Avaya Solution & Interoperability Test Lab

Application Notes for configuring Unified Dispatch Unibook platform with Avaya IP Office 9.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps for Unified Dispatch Unibook platform using Dialogic HMP to successfully interoperate with Avaya IP Office Server Edition 9.1 and 500V2 Expansion. Unified Dispatch Unibook platform is an IVR application server that connects to IP Office via a SIP Trunk.

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.

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 configuration steps for Unified Dispatch Unibook platform (IVR) to successfully interoperate with Avaya IP Office 9.1 Server Edition and 500v2 Expansion (IP Office). IVR connects to IP Office via a SIP Trunk.

Unified Dispatch UniBook platform integrates with PBX systems and ground transportation scheduling and dispatch system to provide real time status and booking for transportation services. This integration enables users to place phone calls between PBX and UniBook System. All other elements of UniBook System interconnecting with dispatch systems, web services, etc. are detailed in various documents and could be provided upon request.

2. General Test Approach and Test Results

The general test approach was to place calls from simulated PSTN using SIP or PRI trunks into IP Office which based upon DNIS forwards the calls to IVR using the SIP trunks established between IVR and IP Office. **Note: Two separate SIP trunks were established; one between IP Office Server and IVR and another between IP Office 500V2 Expansion and IVR**. IVR initiates the **sample application** which has a variety of self-service prompts including transferring the call to an agent on IP Office. The main objectives were to verify the following:

- Inbound call from an IP Office extension to SIP connection between IP Office and IVR
- Inbound call from PSTN to SIP connection between IP Office and IVR using PRI trunk on IP Office 500V2 Expansion
- Inbound call from PSTN to SIP connection between IP Office and IVR using SIP trunk on IP Office Server
- Calls can be transferred to another extension/agent from the IVR application
- IVR sample application can recognize DTMF tones using RFC2833
- IVR sample application can recognize in-band DTMF tones
- Serviceability

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

The interoperability compliance testing included feature and serviceability testing. The focus of interoperability compliance testing was primarily to verify that the IVR sample application can respond to DTMF tones. The scope of testing included the navigation of the paths provided by the IVR sample application using DTMF including transfer to an agent/extension on IP Office.

The serviceability testing focused on verifying the ability of IVR to recover from adverse conditions, such as power failures and disconnecting cables to the IP network.

2.2. Test Results

All functionality and serviceability test cases were completed successfully with following results/observations:

- IVR does not support shuffling/direct media.
- IVR only supports G711MU codec.
- IVR needs to be configured as UDP for transport.
- Outbound calls from IVR are supported but were not tested.
- IVR does not send an OPTIONS message to IP Office but responds to OPTIONS message from IP Office.
- IVR cannot hold the call but calls held by the calling party had no adverse effect
- Since IVR is configured to handle both in-band and out-of-band DTMF tones, making calls using speakerphone may have unpredictable results because of the ambient noise.
- IP Office supports consultative Refer from IVR. Blind Refer is not supported by IP Office.

2.3. Support

To obtain technical support for IVR, contact Unified Dispatch via web, email or phone.

- Web: http://www.unified-dispatch.com/contact/
- Email: support@unified-dispatch.com
- Phone: (626) 219-0800, select Option 1

3. Reference Configuration

Figure 1 illustrates the configuration used for testing. In this configuration, Avaya IP office interfaces with Unified Dispatch Unibook platform via SIP trunk.

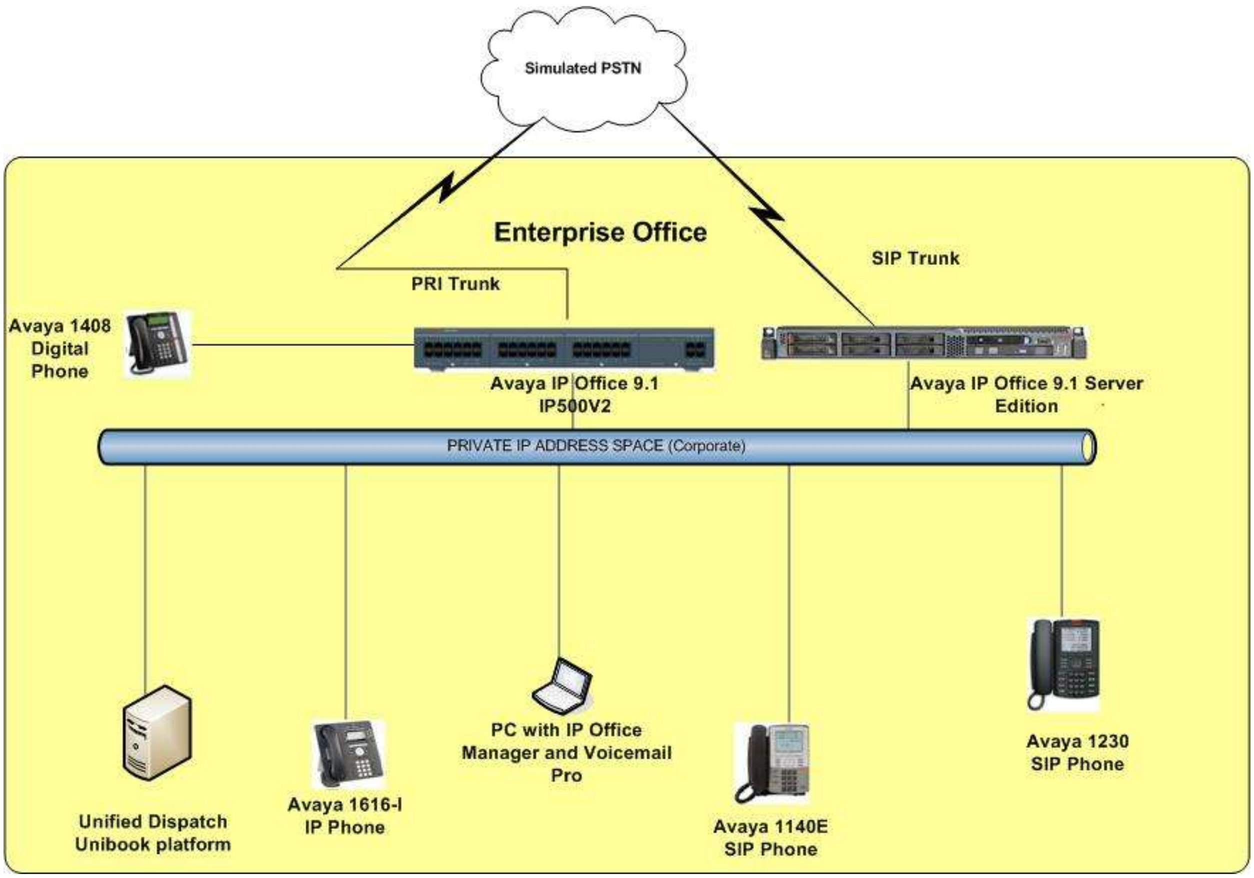


Figure 1: Avaya IP Office Server/IP500V2 Expansion and Unified Dispatch Unibook platform

3.1. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Server Edition running on a HP Server	9.1.2 build 91
Avaya IP Office 500v2 Expansion	9.1.2 build 91
Avaya 11xx/12xx SIP Deskphone	SIP Release SIP12x0.4.4.18
Avaya 1616-1 H323 Deskphone	H323 Release 1.360A
Unified Dispatch Unibook platform running on Windows Server 2008 R2 Standard Pack 1	Release 6.4.2.510015
Dialogic HMP	Release 3.0 Service Update 354

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.

4. Configure Avaya IP Office

The configuration and verification operations illustrated in this section were all performed using IP Office Manager. The information provided in this section describes the configuration of IP Office for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 8**.

