630 (SIP)	2.6.4
	9620 (H.323)	3.1
	9630 (H.323)	3.1
	9650 (H.323)	3.1
Sonexis on Windows Server 2008 with SP 2		10.0

5. Configure Avaya IP Office

This section describes the steps required for configuring Avaya IP Office. During the compliance test, a SIP trunk was utilized between Avaya IP Office and ConferenceManager.

The procedures include the following areas:

- Verify SIP trunk Channels License
- Configure LAN interface
- Enable SIP Trunk
- Create the static SIP line
- Configure SIP URI parameters for the SIP Line
- Configure VoIP Parameters for the SIP Line
- Configure a short code to route calls through the SIP trunk
- Create an Incoming Call Route for the Inbound SIP calls

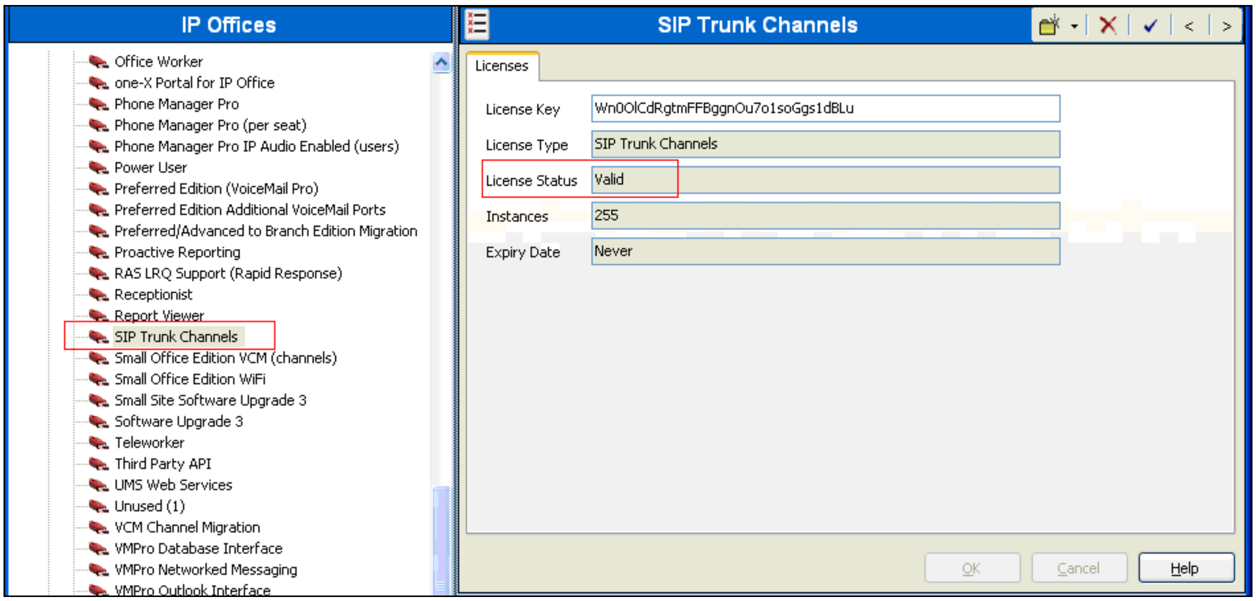
These steps are performed from the Avaya IP Office Manager.

5.1. Verify SIP Trunk Channels License

IP Office is configured via the IP Office Manager application. Log into the PC running the Avaya IP Office Manager application, and select **Start → All Programs → IP Office → Manager** to launch the Manager application. Select the proper IP Office system if there are more than one IP Office system, and log in with the appropriate credentials.

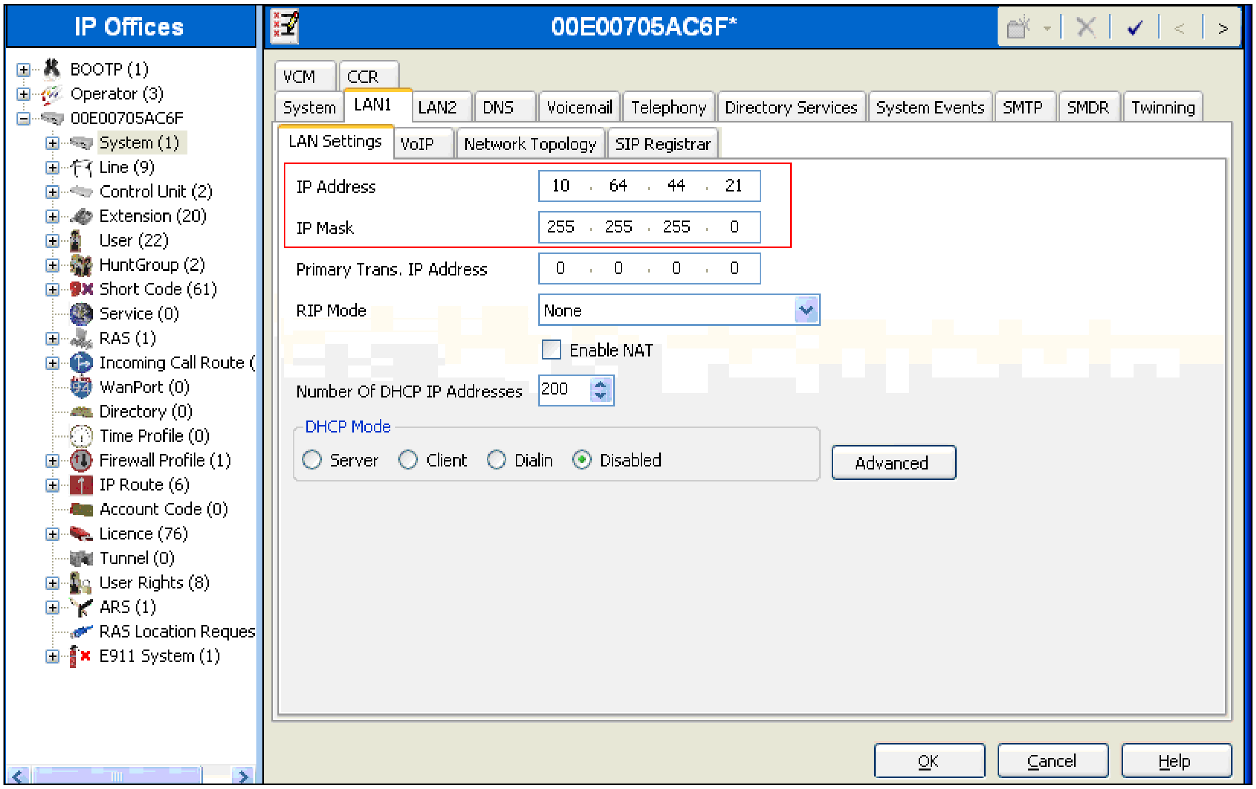
From the configuration tree in the left pane, select **License → SIP Trunk Channels** to display the SIP Trunk Channels screen in the right pane. Verify that the **License Status** field is set to **Valid**.

