Avaya Solution & Interoperability Test Lab

Application Notes for configuring Unified Dispatch Unibook platform as SIP Endpoint 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 board to successfully interoperate with Avaya IP Office Server Edition 9.1 and IP500V2 Expansion. Unified Dispatch Unibook platform is an IVR application that is registered to IP Office as a SIP endpoint.

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 sample application to successfully interoperate with Avaya IP Office 9.1 Server Edition and IP500V2 Expansion (IP Office). Unified Dispatch Unibook platform is set to register as a SIP Endpoint (IVR Endpoint) with IP Office.

Unified Dispatch UniBook platform integrates with PBX systems and ground transportation scheduling and dispatch systems 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 call to IVR Endpoint. **Note: IVR Endpoint registered as a SIP Endpoint to both IP Office Server and IP Office IP500V2 Expansion**. Once the IVR Endpoint receives a call, it initiates the **sample application** which has a variety of self-serving prompts including transferring the call to another agent/extension registered to IP Office.

The main objectives were to verify the following:

- Inbound call from an IP Office extension to the IVR Endpoint
- Inbound call from PSTN to IVR Endpoint using PRI trunk on IP Office IP500V2 Expansion
- Inbound call from PSTN to IVR Endpoint using SIP trunk on IP Office Server
- Calls can be transferred to another extension/agent from the IVR Endpoint
- IVR Endpoint can recognize DTMF tones using RFC2833
- IVR Endpoint 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 Endpoint using **sample application** can respond to DTMF tones. The scope of testing included the navigation of the paths provided by the IVR Endpoint **sample application** using DTMF including transfer to an agent/extension on IP Office.

The serviceability testing focused on verifying the ability of IVR Endpoint 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 the following results/observations:

- IVR Endpoint does not support shuffling/direct media
- IVR Endpoint only supports G711MU codec
- IVR Endpoint needs to be configured as UDP for transport
- Outbound calls from IVR Endpoint are supported but were not tested
- IVR Endpoint cannot hold the call but calls held by the calling party had no adverse effect
- Since IVR Endpoint 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
- IVR Endpoint was configured to support Blind Refer for transferring calls

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, Unified Dispatch Unibook platform IVR applications register as SIP Endpoints with Avaya IP office.

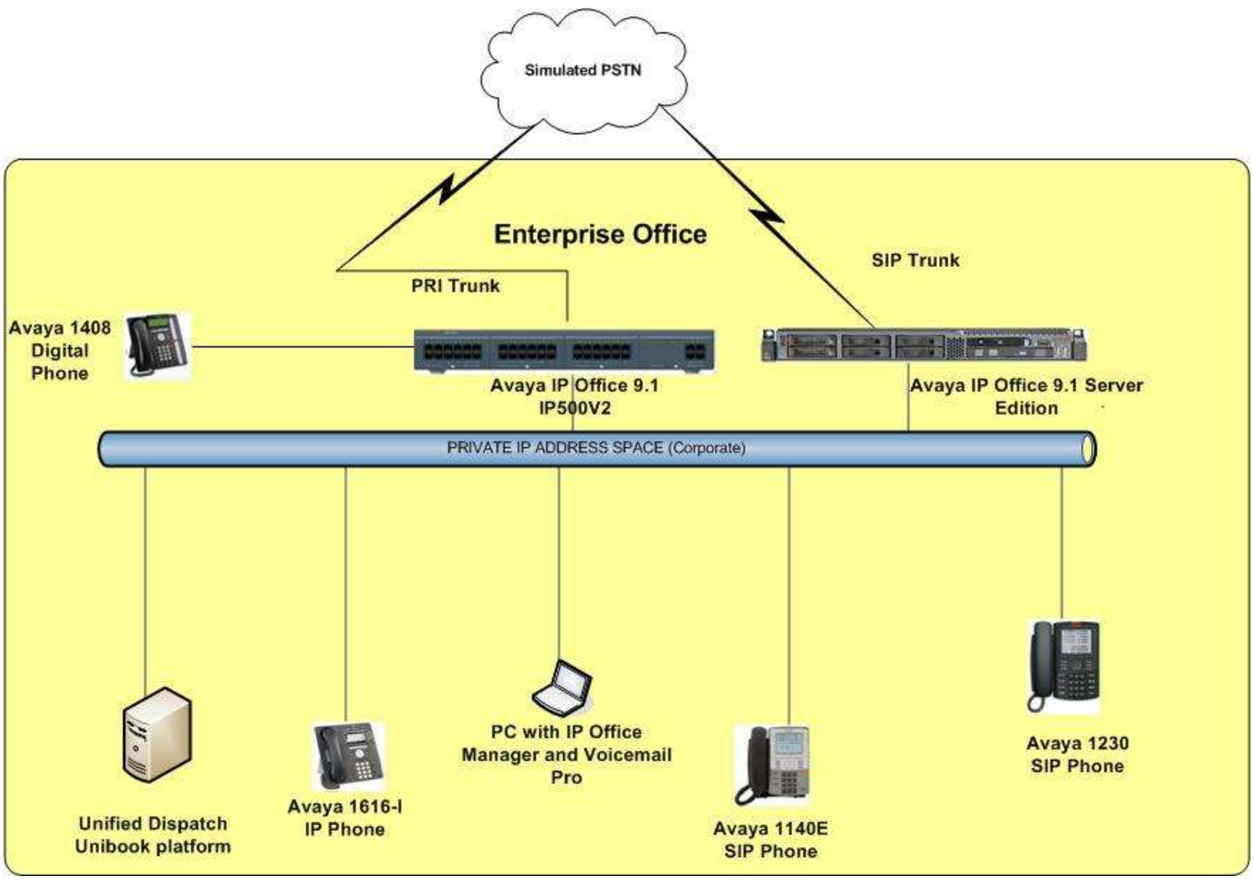


Figure 1: Avaya IP Office Server/IP500V2 Expansion and Unified Dispatch Unibook platform

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 IP500V2 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 IP500V2 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:

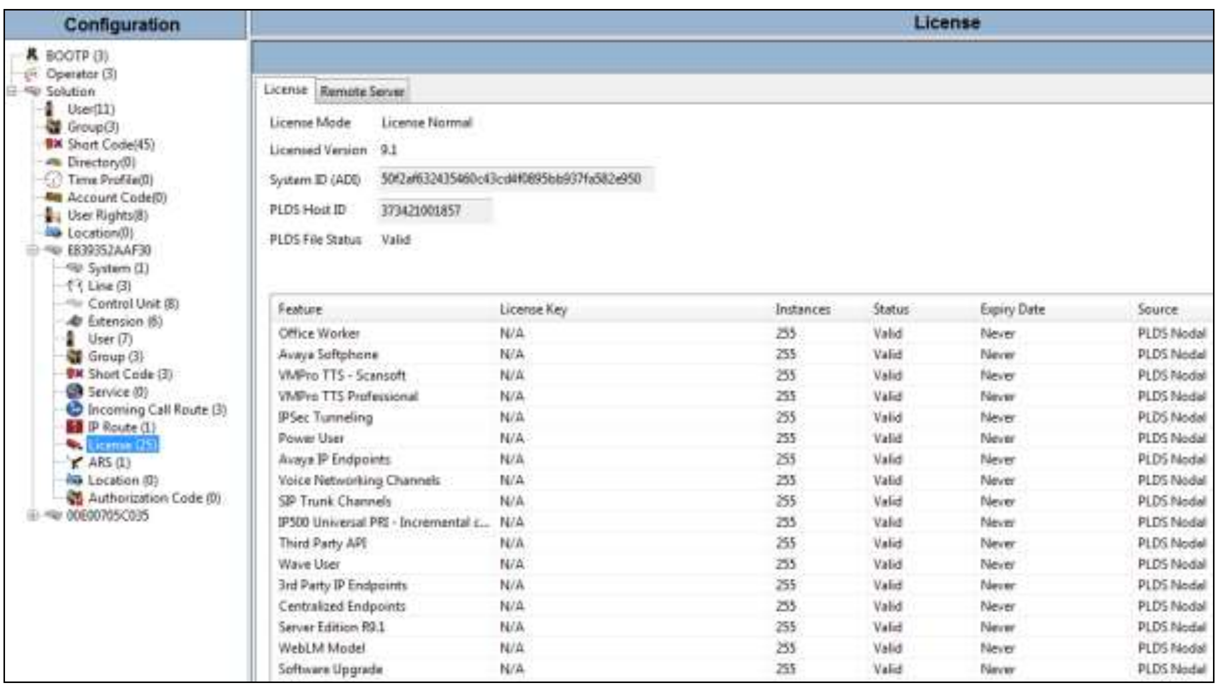
- Verify Third Party IP Endpoint licenses
- Verify SIP Sessions
- Administer SIP Endpoint

Note: For PSTN calls, a PRI trunk was configured to IP Office IP500V2 Expansion and a SIP Trunk was configured to IP Office Server. IP Office Server does not support PRI Trunks.