The configuration operations described in this section can be summarized as follows:

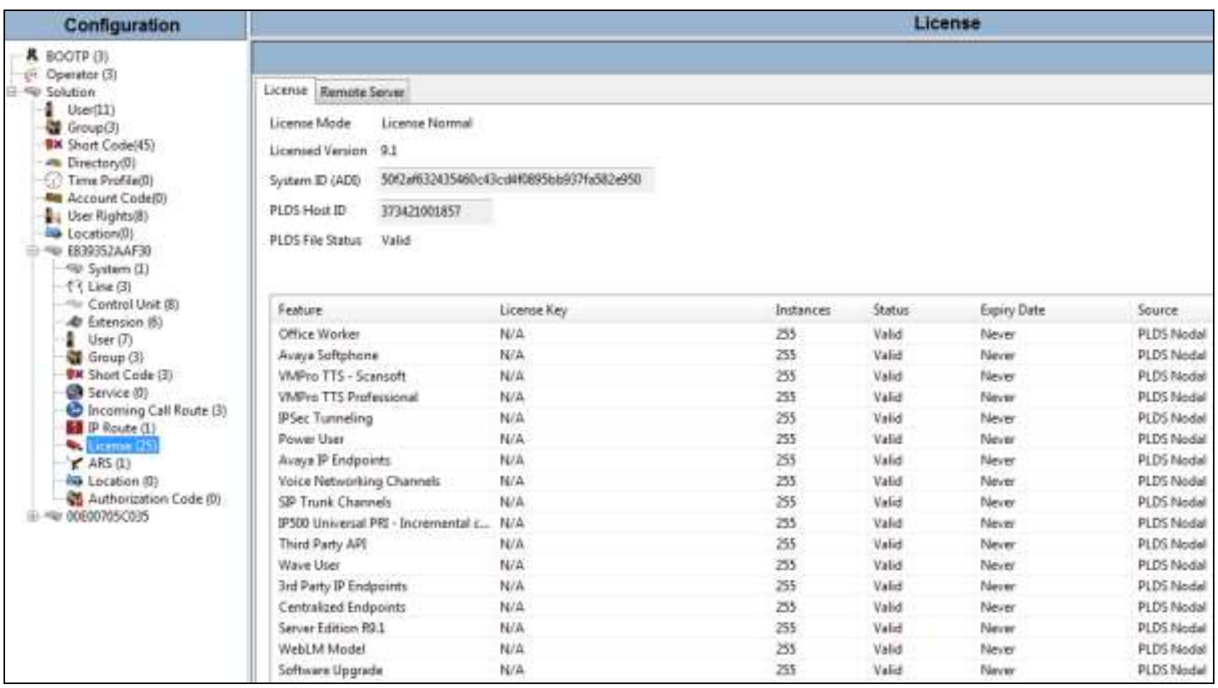
- Verify Licensed SIP Trunks
- Verify SIP Sessions
- Administer SIP Trunk
- Add IVR dial Short code

Note: For PSTN calls, a PRI trunk was configured to IP Office 500V2 Expansion and a SIP Trunk was configured to IP Office Server. IP Office Server does not support PRI Trunks.

4.1. Verify Licensed SIP Trunks

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

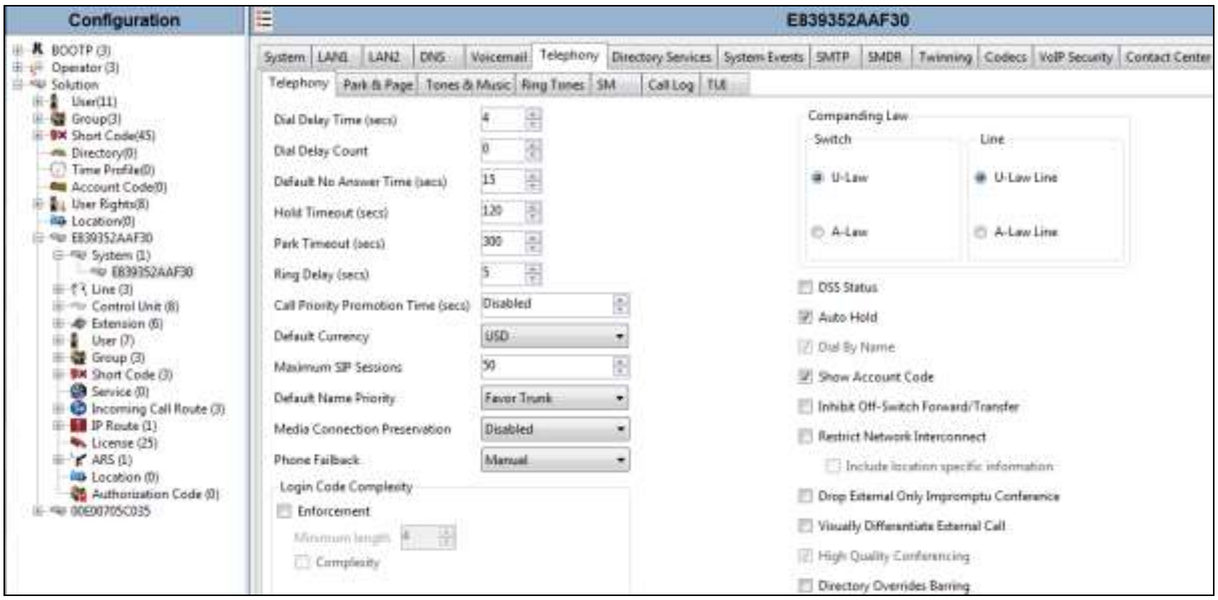
- Navigate **Solution**→<*IP Office Name*>→**License** from the **Configuration** menu.
- Verify that **SIP Trunk Channels** has sufficient licenses available.



4.2. Verify SIP Sessions

To allow SIP sessions to be established between IP Office and IVR, the maximum number of SIP Session must be verified as follows:

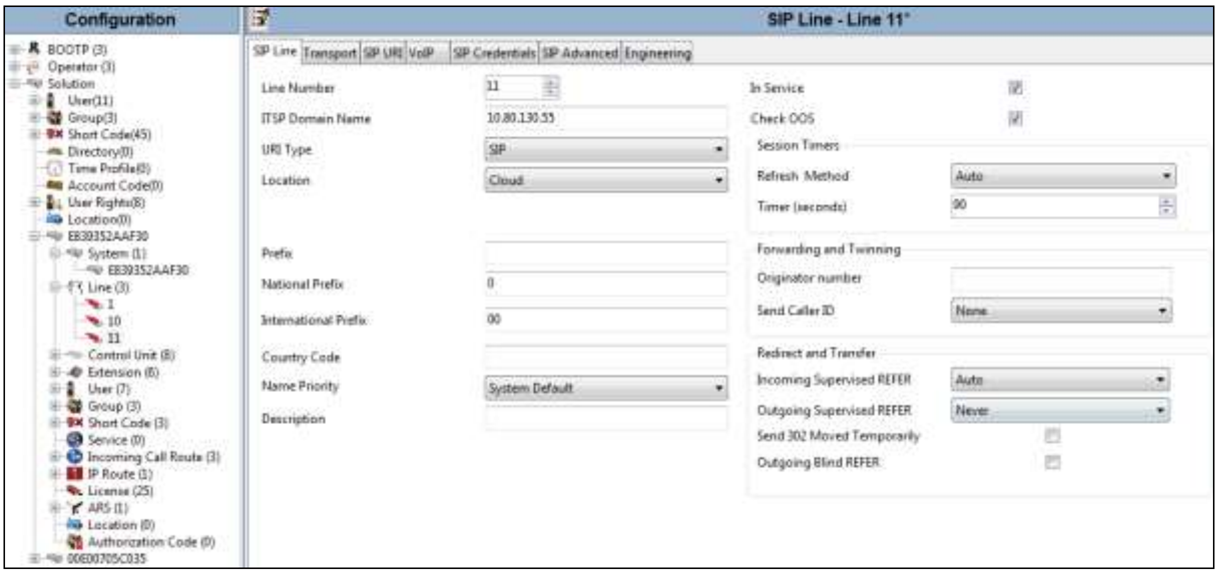
- Select **Solution**→<***IP Office Name***>→**System** from the **Configuration** menu.
- Select the **Telephony** tab.
- Verify the **Maximum SIP Sessions** value. Note if there are not enough sessions configured, the calls will fail.