If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



5.2. Configure LAN interface

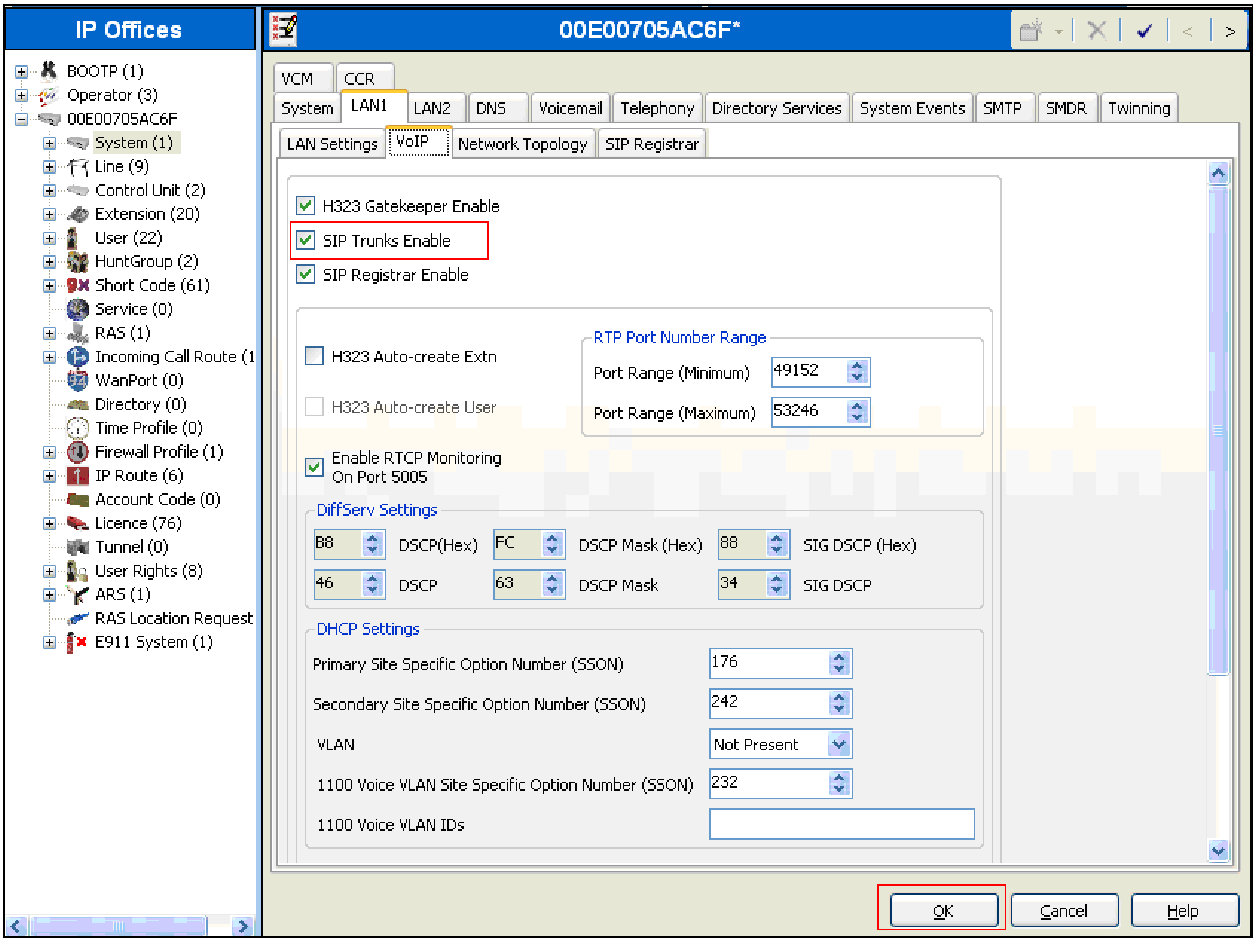
From the configuration tree in the left pane, select **System** to display the System screen in the right pane. Click the **LAN1** tab. Under the **LAN1** tab, select the **LAN Settings** sub-tab, and provide **IP Address** and **IP Mask**.