4.1. Verify Third Party IP Endpoint Licenses

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.

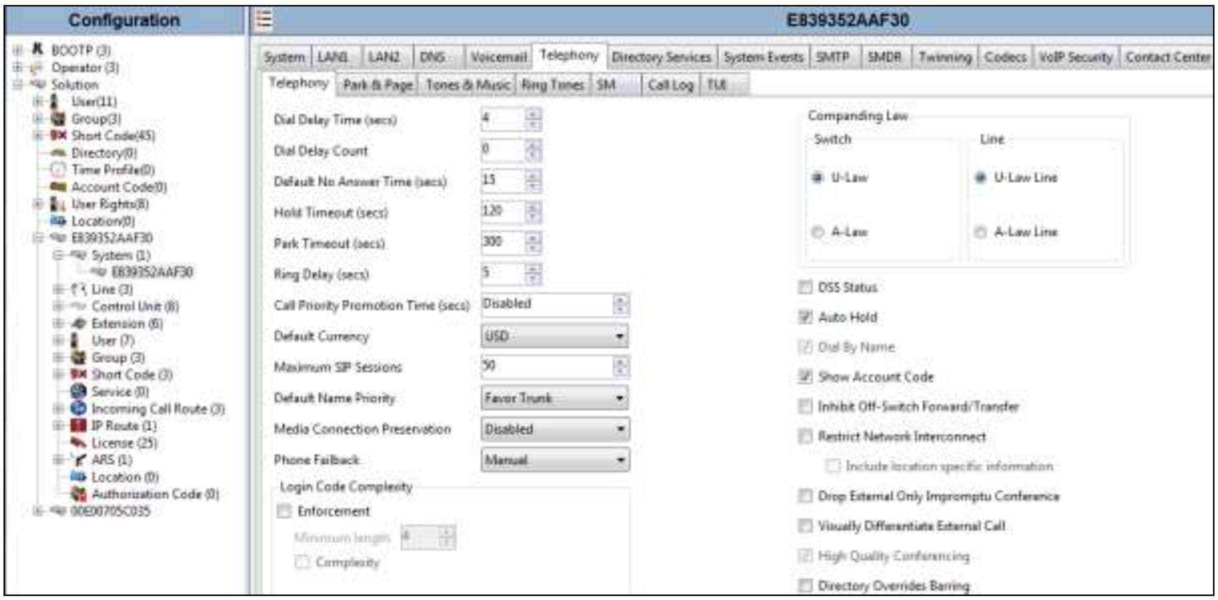
- Navigate **Solution**→<***IP Office Name***>→**License** from the **Configuration** menu
- Verify that **3rd Party IP Endpoints** has sufficient licenses available



4.2. Verify SIP Sessions

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

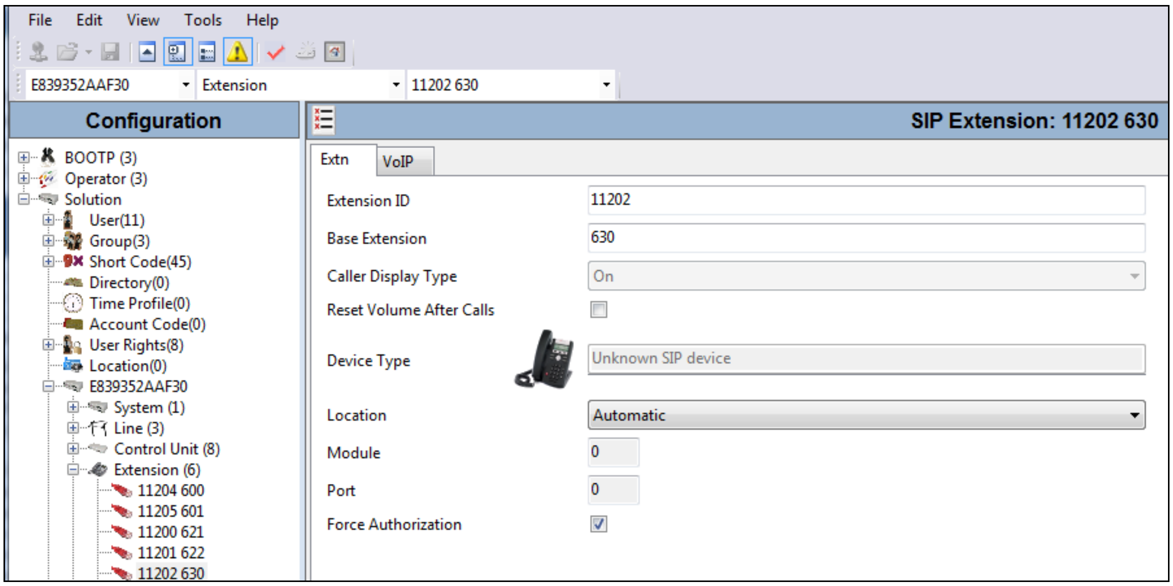
- Select **Solution**→<***IP Office Name***>→**System** from the **Configuration** menu
- Select the **Telephony** tab
- Verify the **Maximum SIP Sessions** value. Note, if there are not enough sessions configured, the calls from PSTN will fail.



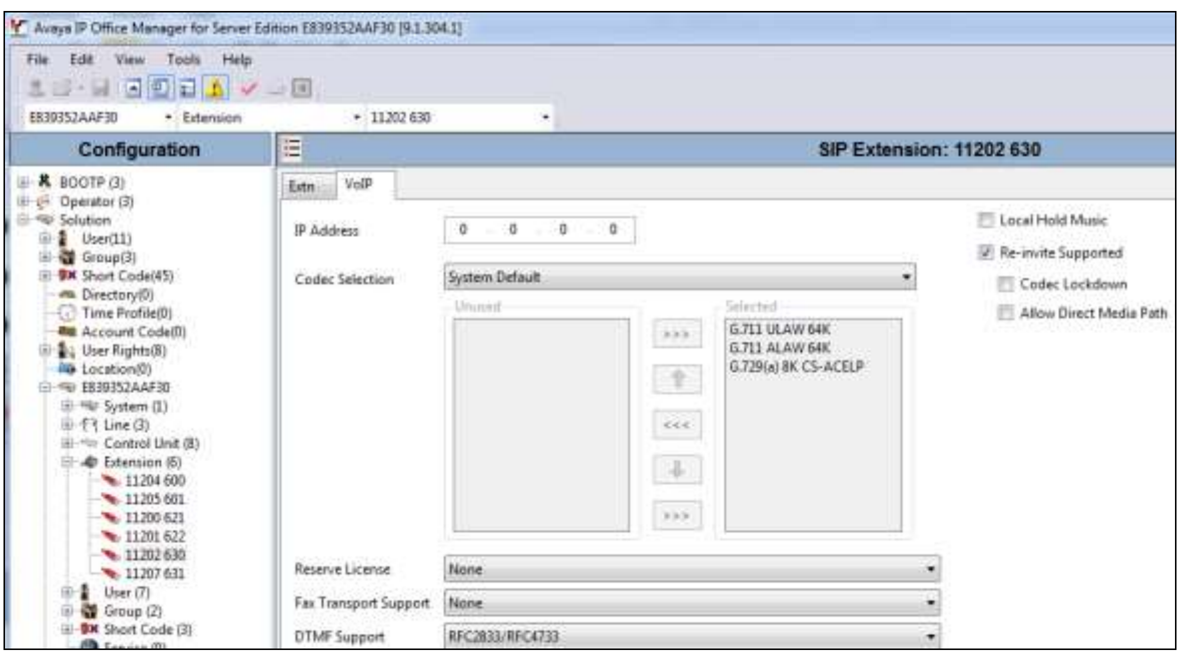
4.3. Administer SIP Endpoint

A SIP Endpoint was created for IVR to register with IP Office.