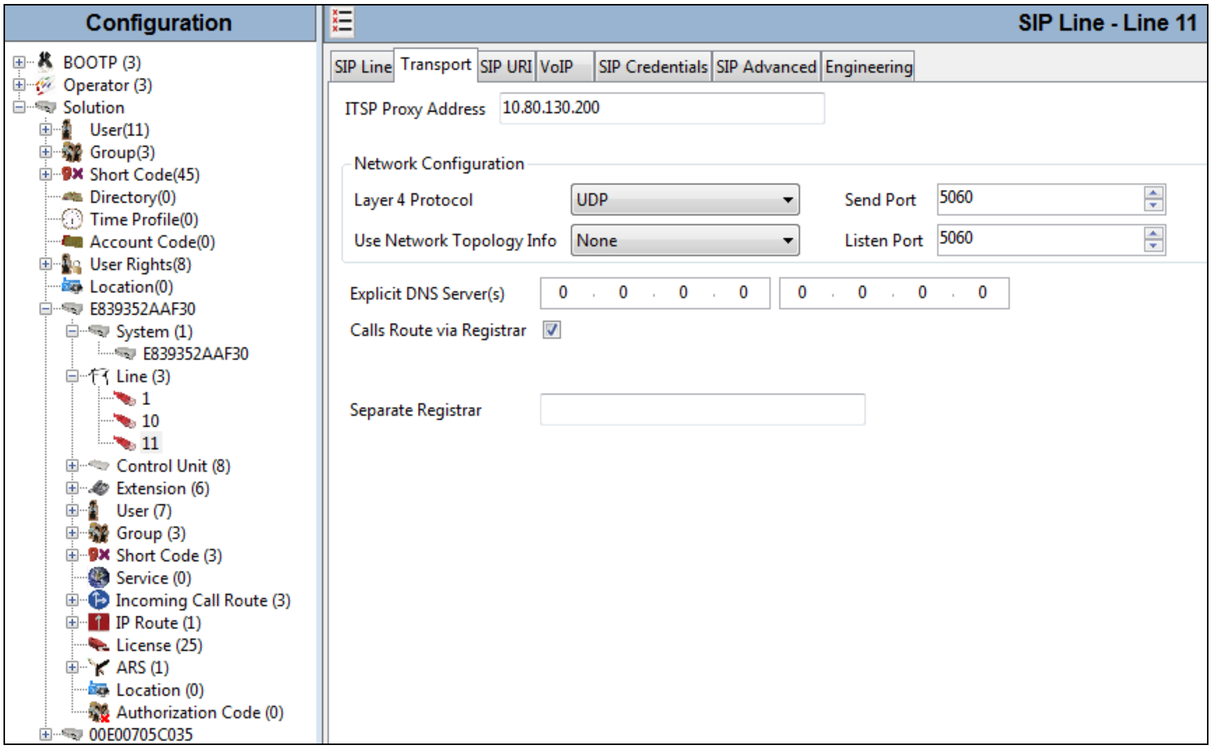
4.3. Administer SIP Line

A SIP Line is required to make calls to IVR.

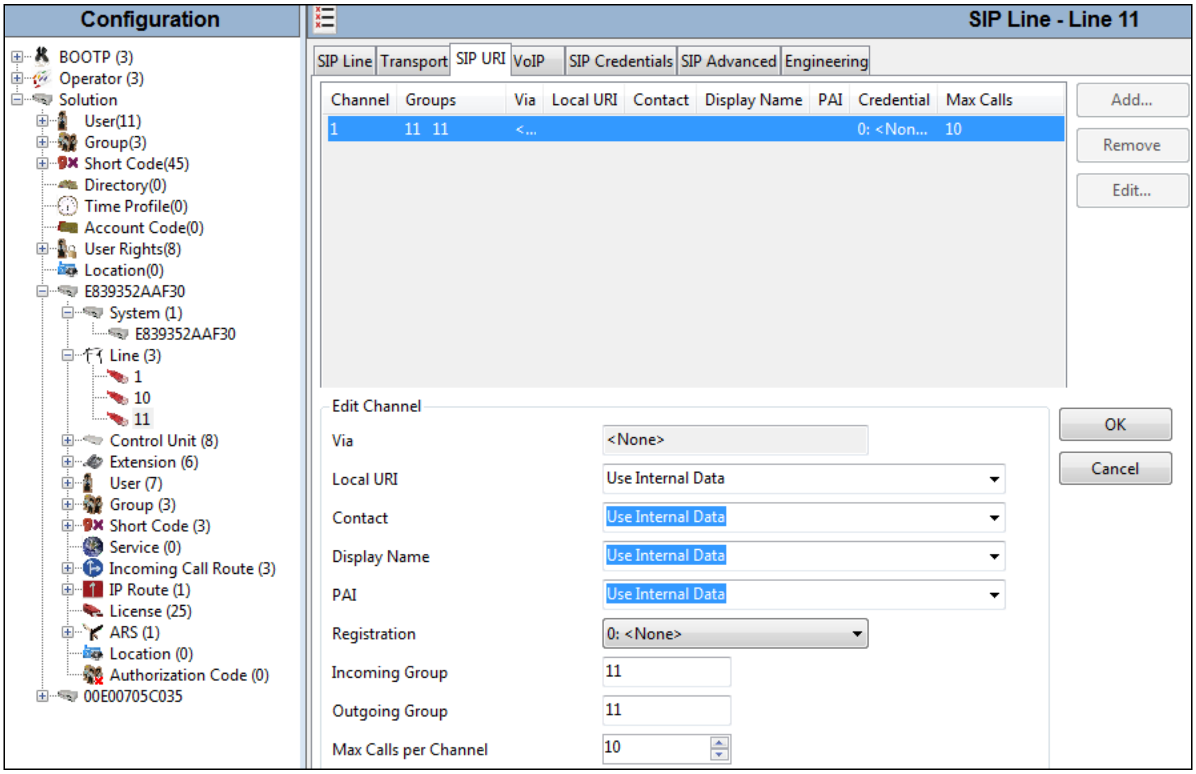
- Navigate to **Solution→**<*IP Office Name*>**→Line** from the **Configuration** menu.
- Right click and select **New→SIP Line** (not shown).
- Select the **SIP Line** tab.
- Enter a **Line Number** from a drop-down list.
- Enter the **ITSP Domain Name** which is set to the IP address of IP Office.
- Make sure **In Service** and **Check OOS** fields are checked.



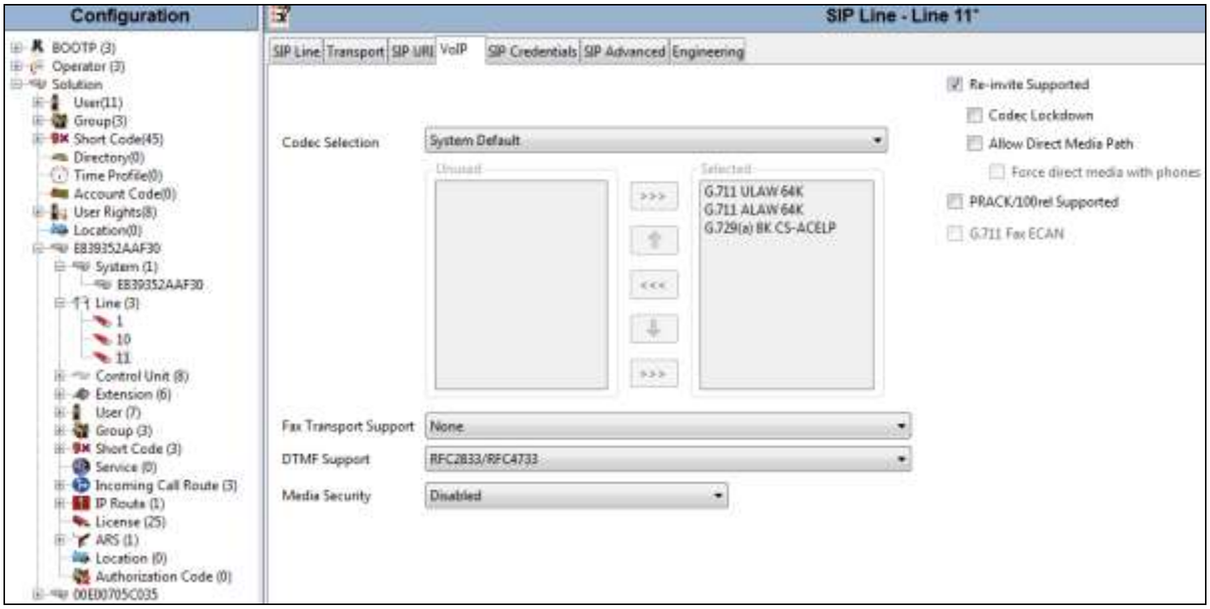
- Select the **Transport** tab.
- Enter the IVR server IP Address in **ITSP Proxy Address**.
- Set the **Layer 4 Protocol** to **UDP** as required by IVR.
- Set the **Send** and **Listen Port** to **5060** (default).