5.3. Enable SIP Trunk

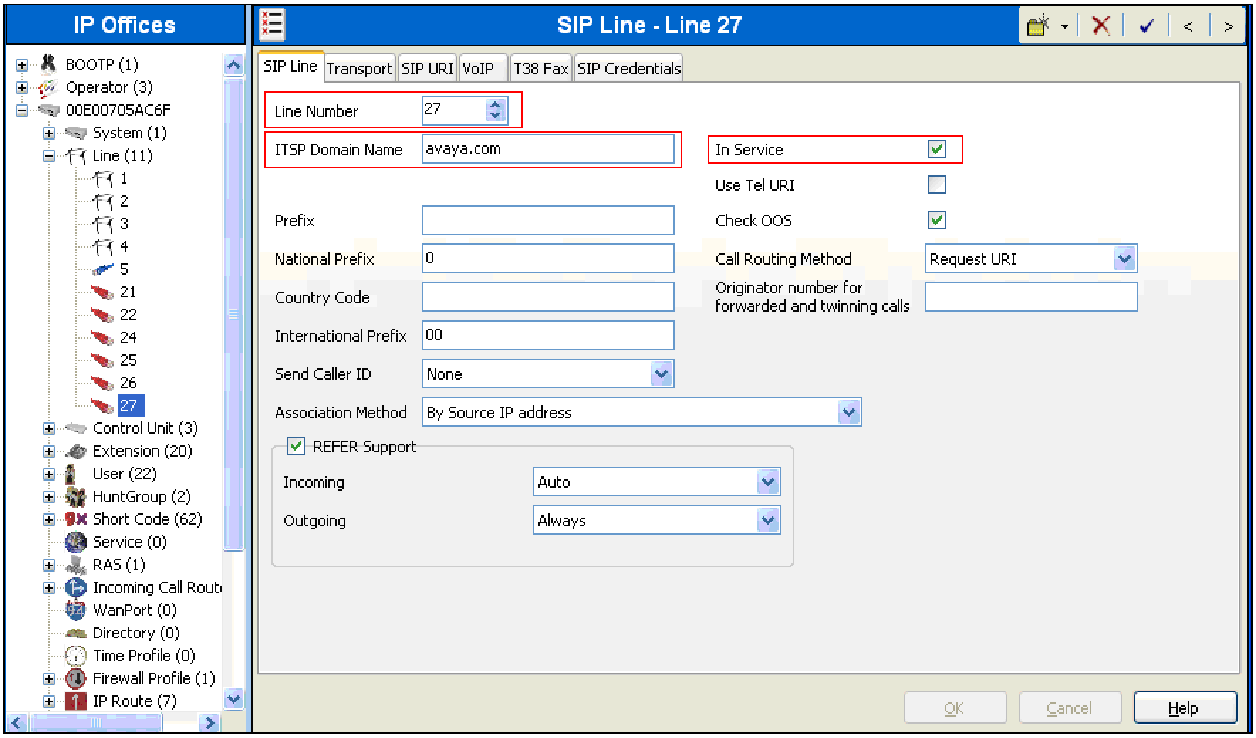
Under the **LAN1** tab, select the **VoIP** sub-tab and check the **SIP Trunks Enable** box. Click the **OK** button.

Note: During the initial configuration of Avaya IP Office, the LAN1 was configured as a private network (LAN) and the LAN2 was configured as a public network (WAN). Avaya IP Office can support SIP extensions on the LAN1 and/or LAN2 interfaces. However, the compliance test used the LAN1 interface for a SIP trunk termination.



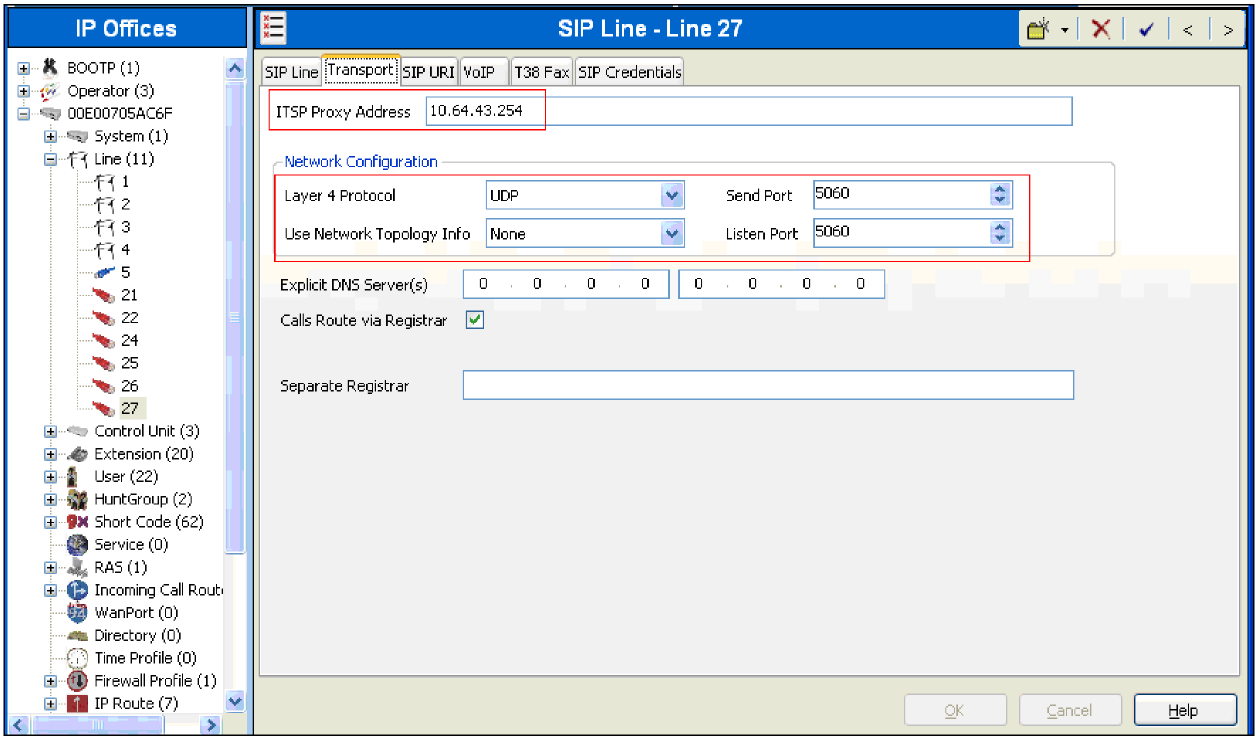
5.4. Create SIP Lines for a SIP Trunk

Select **Line** in the left pane. Using the right mouse click, select **New → SIP Line** [not shown], and create a new **Line Number**. Enter an appropriate domain on the ITSP Domain Name field, and check the **InService** check box. During the compliance test, a SIP line (27) was configured.