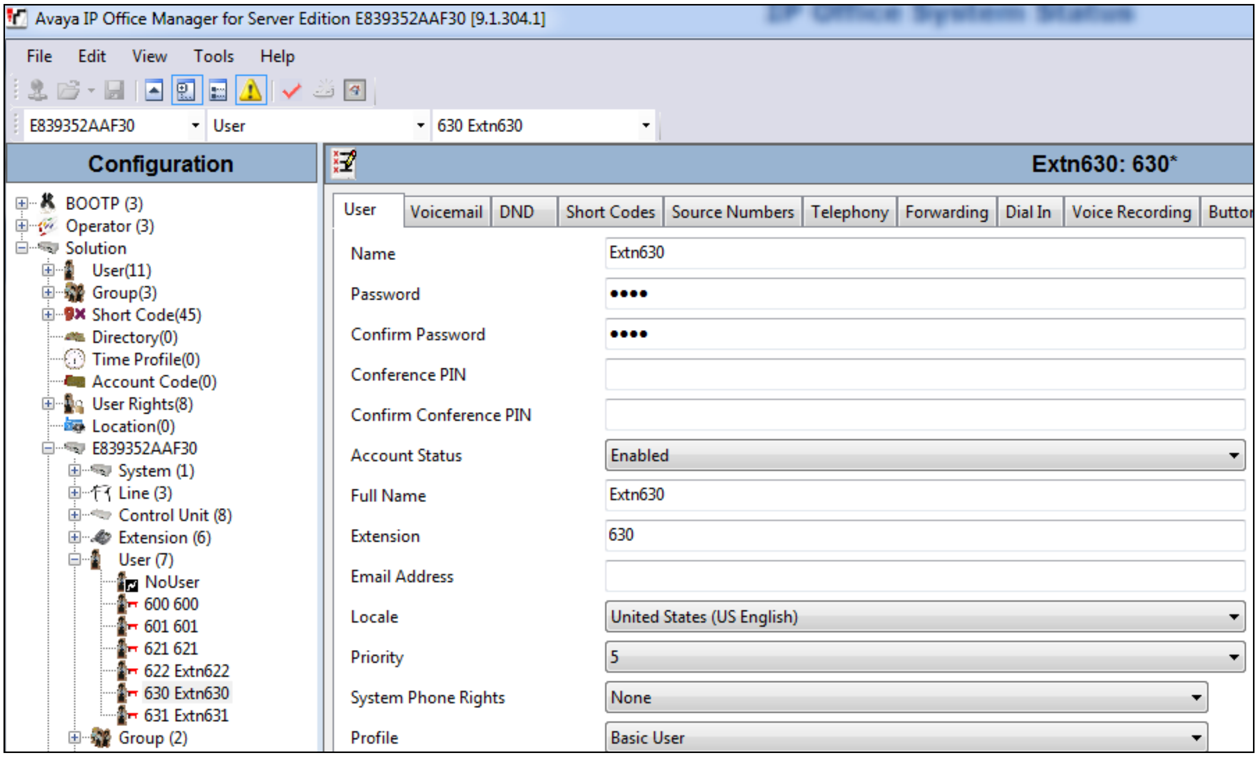
- Navigate to **Solution→**<***IP Office Name***>**→Extension** from the **Configuration** menu
- Right click and select **New→SIP Extension** (not shown)
- Select the **Extn** tab
- Enter an unused **Extension ID** and **Base Extension**



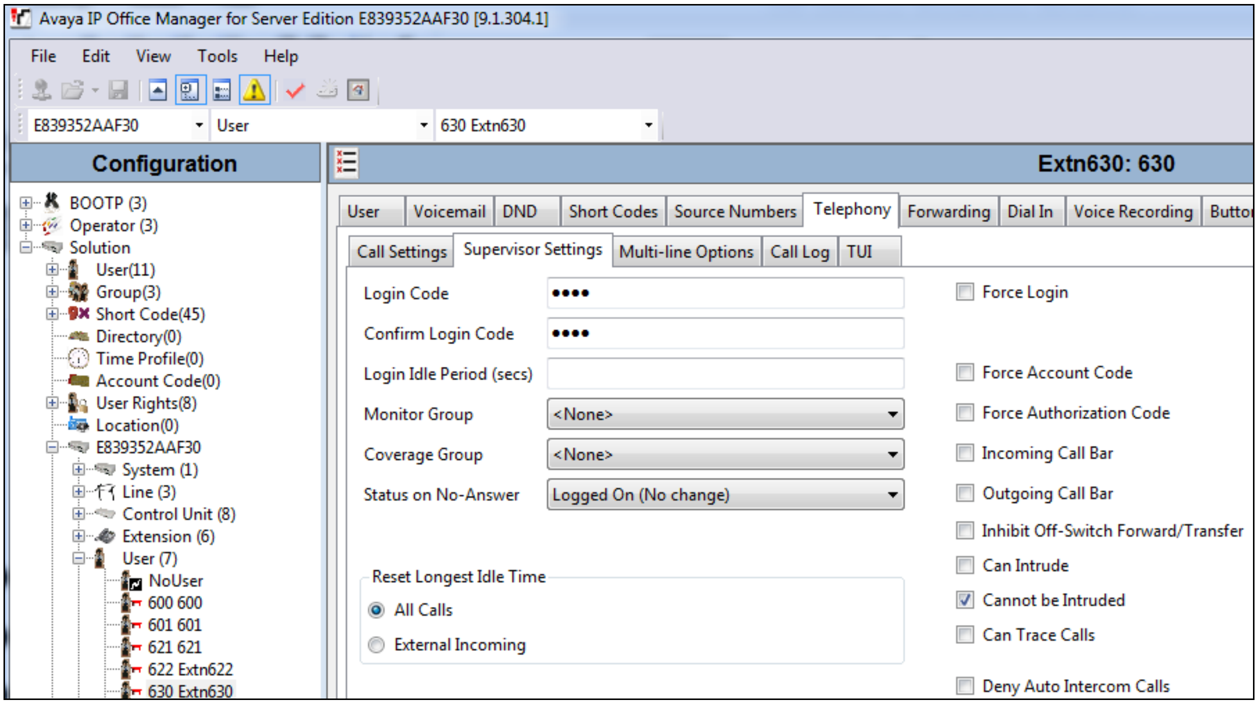
- Select the **VoIP** tab and uncheck the **Allow Direct Media Path**. Also, note that **DTMF Support** field can be modified for in band or out of band DTMF support.



- Navigate to **Solution→**<***IP Office Name***>**→User** from the **Configuration** menu
- Right click and select **New** (not shown)
- Select the **User** tab
- Enter **Name**, **Password**, **Confirm Password**, **Full Name** and **Extension** fields. Note that the **Extension** field matches the **Extension ID** created in the beginning of this section.



- Navigate to the **Telephony→ Supervisor Settings**
- Enter the **Login Code** and **Confirm Login Code** fields. This is the SIP endpoints password to register with IP Office.

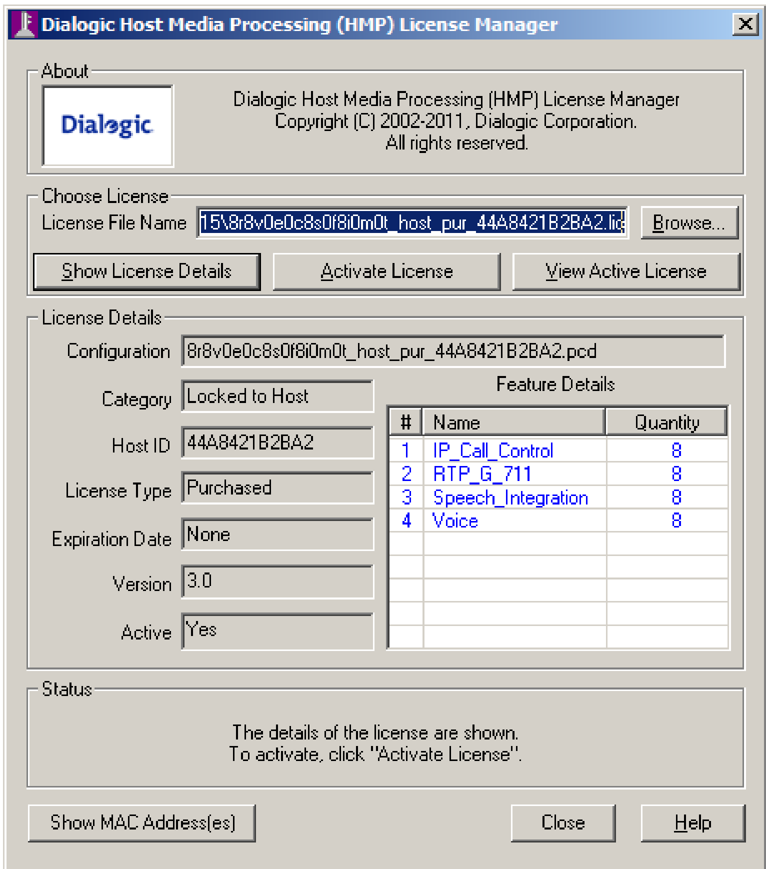


- Repeat the steps in this section to configure additional SIP endpoints. In this reference configuration, four SIP endpoints were created for IVR, two each for registration with IP Server and IP Office 500V2 Expansion.