- Select the **SIP URI** tab.
- Click on **Edit…** button.
- Enter the **Incoming Group** and **Outgoing Group** that will be used to dial IVR.



- Select the **VOIP** tab.
- Make sure **Allow Direct Media Path** field is unchecked as IVR does not support shuffling as noted in **Section 2.2**.

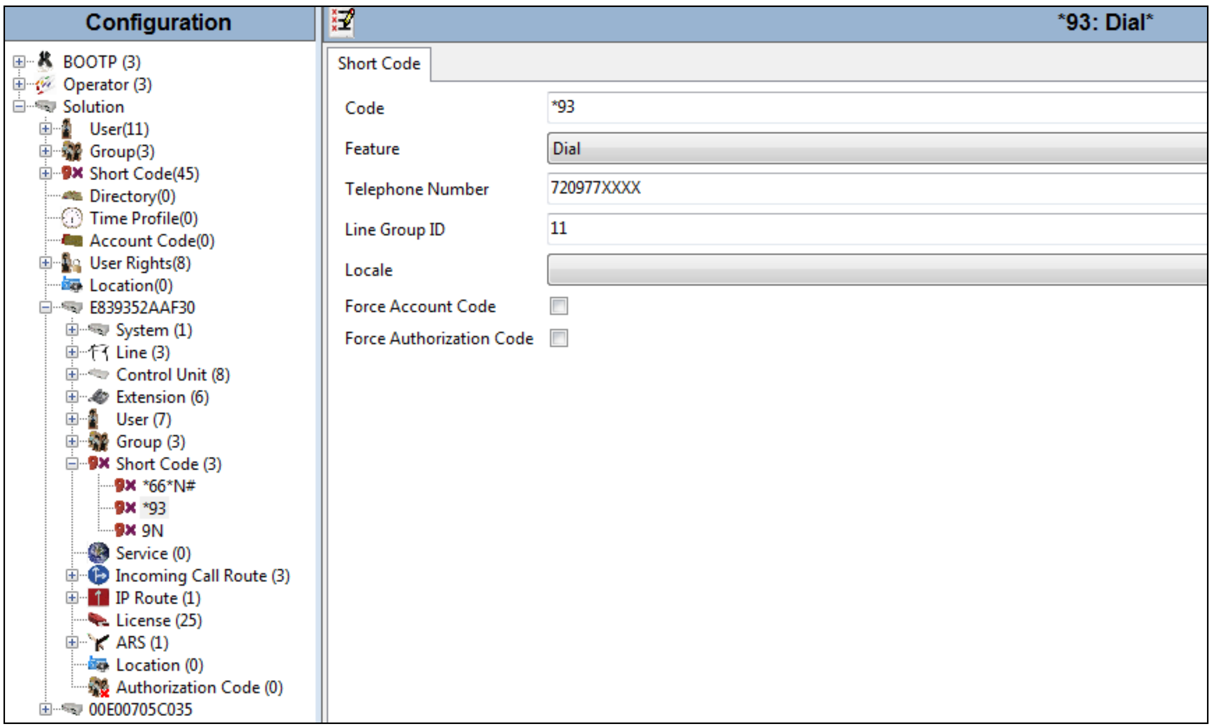


- Repeat the steps in this section to configure additional SIP trunks. In this reference configuration, two SIP trunks were created, one between IP Server and IVR and another one between IP Office IP500V2 Expansion and IVR.

4.4. Administer Short Code to Dial IVR

A number is allocated to dial the IVR sample application.

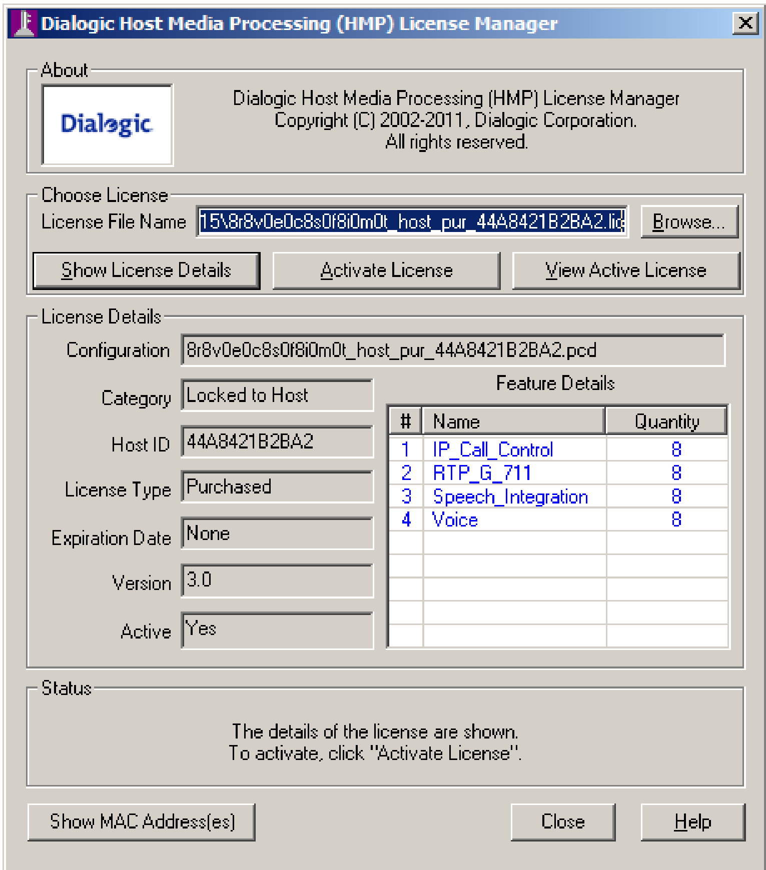
- Select **Solution**→<***IP Office Name***>→**Short Code** from the **Configuration** menu.
- Right click and select **New** (not shown).
- Enter the number to dial in **Code** field.
- Set the **Telephone Number** field to the DNIS recognized by the IVR sample application.
- Select **Dial** option for **Feature**.
- Select the **Line Group ID** of the SIP Line administered in **Section 4.3**.



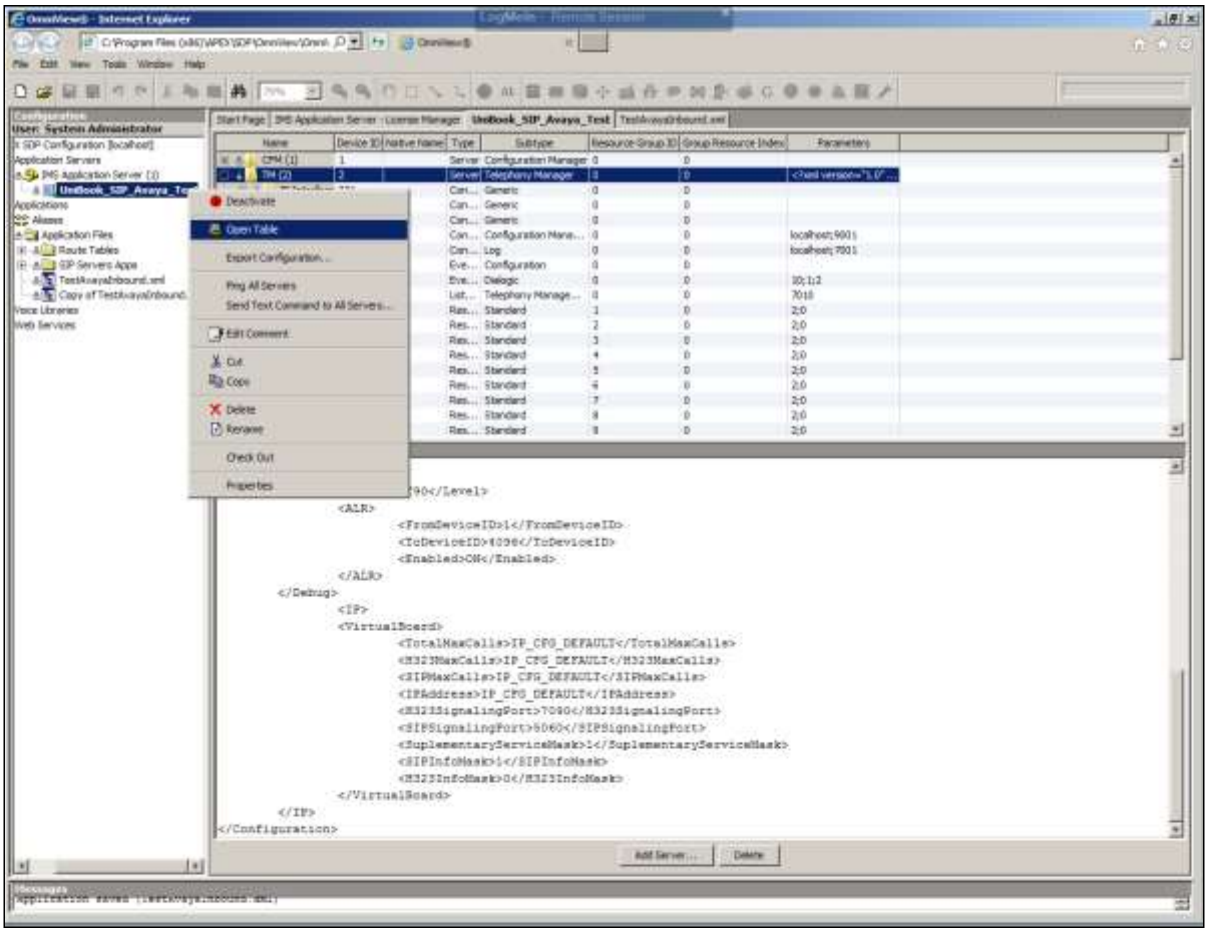
5. Configuration of Unified Dispatch Unibook platform