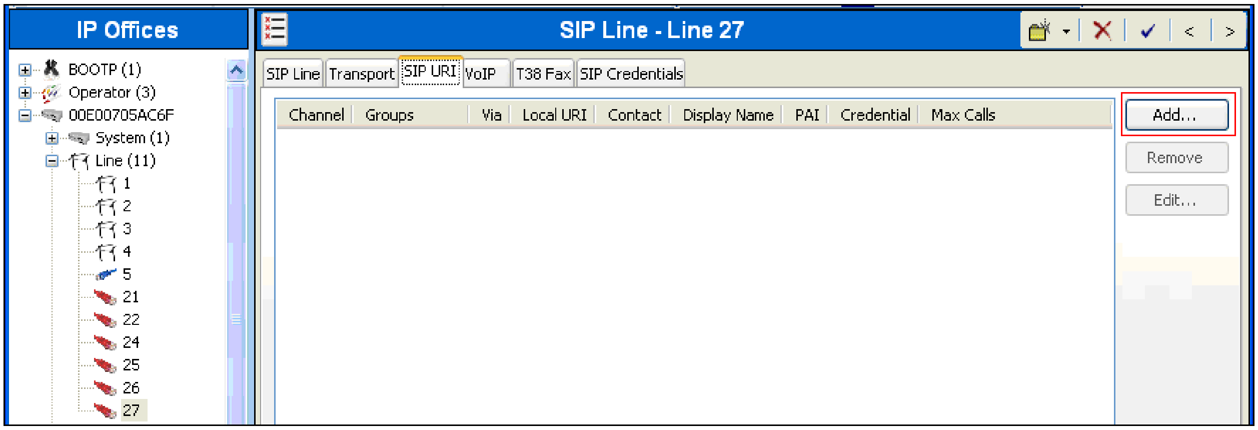
Select the **Transport** sub-tab, and provide the following information:

- **ITSP Proxy Address** – Enter the IP address of the far-end SIP termination point.
- **Layer 4 Protocol** – Select **UDP**.
- **Use Network Topology Info** – Select **None**.

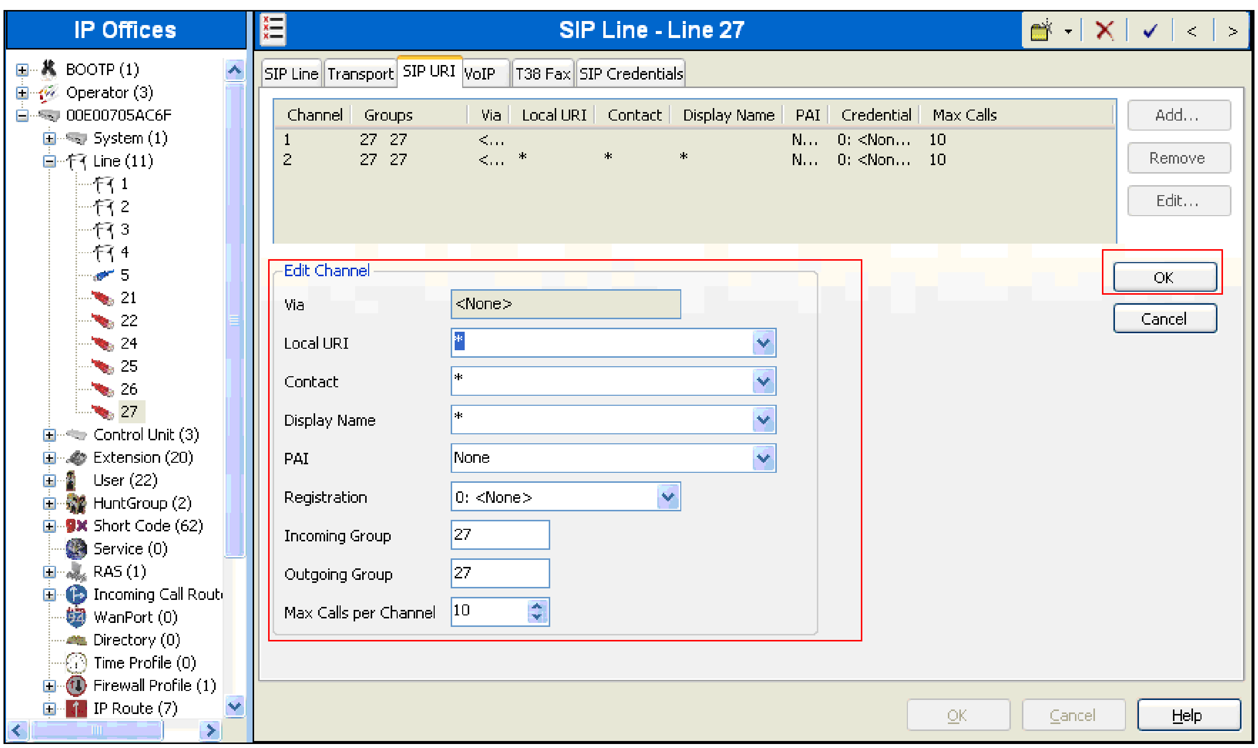


5.5. Configure SIP URI Parameters for the SIP Line

Select the **SIP URI** tab to configure SIP URI parameters for the SIP Line. Click on the **Add** button.