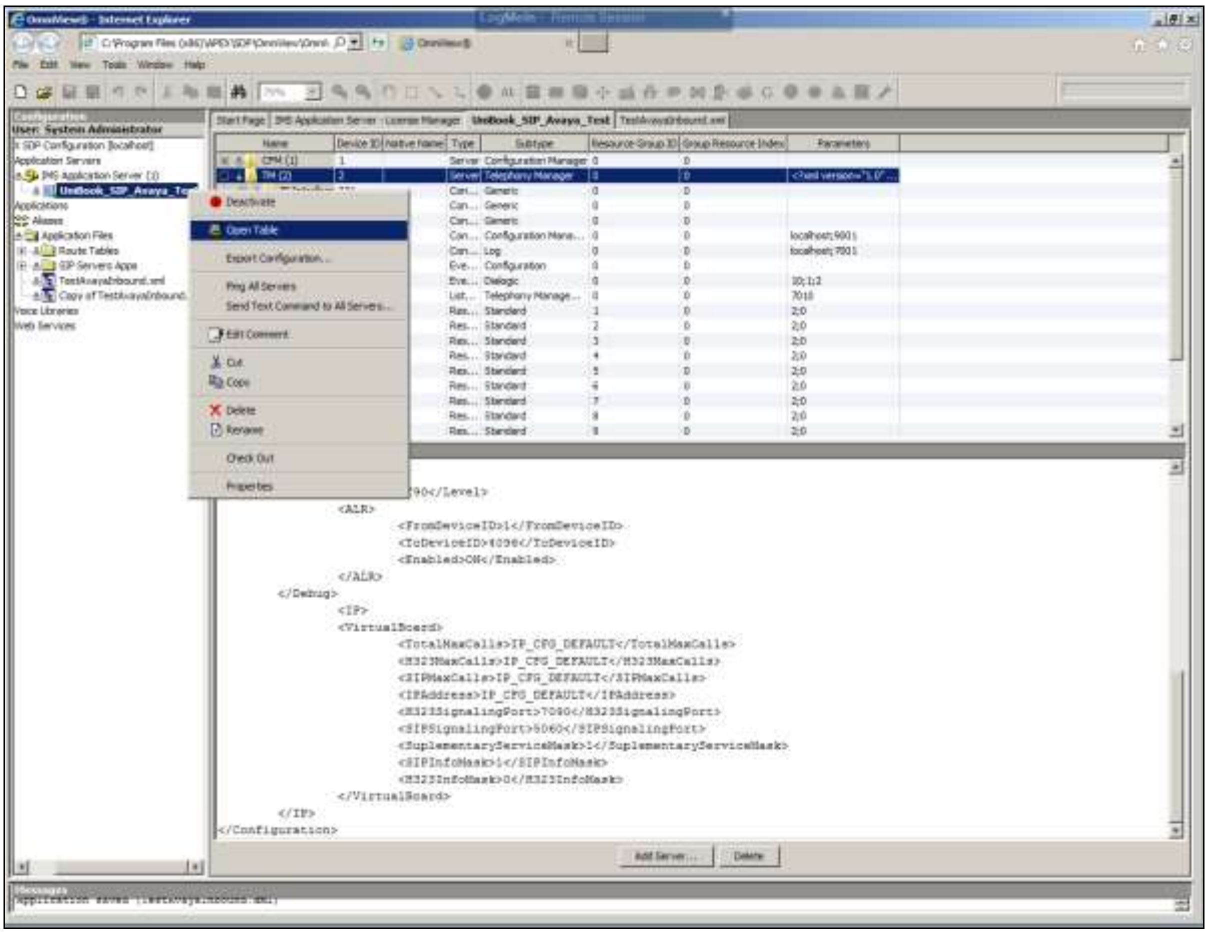
5. Configuration of Unified Dispatch Unibook platform

5.1. Dialogic HMP Setup

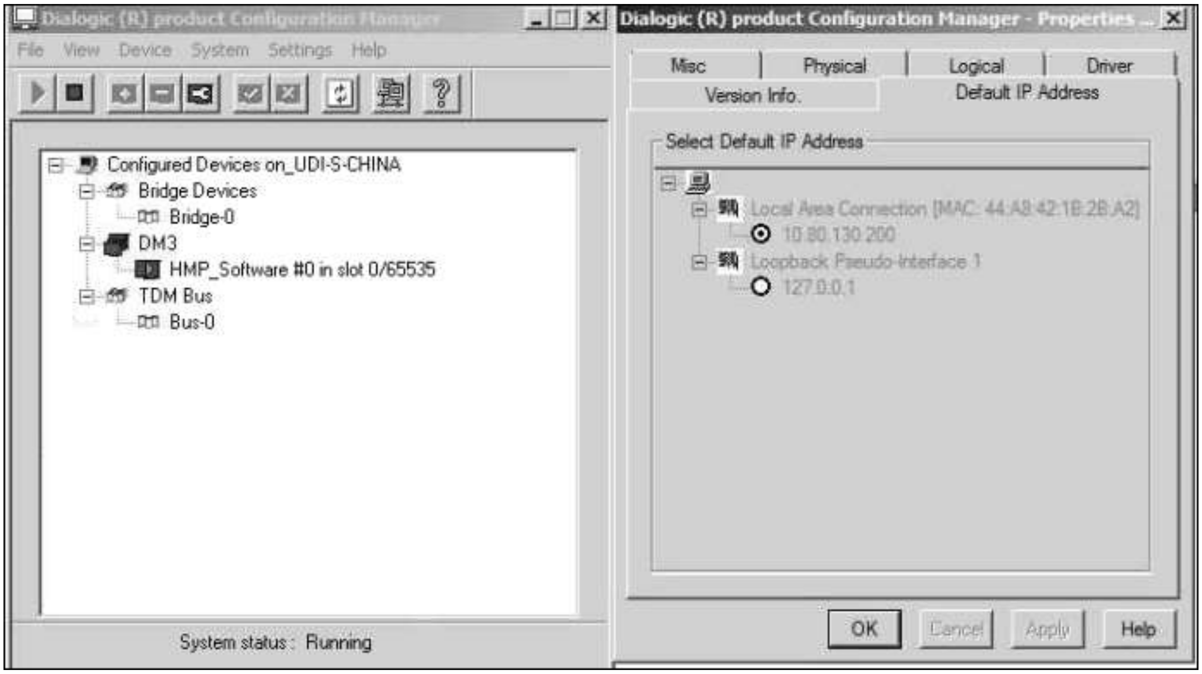
- Verify License SIP sessions for:
  - **IP_Call_Control** - SIP Call Control
  - **RTP_G_711** - RTP for G.711
  - **Voice** and **Speech_Integration** - Voice and Speech Integration (using Dialogic HMP Media Server)



- Verify default settings that Telephony Manager (SIP client) is setting up with Dialogic HMP. These settings are found in the parameters area of the 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 the **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**

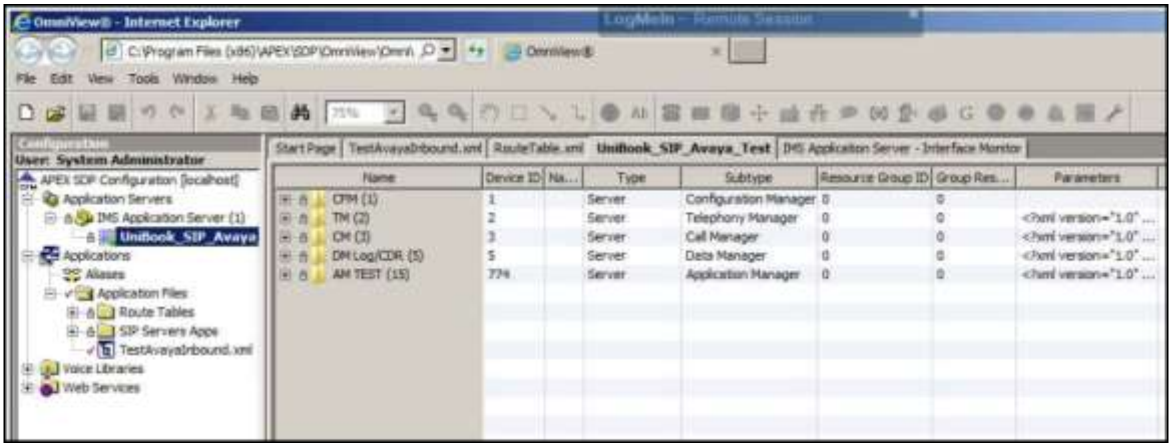


- Verify that **Server IP Address 10.80.130.200** is set as the default IP address in Dialogic Configuration ManagerNote that in case the server has multiple network cards, the setup of IP address that is handling telephony traffic may be done here.