5.1. Dialogic HMP Setup

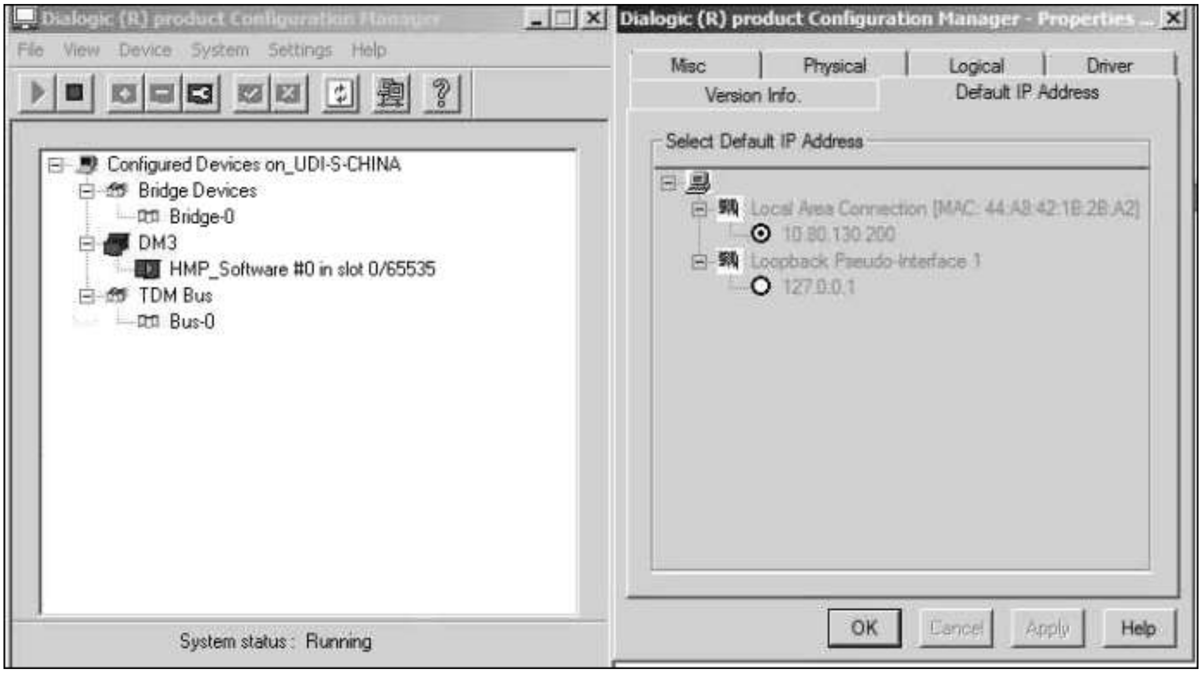
- Verify License SIP sessions for:
  - **IP_Call_Control** - SIP Call Control
  - **RTP_G_711** - RTP for G.711
  - **Voice** and **Speech_Integration** - Voice and Speech Integration (using Dialogic HMP Media Server)



- Verify Default Settings that Telephony Manager (SIP client) is setting up with Dialogic HMP. These settings are found in the parameters area of Telephony Manager device. In the IVR GUI, select the **Application Server** in the left panel, right click and from drop down menu select **Open Table**. The configuration table will open on the right panel, select **Telephony Manager** device line. In the parameters area verify that the IP Virtual Board section contains the following settings:
  - **SipInfoMask** – Set to **1**, which sets SIP parameters for transport to **UDP** and codec to **G.711**.
  - **SIPSignalingPort** - Set to **5060**.

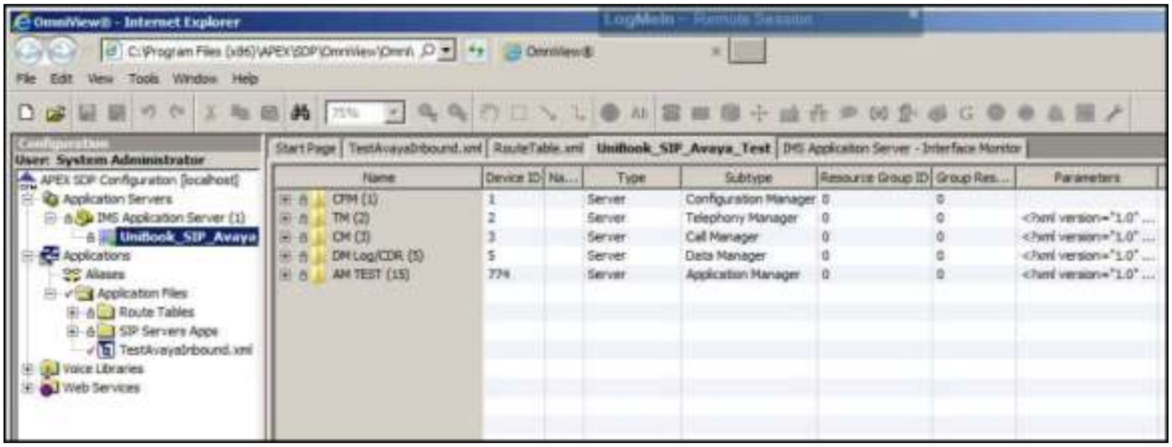


- Verify that **Server IP Address 10.80.130.200** is set as default IP address in Dialogic Configuration Manager to match the value used in **Section 4.3**. Note that in case the server has multiple network cards, the setup of ip address that is handling telephony traffic may be done here.