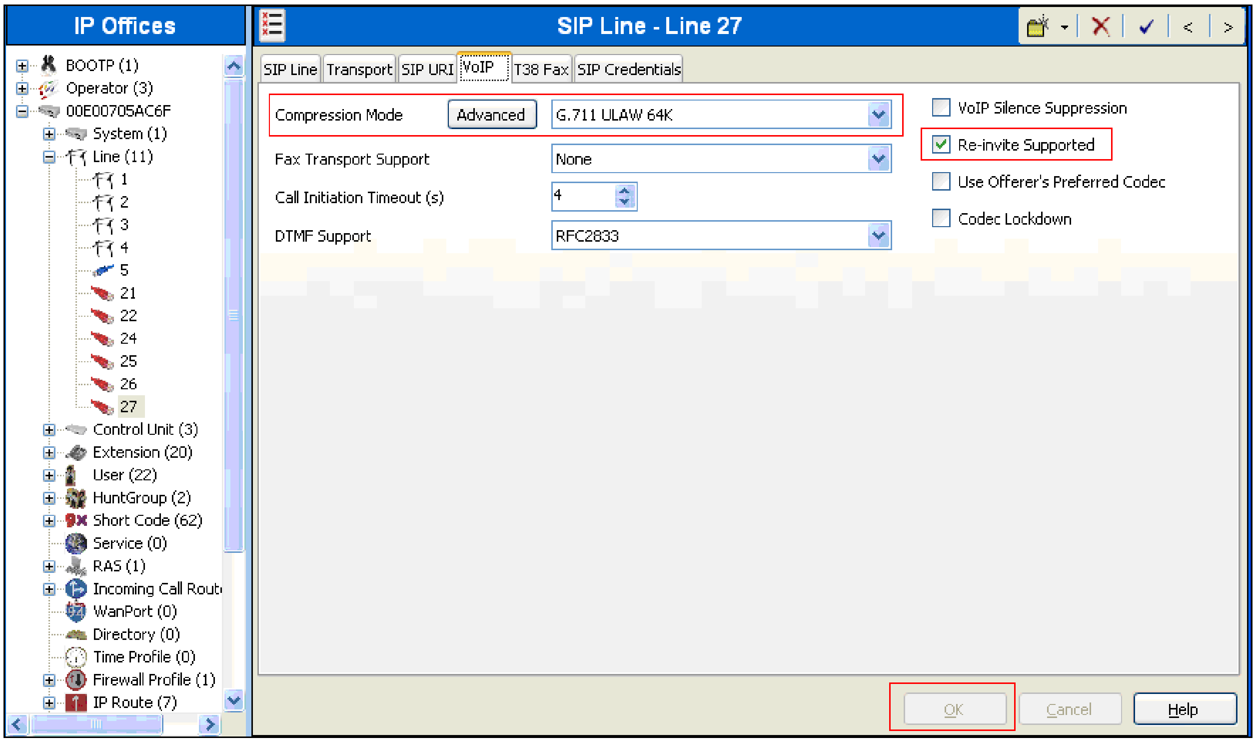
Select * for the **Local URI**, **Contact**, and **Display Name** fields. Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. The **Incoming Group** field will be used for mapping inbound calls from the SIP trunk to local stations. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in **Section 5.7**. Use default values for all other fields. Click the **OK** button.



5.6. Configure VoIP Parameters for the SIP Line

Select the **VoIP** tab to Configure VoIP parameters for the SIP Line. For **Compression Mode**, select **G.711 ULAW 64K**, since ConferenceManager only supports G.711MU. Check **Re-invite Supported** check boxes.

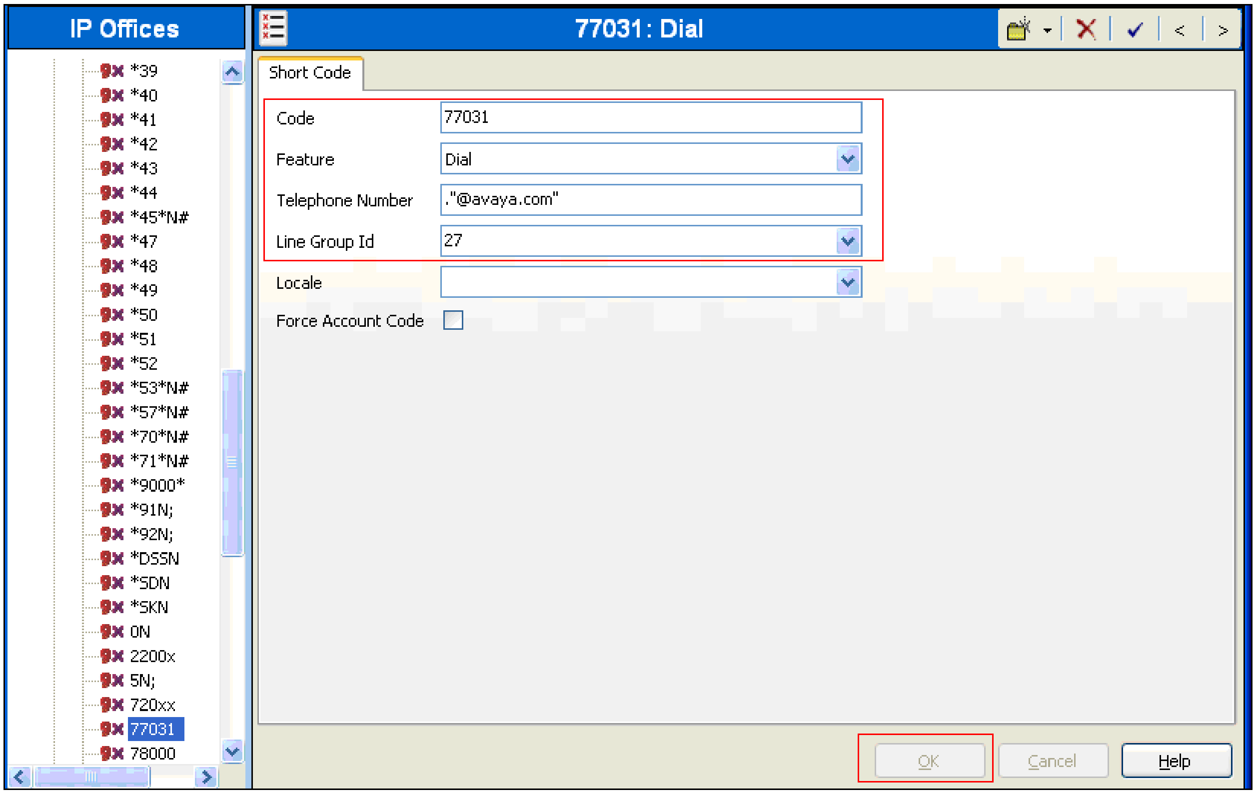
Click the **OK** button.