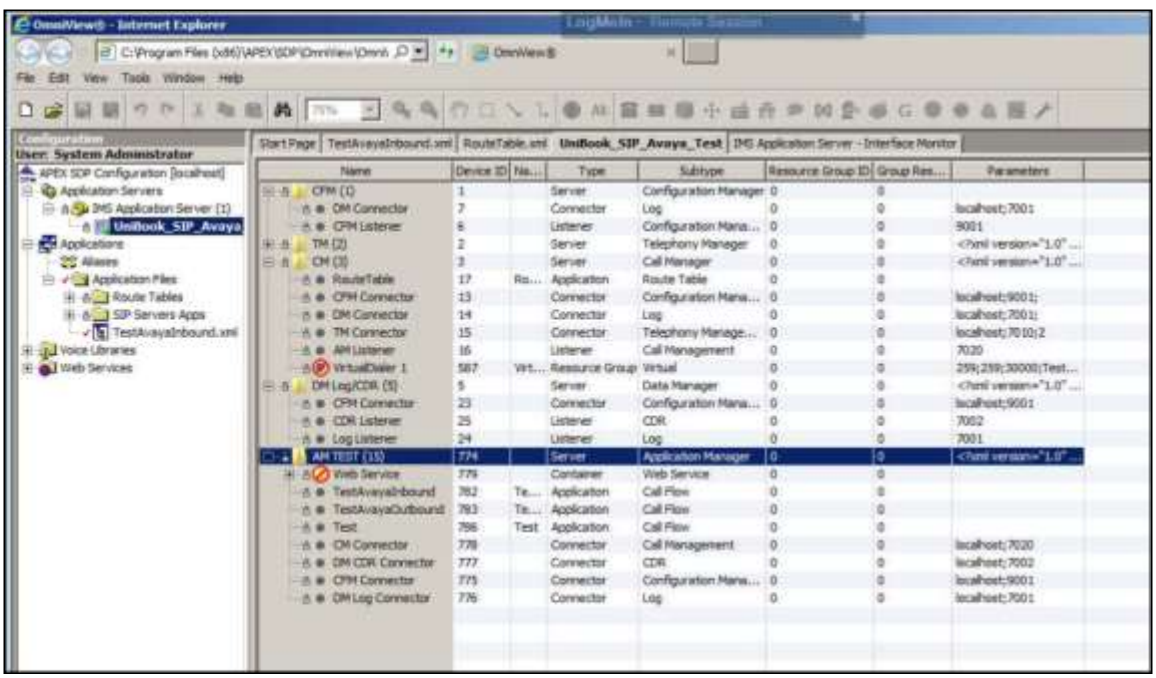


5.2. Unibook Setup

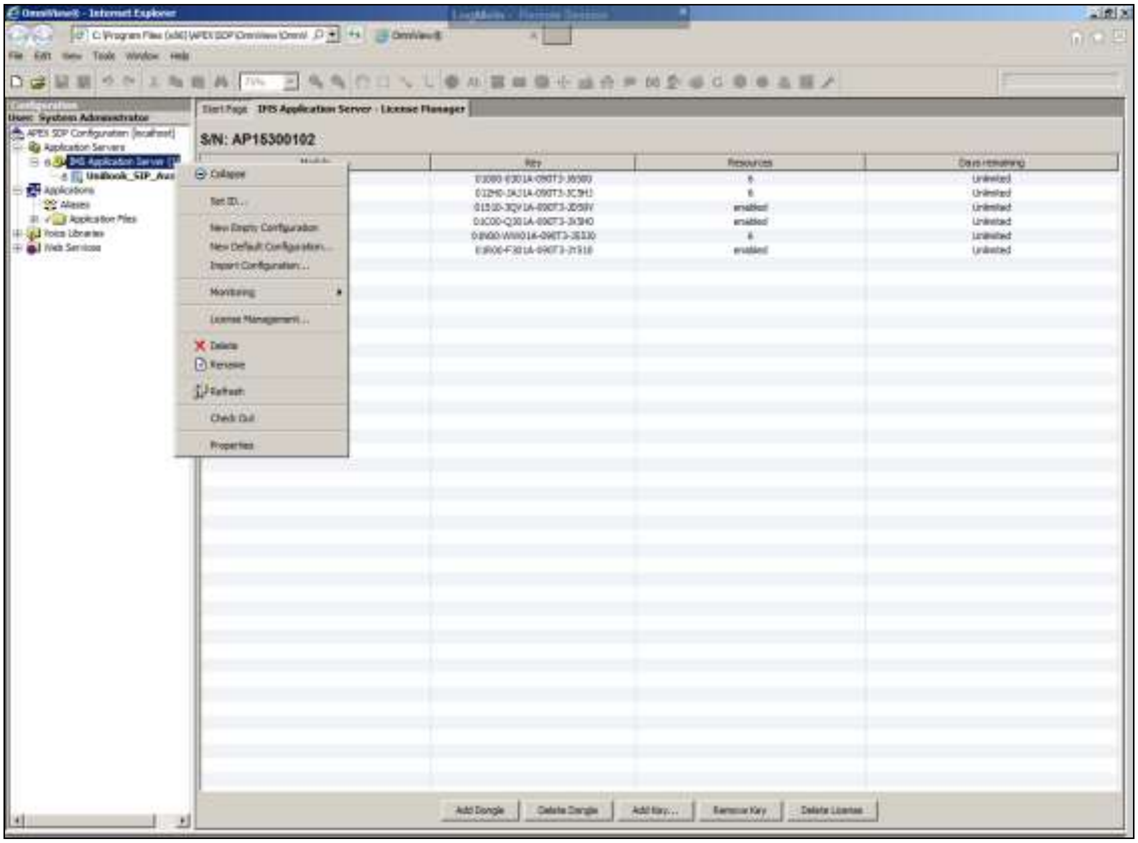
- Use any browser to access OmniView GUI, import a default configuration and configure the Unibook Managers needed for the sample application:



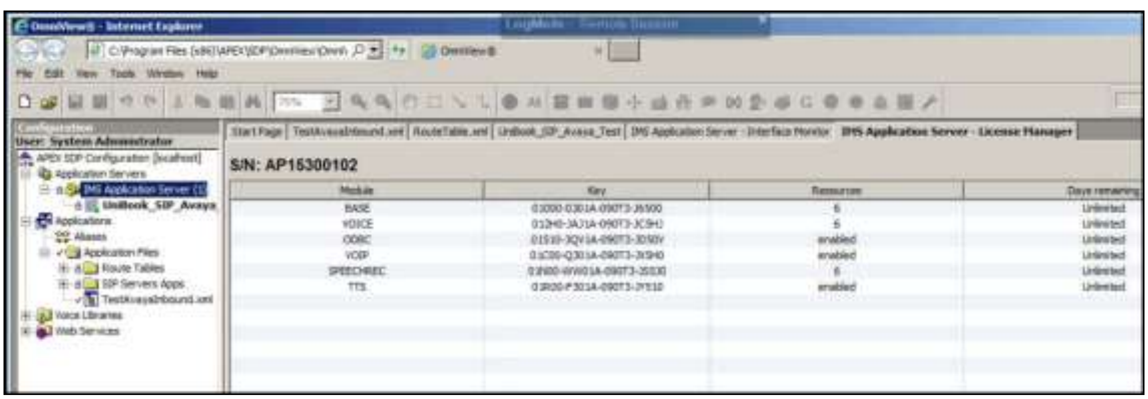
- The following screen shows all the services configured along with the ports used.
  - CFM Configuration Manager using port 9001 on localhost
  - DM Data Manager using port 7001, 7002 on localhost
  - AM TEST (15) Application Manager using port 7020 on localhost
  - CM Call Manager ports 7020, 7010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