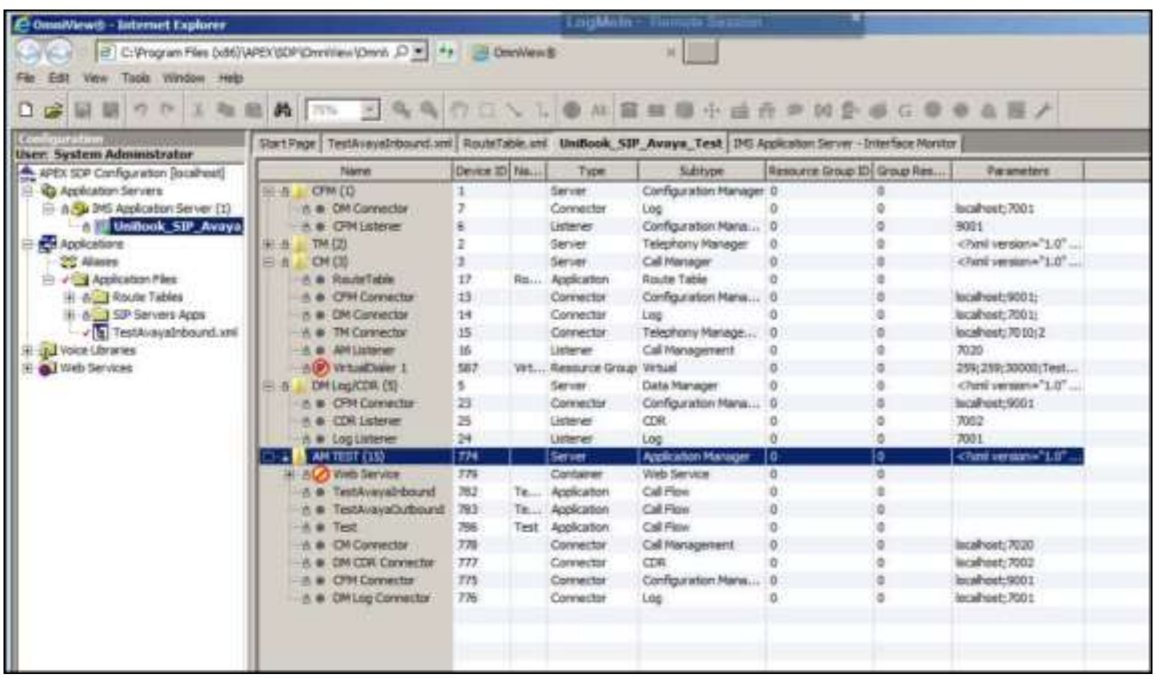


5.2. Unibook Setup

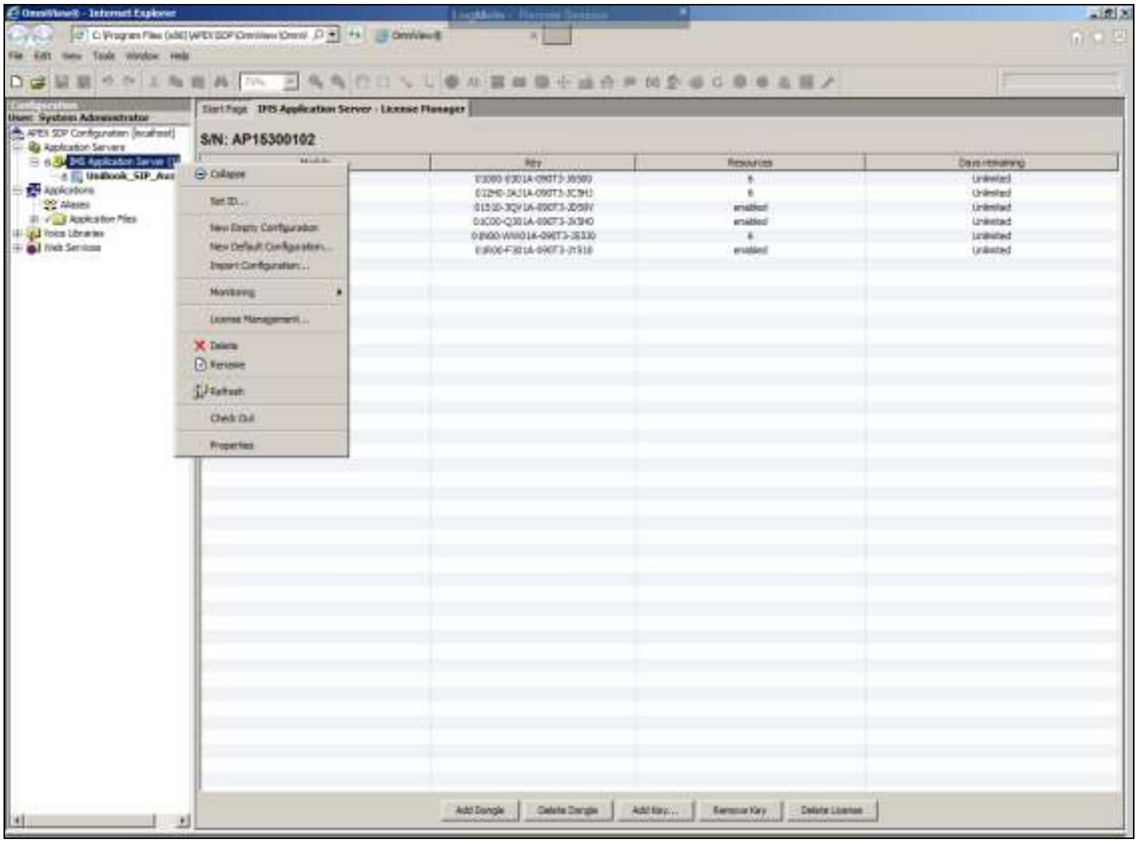
- Use any browser to access OmniView GUI, import a default configuration and configure the Unibook Managers needed for the sample application:



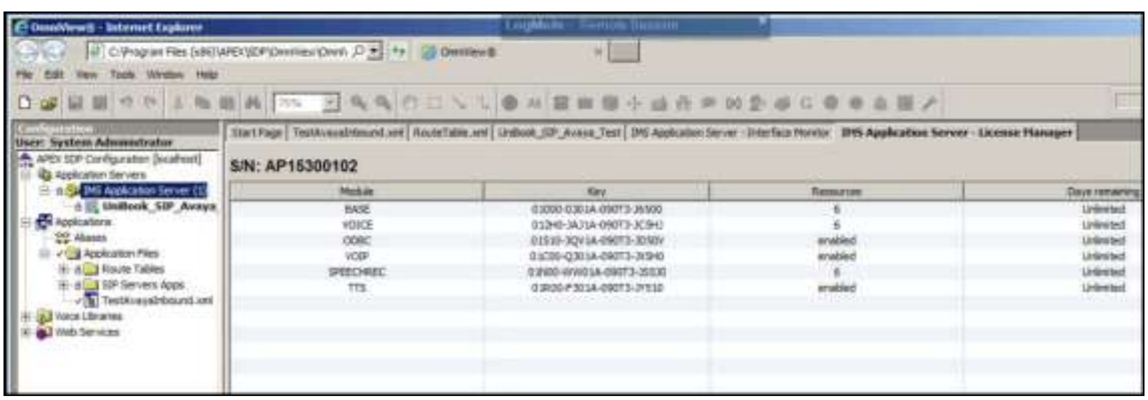
- The following screen shows all the services configured along with the ports used.
  - CFM Configuration Manager using port **9001** on localhost.
  - DM Data Manager using ports **7001**, **7002** on localhost.
  - AM TEST (15) Application Manager using port **7020** on localhost.
  - CM Call Manager ports **7020**, **7010** on localhost.
  - TM Telephony Manager using port **7020** on localhost.