5.7. Configure a Short Code to Route Calls through the SIP trunk

Select **Short Code** in the left panel. Right click and select **Add**. Enter **77301;** where extension **77301** will be routed to ConferneceManager, in the **Code** text box. Select **Dial** for the **Feature** field. Enter the **Outgoing Group** number created in **Section 5.5** for the **Line Group Id** field. Enter **."@avaya.com"** for the **Telephone Number** field. Use default values for all other fields. Click the **OK** button.

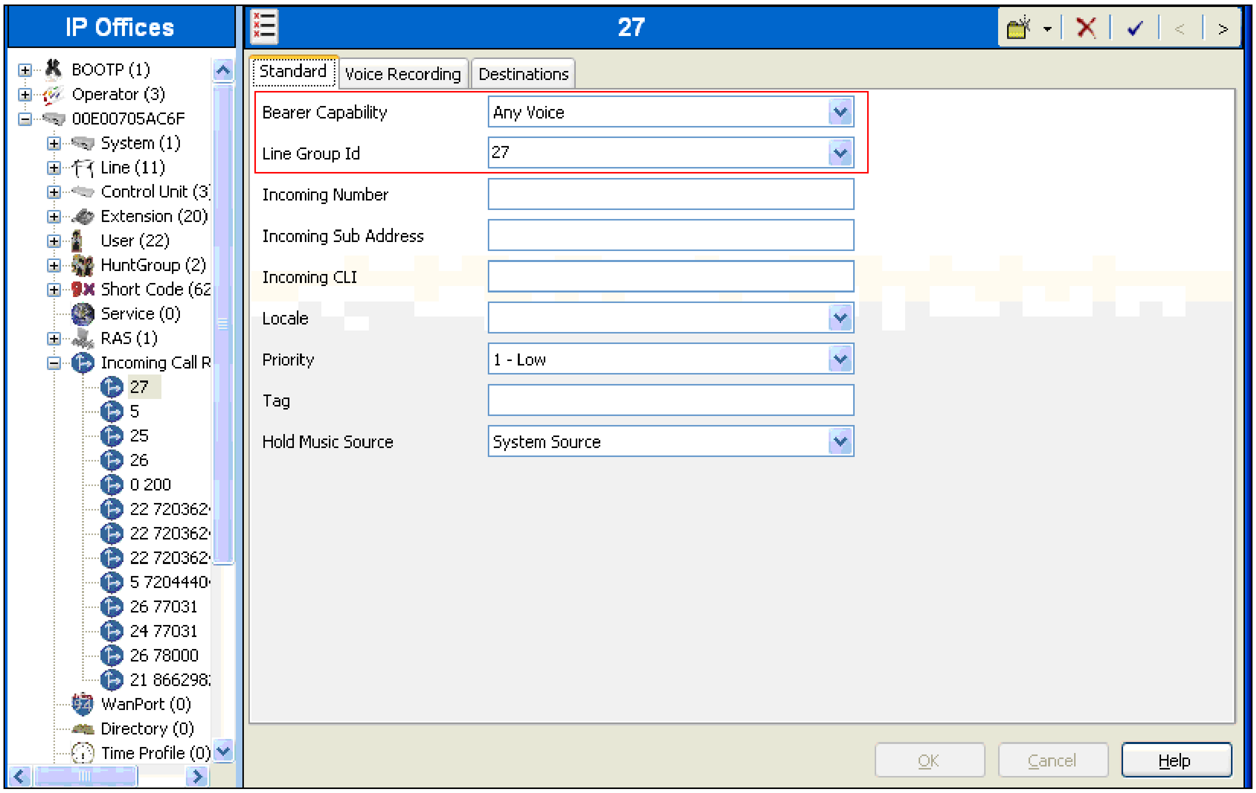
Note: When extension 77031 was dialed, the call routed thru the SIP trunk 27.