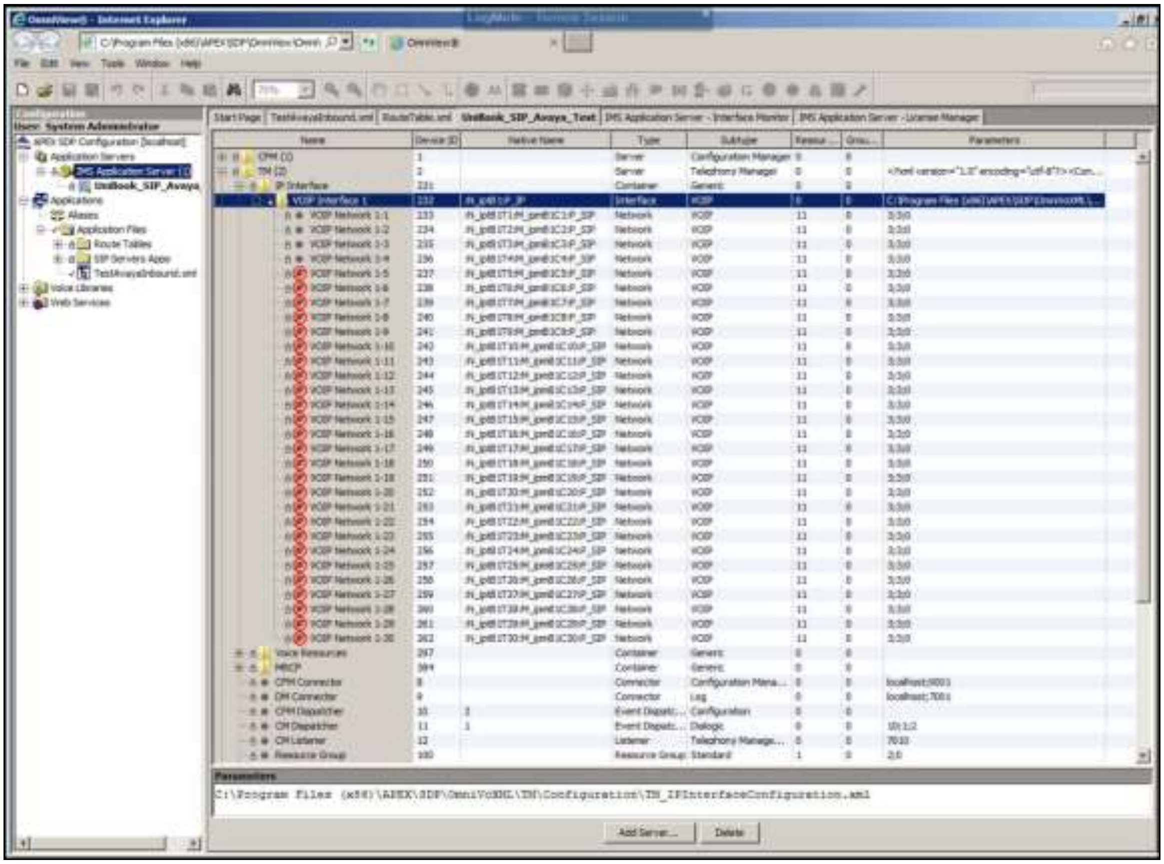
- Navigate to **Application Servers→IMS Application** Server and select **License Management** from drop down list to verify that proper licenses have been applied.



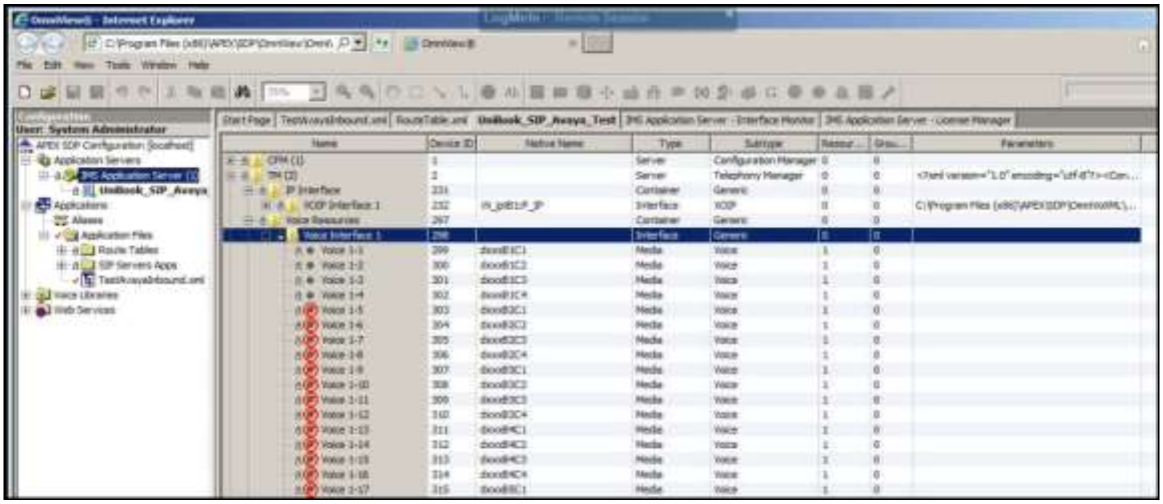
- Select **Add Dongle** followed by **Add Key** for adding all needed license keys. The screen below shows **BASE**, **VOICE** and **VOIP** keys required for telephony implementation to work.



- 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 Endpoint Configuration

The SIP Endpoints configuration is done using an XML config file named TM_IPInterfaceConfiguration.xml, located at C:\Program Files\(x86)\APEX\SDP\OmniVoXML\TM\Configuration. During the startup of SIP client (Telephony Manager) the registration events are sent from IVR to IP Office and upon authentication the IVR SIP Endpoints are registered.