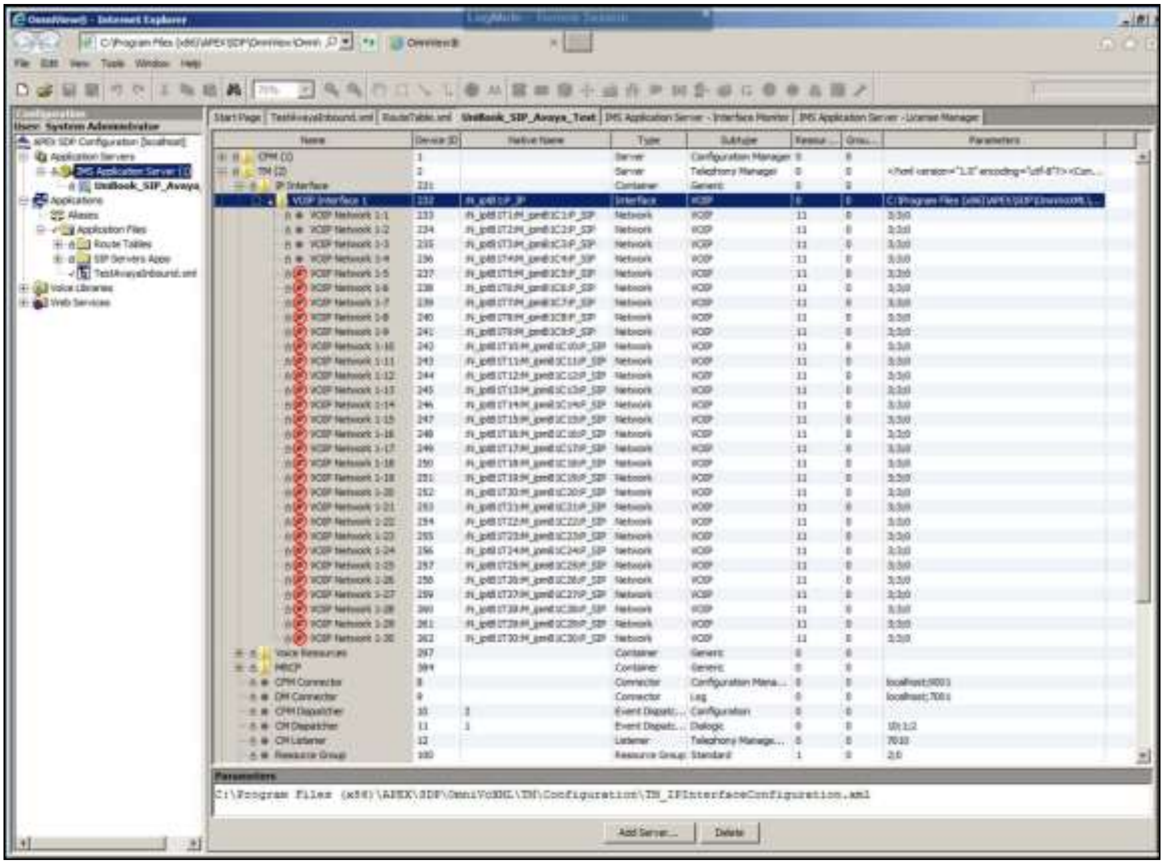
- Navigate to **Application Servers→IMS Application** Server and select **License Management** from drop down list to verify that proper licenses have been applied.



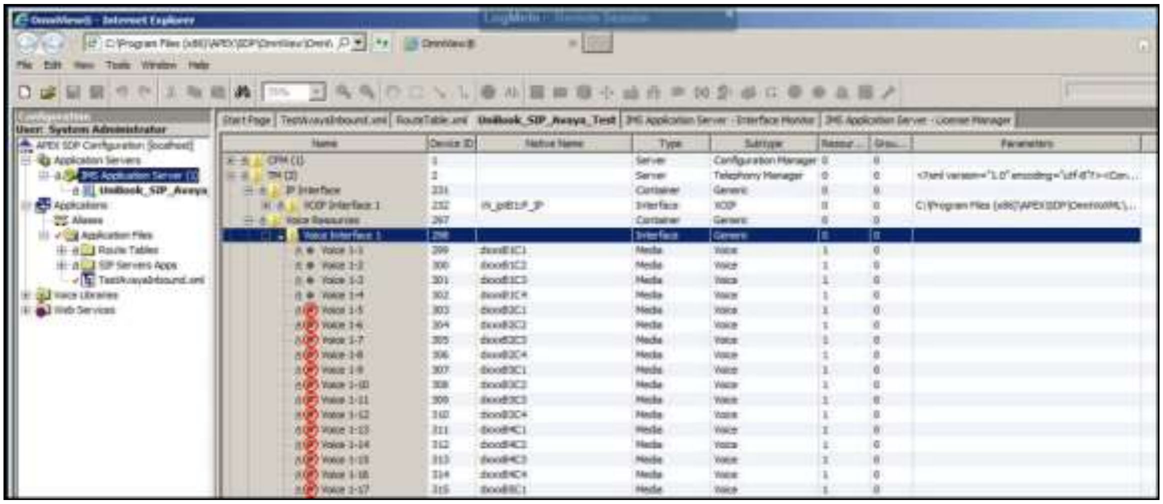
- Select **Add Dongle** followed by **Add Key** (not shown) for adding all needed license keys. The screen below shows **BASE**, **VOICE** and **VOIP** keys required for telephony implementation to work.



- Configure Telephony Manager SIP network devices by setting up the IP interface for SIP as shown below:

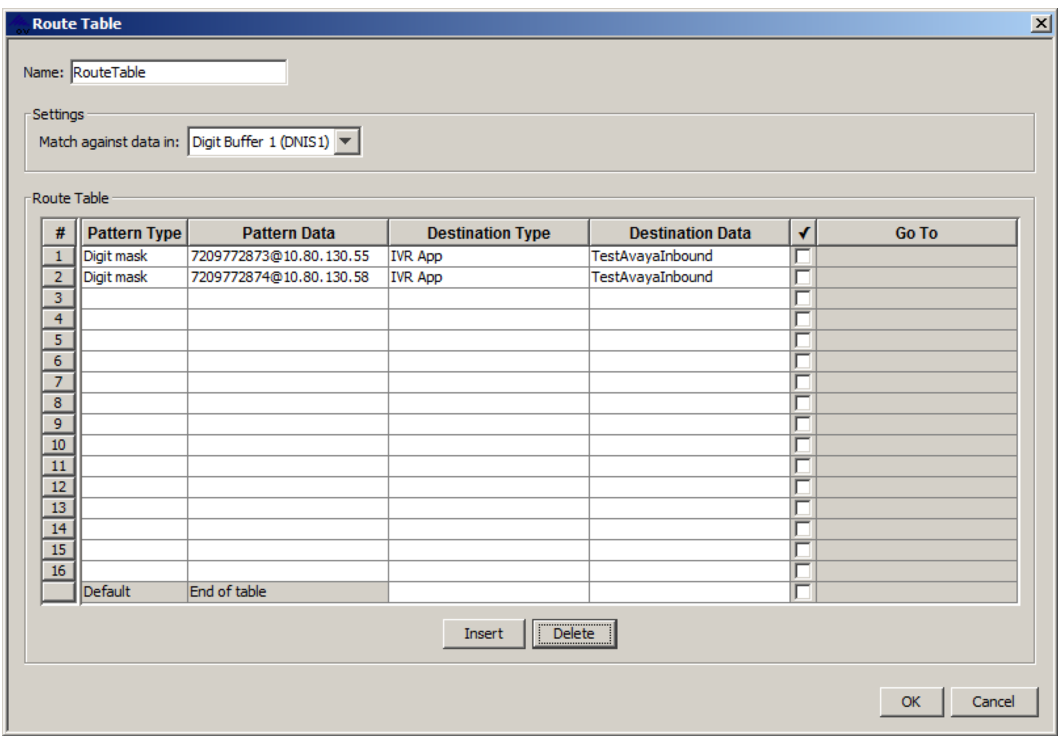


- Configure Telephony Manager SIP voice/media devices by setting up the IP interface for SIP as below:



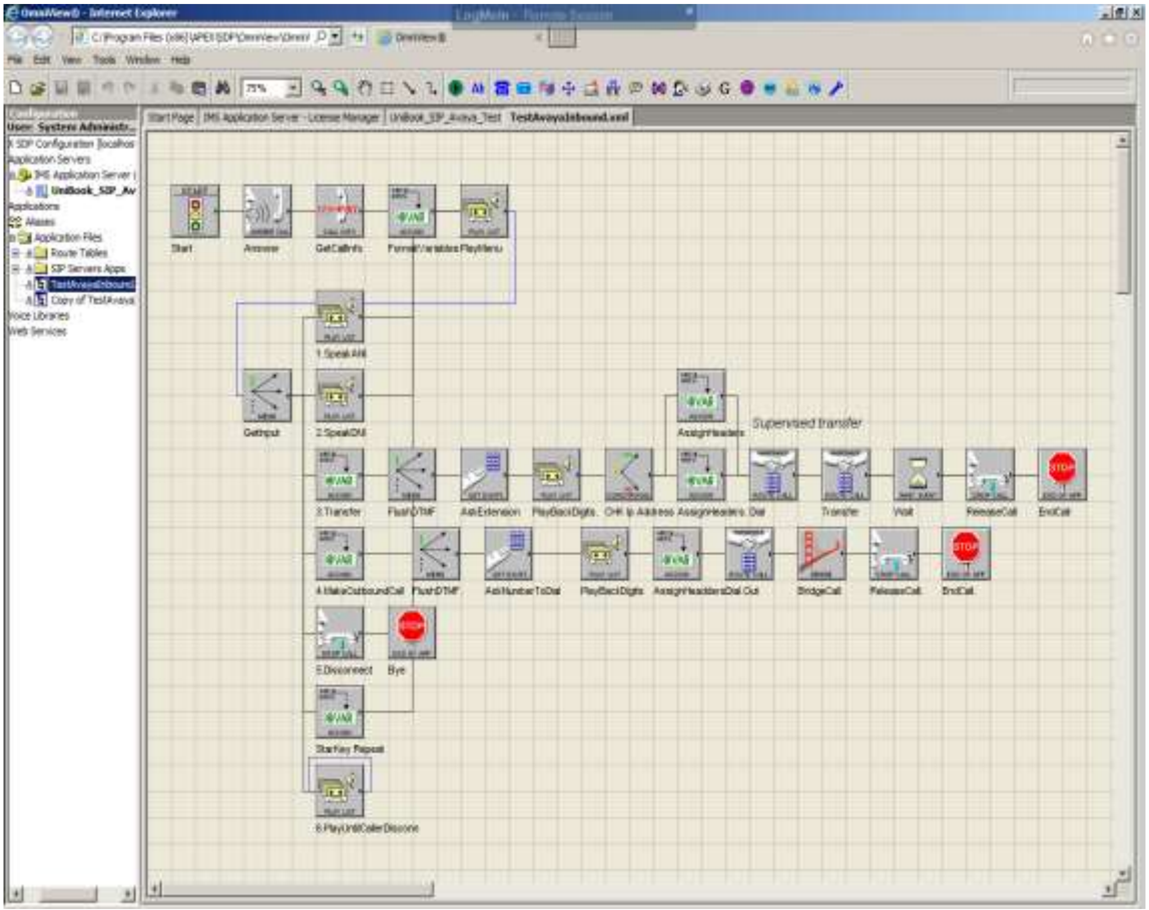
5.3. SIP Trunk Configuration

In this case, because all network elements are within the same network, IP Office did not require the SIP Client on IVR to register and authenticate for inbound calls. Avaya IP Office routes the calls to IVR based on DNIS as shown below in the routing table to the **TestAvayaInbound** sample application.



5.4. Develop and Deploy IVR Callflow

The following screenshot shows the sample IVR callflow developed on Unibook platform to test SIP trunking solution with IP Office.



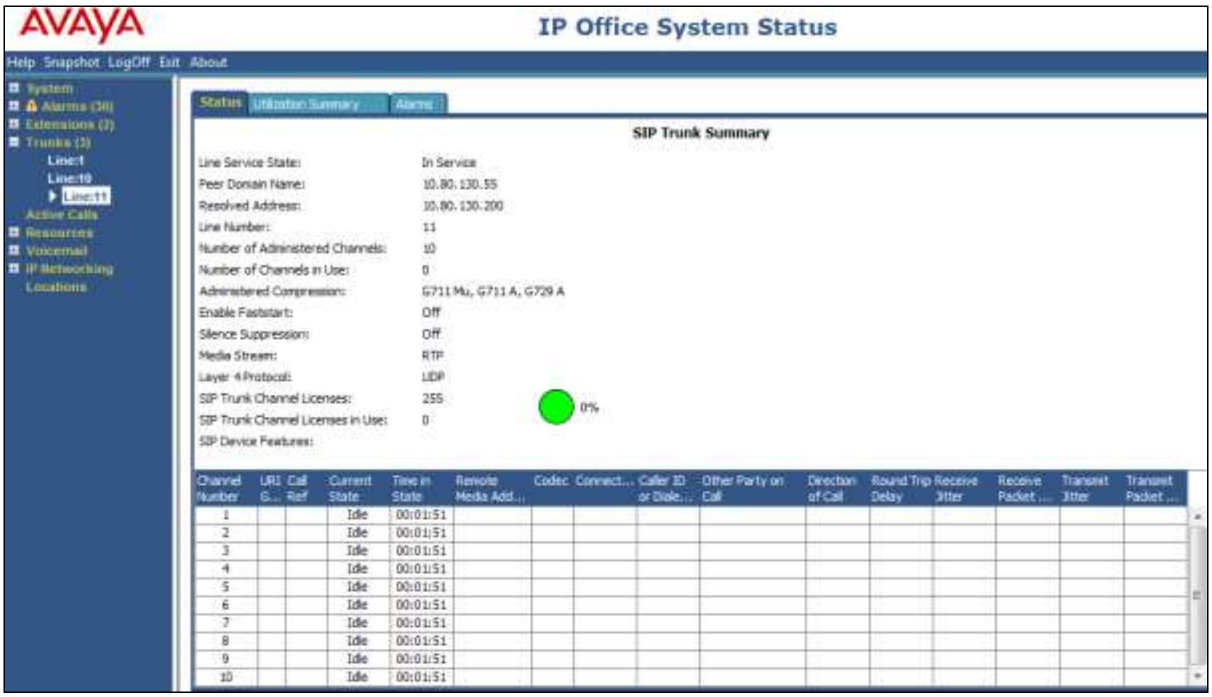
6. Verification Steps

To verify a successful configuration of IVR and IP Office, a call is placed from an IP Office telephone to IVR with the caller hearing the prompts from IVR.

6.1. SIP Trunk to IVR

Log in to the IP Office System Status application. From the left hand menu navigate **Trunks→**<**SIP Line configured**>.

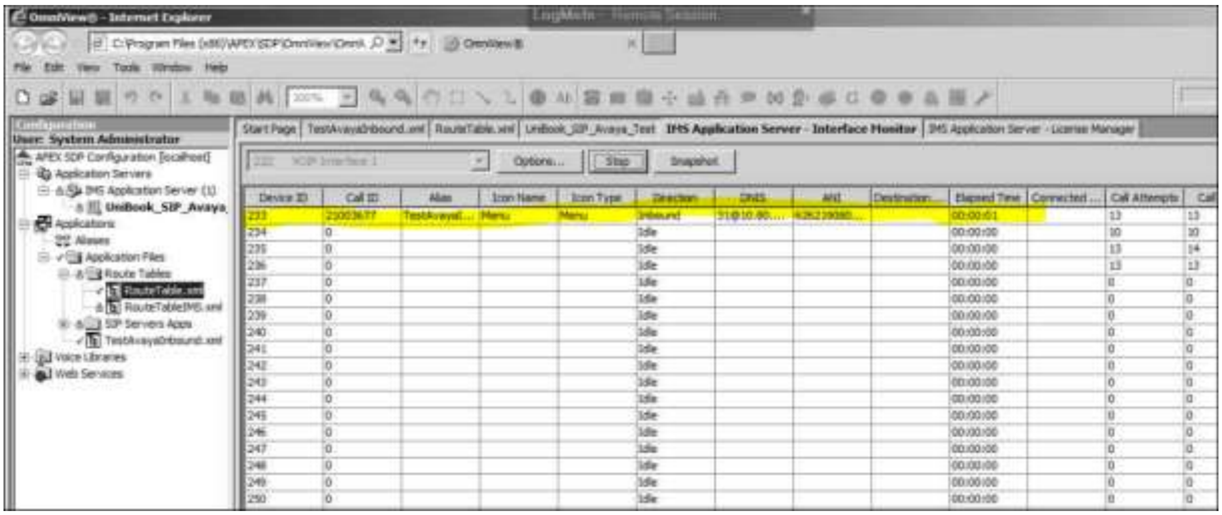
- Verify **Line Service State:** is **In Service**.
- Verify **Channel Current State** is **Idle** (**Active** if a call is in progress).



6.2. Verify Unified Dispatch IVR Application

Verify the inbound calls are received and handled by checking the monitoring tool as shown below:

- IVR prompts are played and follow on the correct selections.
- DTMF input is received and accepted.
- Calls are transfered to an agent queue if requested.
- Upon disconnect, the IVR resource is freed and ready to accept new call request.



7. Conclusion

These Application Notes describe the configuration steps required for Unified Dispatch Unibook IVR platform to successfully interoperate with Avaya IP Office Server with IP500V2 Expansion. All feature functionality and serviceability test cases were completed successfully with exceptions noted in **Section 2.2**.

8. Additional References

This section references the Avaya and Unified Dispatch documentation relevant to these Application Notes.

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

[1] *Deploying IP Office Server Edition Solution, October 2015*

[2] *Administering Avaya IP Office Platform with Manager, November 2015*

[3] *Using IP Office Platform System Status, August 2015*

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.