5.8. Create an Incoming Call Route for the Inbound SIP Calls

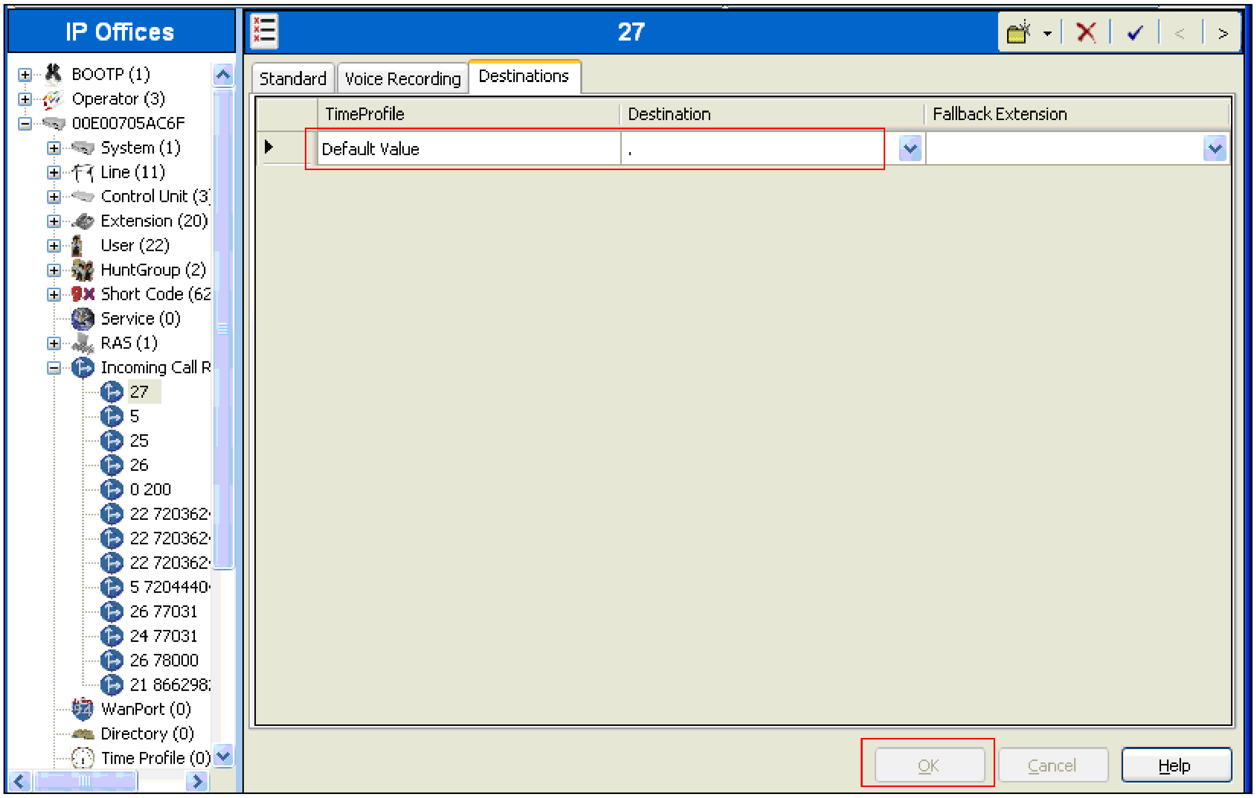
Select **Incoming Call Route** in the left pane. Right-click and select **New**.
Enter the following:

- **Any Voice** for the **Bearer Capability** field.
- Enter the **Incoming Group** number created for the URI in **Section 5.5** in the **Line Group Id** field.
- Use default values for all other fields.



Next, navigate to the **Destinations** tab and enter "**.**" under the **Destination** field.

Click the **OK** button.



After making the changes, click on the floppy disk icon (not shown) to push the changes to the IP Office system and have them take effect

Note: *Changes will not take effect until this step is completed. This may cause a reboot of Avaya IP Office causing service disruption.*

6. Configure the Sonexis ConferenceManager

Sonexis installs, configures, and customizes the ConferenceManager application for their end customers. Thus, this section only describes the interface configuration, so that ConferenceManager can talk to Avaya IP Office. By the request of Sonexis, the only codec tested during the compliance test was G.711MU.

The procedures for setting up ConferenceManager for a SIP trunk include the following areas:

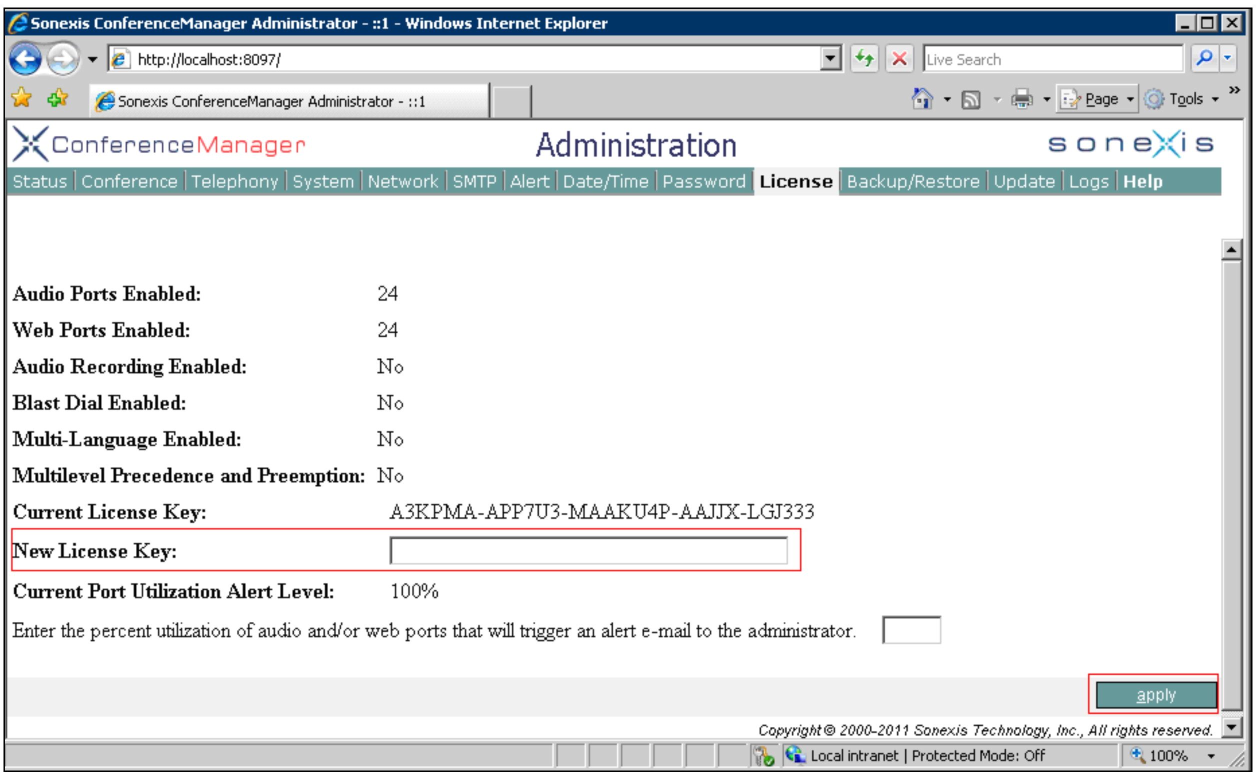
- Installing License
- Configure Telephony