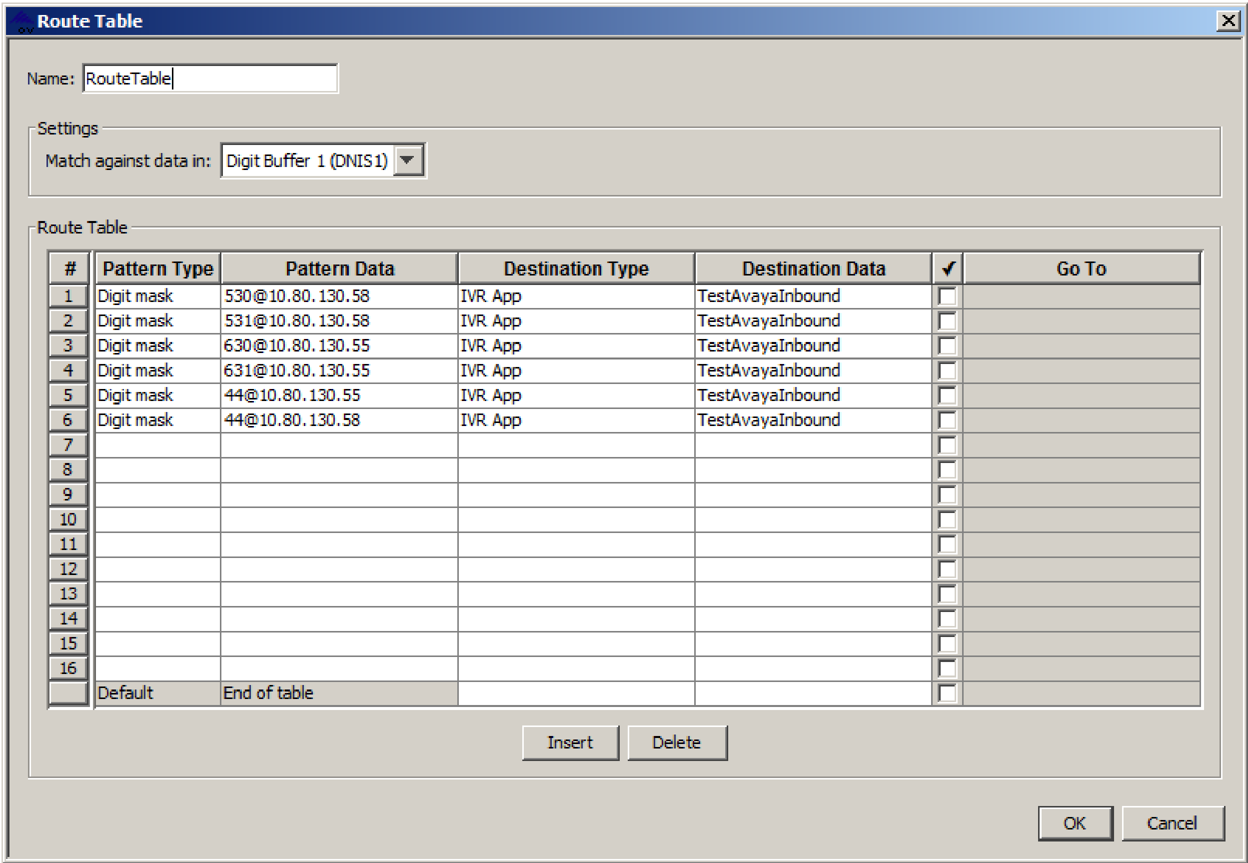
```
<Configuration>
<!--
    -       To use this configuration file, specify the path name for this file in the parameters field of the Interface/VOIP devices in
    TM.
    Normally, the path name should be set to
    ..\Configuration\TM_IPInterfaceConfiguration.xml
    -       Registration Protocol:
            SIP
            H323
    -       Alias Type
            Transparent
            DotNotation
            H323_ID
            eMail
            Phone
            URL
    -->
            <TimerForServiceUnavailable>20</TimerForServiceUnavailable>
            <RegistrationOptions>1</RegistrationOptions>
            <Registration>
                    <Protocol>SIP</Protocol>
                    <Server>10.80.130.58</Server>
                    <Client>sip:530@10.80.130.58:5060</Client>
                    <MaxMulticastHops>16</MaxMulticastHops>
                    <TimeToLive>300</TimeToLive>
                    <Alias>
                            <Type>Transparent</Type>
                            <Data>sip:530@10.80.130.200:5060</Data>
                    </Alias>
            </Registration>
```

```
        <Registration>
                <Protocol>SIP</Protocol>
                <Server>10.80.130.58</Server>
                <Client>sip:531@10.80.130.58:5060</Client>
                <MaxMulticastHops>16</MaxMulticastHops>
                <TimeToLive>300</TimeToLive>
                <Alias>
                        <Type>Transparent</Type>
                        <Data>sip:531@10.80.130.200:5060</Data>
                </Alias>
        </Registration>
        <Registration>
                <Protocol>SIP</Protocol>
                <Server>10.80.130.55</Server>
                <Client>sip:630@10.80.130.55:5060</Client>
                <MaxMulticastHops>16</MaxMulticastHops>
                <TimeToLive>300</TimeToLive>
                <Alias>
                        <Type>Transparent</Type>
                        <Data>sip:630@10.80.130.200:5060</Data>
                </Alias>
        </Registration>
        <Registration>
                <Protocol>SIP</Protocol>
                <Server>10.80.130.55</Server>
                <Client>sip:631@10.80.130.55:5060</Client>
                <MaxMulticastHops>16</MaxMulticastHops>
                <TimeToLive>300</TimeToLive>
                <Alias>
                        <Type>Transparent</Type>
                        <Data>sip:631@10.80.130.200:5060</Data>
                </Alias>
        </Registration>
        <Authentications>
                <Authentication>
                        <Realm>ipoffice</Realm>
                        <Identity>sip:530@10.80.130.58:5060</Identity>
                        <UserName>530</UserName>
                        <Password>1234</Password>
                </Authentication>
                <Authentication>
                        <Realm>ipoffice</Realm>
                        <Identity>sip:531@10.80.130.58:5060</Identity>
                        <UserName>531</UserName>
                        <Password>1234</Password>
                </Authentication>
        <Authentication>
                        <Realm>ipoffice</Realm>
                        <Identity>sip:630@10.80.130.55:5060</Identity>
                        <UserName>630</UserName>
                        <Password>1234</Password>
                </Authentication>
                <Authentication>
                        <Realm>ipoffice</Realm>
                        <Identity>sip:631@10.80.130.55:5060</Identity>
                        <UserName>631</UserName>
                        <Password>1234</Password>
                </Authentication>
        </Authentications>
        <Coders>
<!--
Options:
Capability      FramesPerPacke   VAD    FrameSize
G711_ALaw_64K           5, 10, 20, 30           // milliseconds
G711_ALaw_56K           5, 10, 20, 30           // milliseconds
G711_ULaw_64K           5, 10, 20, 30           // milliseconds
G711_ULaw_56K           5, 10, 20, 30           // milliseconds
-->
                <Coder>
```

```
                <Capability>G711_ALaw_64K</Capability>
                <Direction>Transmit</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
            <Coder>
                <Capability>G711_ALaw_64K</Capability>
                <Direction>Receive</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
            <Coder>
                <Capability>G711_ALaw_56K</Capability>
                <Direction>Transmit</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
            <Coder>
                <Capability>G711_ALaw_56K</Capability>
                <Direction>Receive</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
            <Coder>
                <Capability>G711_ULaw_64K</Capability>
                <Direction>Transmit</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
            <Coder>
                <Capability>G711_ULaw_64K</Capability>
                <Direction>Receive</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
            <Coder>
                <Capability>G711_ULaw_56K</Capability>
                <Direction>Transmit</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
            <Coder>
                <Capability>G711_ULaw_56K</Capability>
                <Direction>Receive</Direction>
                <Type>Audio</Type>
                <FramesPerPacket>20</FramesPerPacket>
                <VoiceActivityDetection>OFF</VoiceActivityDetection>
                <PayloadType>Data</PayloadType>
            </Coder>
        <!-- T38 fax codec -->
        <!--
            <Coder>
                <Capability>T.38UDP</Capability>
                <Direction>Both</Direction>
                <Type>RawData</Type>
```

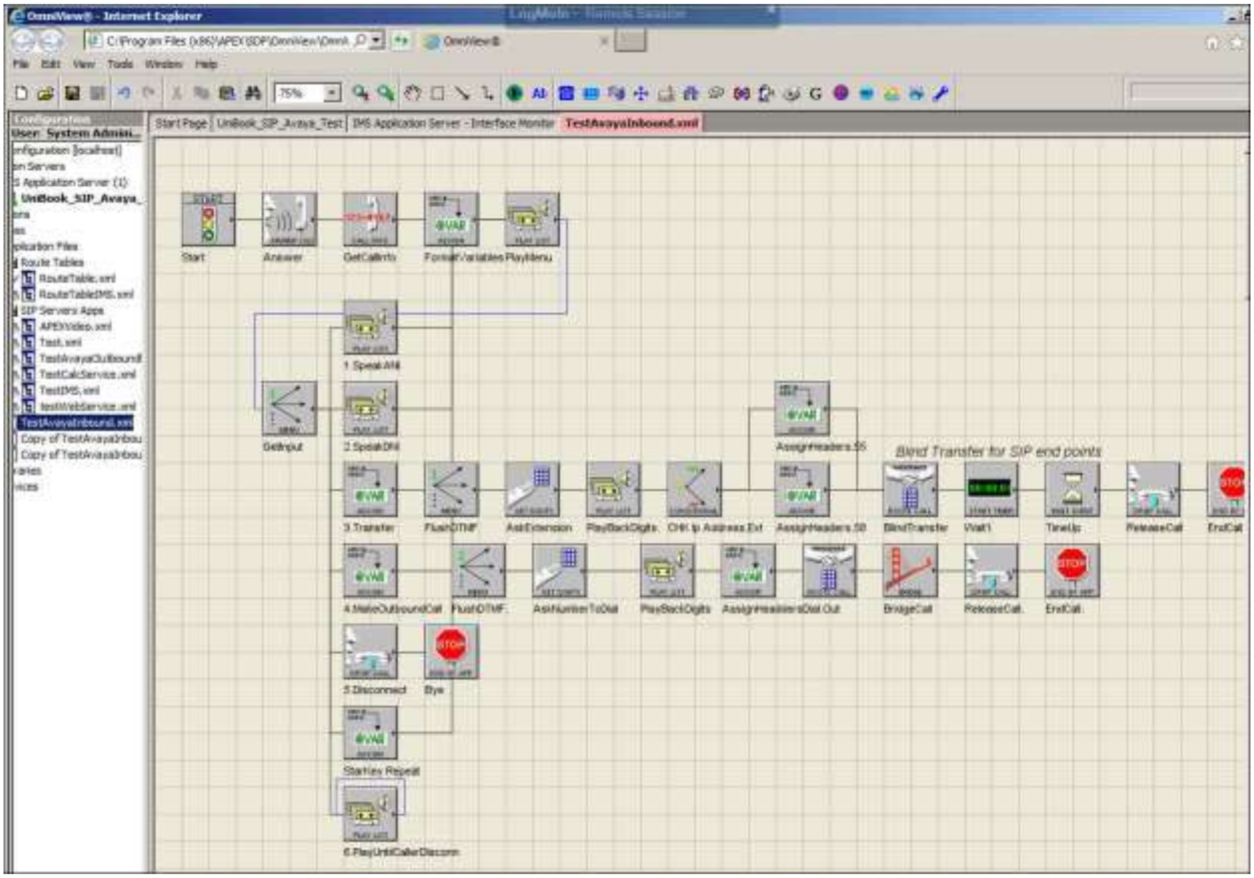
```
                                        <PayloadType>Data</PayloadType>
                                        <DataMaxBitRate>144</DataMaxBitRate>
                                </Coder>
        -->
        </Coders>
        <Alarms>
        <!--
        QoS Alarm Type:
        LostPackets
        Jitter
        -->
        <!--
        <Alarm>
                <Type>LostPackets</Type>
                <Threshold>20</Threshold>
                <DebounceOn>10000</DebounceOn>
                <DebounceOff>10000</DebounceOff>
                <Interval>1000</Interval>
                <PercentSuccess>60</PercentSuccess>
                <PercentFail>40</PercentFail>
        </Alarm>
        <Alarm>
                <Type>Jitter</Type>
                <Threshold>60</Threshold>
                <DebounceOn>20000</DebounceOn>
                <DebounceOff>60000</DebounceOff>
                <Interval>5000</Interval>
                <PercentSuccess>60</PercentSuccess>
                <PercentFail>40</PercentFail>
        </Alarm>
        -->
        </Alarms>
        <!--
        Operating Mode:
                IP_T38_AUTOMATIC_MODE
                IP_T38_MANUAL_MODE
                IP_T38_MANUAL_MODIFY_MODE
        -->
                <OperatingMode>IP_T38_MANUAL_MODIFY_MODE</OperatingMode>
                <StreamingStatus>No</StreamingStatus>
                <EnableEchoCancellation>No</EnableEchoCancellation>
        <!--
        SIP Extensions for Caller Identity and Privacy
        rpi-token   = rpi-screen | rpi-pty-type |rpi-id-type | rpi-privacy | other-rpi-token
        rpi-screen  = "screen" "=" ("no" | "yes" )
        rpi-pty-type= "party" "=" ( "calling" | "called"  | token )
        rpi-id-type = "id-type" "=" ( "subscriber" | "user" | "alias" | "return" | "term"  | token )
        rpi-privacy = "privacy" "=" 1#(
                        ("full" | "name" | "uri" | "off" | token )
                        [ "-" ( "network" | token ) ]  )
                        [ "-" ( "network" | token ) ]  )
        other-rpi-token    = token ["=" (token | quoted-string)]
        other-user         = token  | "private"
        -->
        <!-- <RemotePartyID>party=calling;privacy=off</RemotePartyID>
         -->

</Configuration>
```