6.1. Install SIP Trunk license

Launch a web browser, enter **https://<IP address of ConferenceManager>:8097** in the URL, and log in with the appropriate credentials. Navigate to the **License** menu. Enter an appropriate license for SIP trunk in the New License Key field.

Click on the **Apply** button.

Note: During the test, Sonexis provide the licenses for SIP, H323 and PRI trunks.

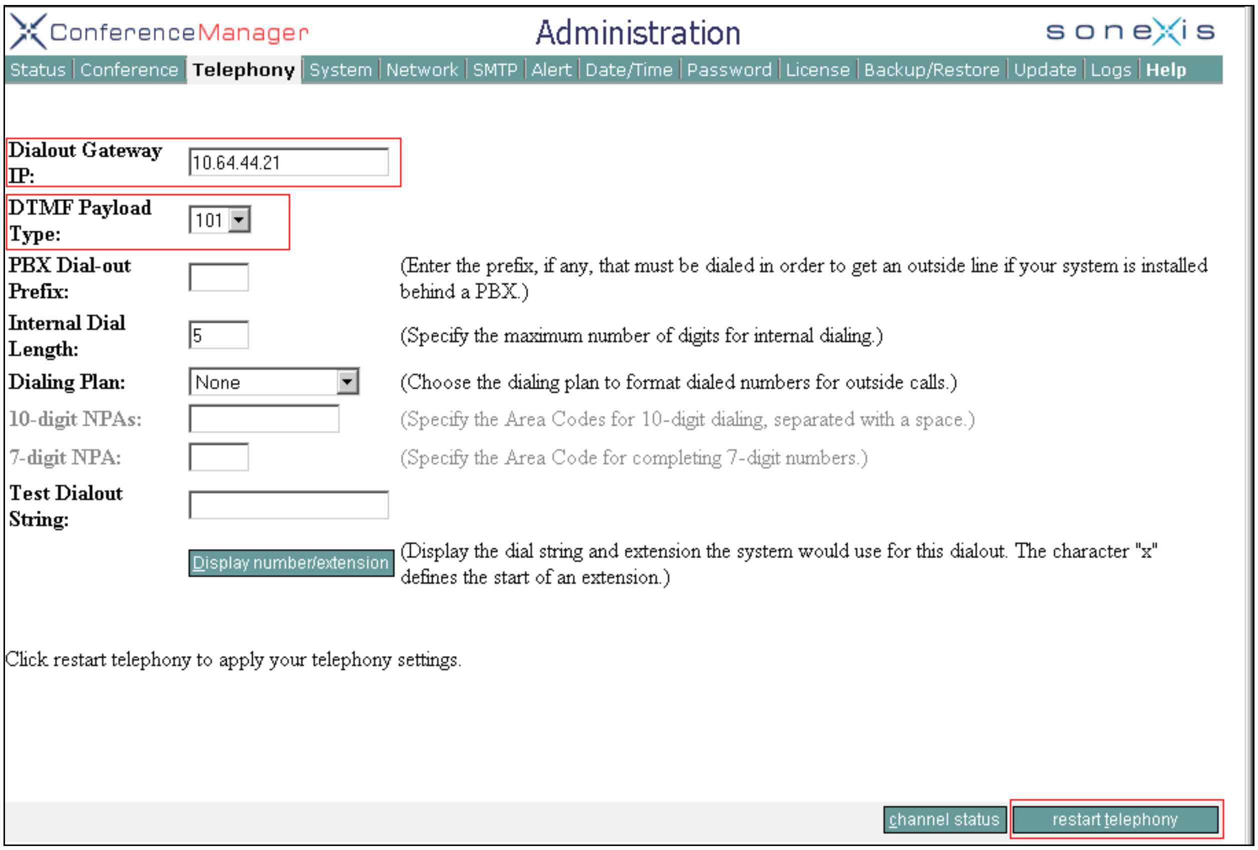


6.2. Configure Telephony

Select the **Telephony** tab and provide the following information:

- **Dialout Gateway IP:** Enter the far-end SIP trunk termination point. During the compliance test, it should be the IP Office LAN1 IP address.
- **DTMF Payload Type**: Set the payload type to **101**

Click on the **restart telephony** button.



7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya IP Office and ConferenceManager.

7.1. Verify Avaya IP Office

From a PC running the Avaya IP Office Monitor application, select **Start** → **All Programs** → **IP Office** → **System Status** to launch the application. From the **Avaya IP Office System Status** screen, select **Trunks** → **Line 27** from the left pane and verify the trunk is **Idle** under the **Current State** field.

8. Conclusion

These Application Notes describe the procedures required to configure Sonexis ConferenceManager to interoperate with Avaya IP Office through a SIP trunk. Sonexis ConferenceManager successfully passed compliance testing.

9. Additional References

The following Avaya product documentation can be found at http://support.avaya.com

[1]*IP Office 7.0 Standard Version Installation,* Issue 23k, May 2011, Document Number 15-601042

[2]*IP Office Release 7.0 Manager 9.0*, Issue 26h, May 2011, Document Number 15-601011

Sonexis product documentation can be requested at the following site:
http://www.sonexis.com/access/index.asp?id=40&Program=DevConnect

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.