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.



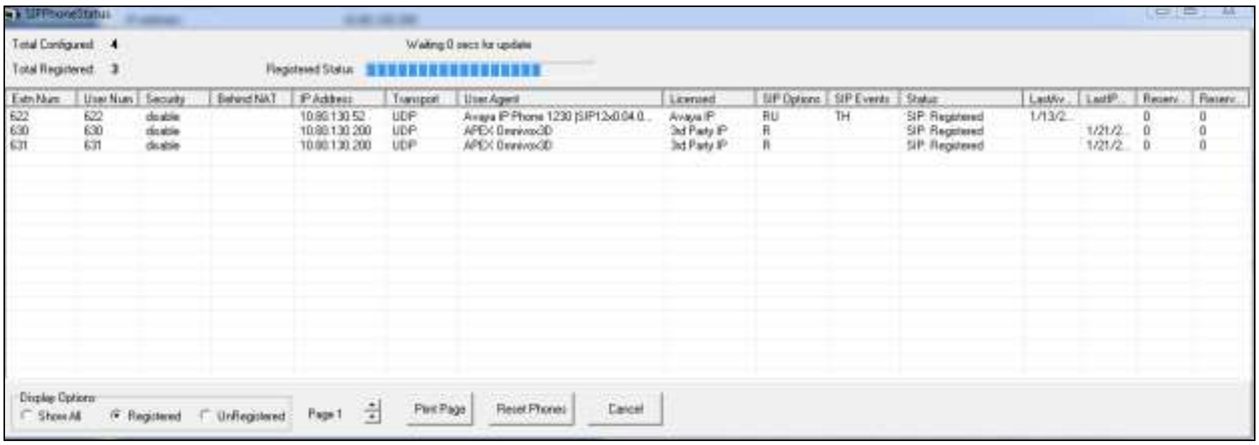
6. Verification Steps

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

6.1. IVR SIP Endpoint to IP Office

Log in to the IP Office Monitor application. From the left hand menu, navigate **Status→SIP Phone Status**

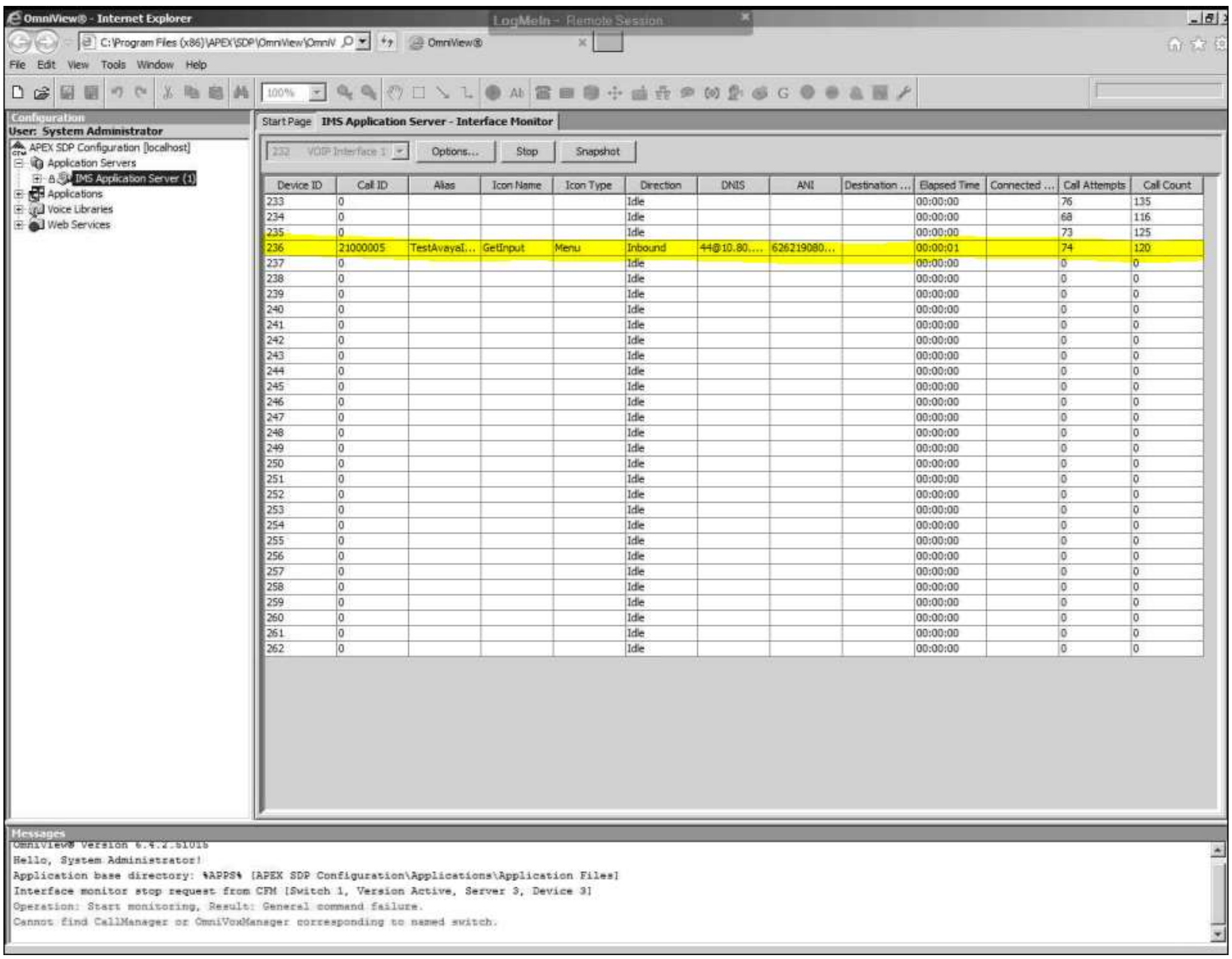
- Verify **Status** column is set to **SIP:Registered** for IVR SIP Endpoint



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.