Avaya Solution & Interoperability Test Lab

Application Notes for AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone with Avaya IP Office Server Edition - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephones with Avaya IP Office Server Edition. SVX-8208-SMBU IP SIP DECT Handset/Base Telephones serve the hospitality industry and provide the following features: speakerphone, hold and message waiting indicator (MWI). In the compliance test, SVX-8208-SMBU IP SIP DECT Handset/Base Telephones successfully registered with IP Office, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya IP Office Short Codes.

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 steps required to integrate AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephones with Avaya IP Office Server Edition. SVX-8208-SMBU IP SIP DECT Handset/Base Telephones serve the hospitality industry and provide the following features: speakerphone, hold and message waiting indicator (MWI). In the compliance test, SVX-8208-SMBU IP SIP DECT Handset/Base Telephones successfully registered with Avaya IP Office, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya IP Office Short Codes.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephones and Avaya SIP and H.323 telephones. Basic telephony features, such as hold, speaker, and hospitality features like do not disturb were exercised. In addition, other extended telephony features, such as call forwarding and call pickup were also exercised using Short Codes.

The serviceability testing focused on verifying that the AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone comes back into service after re-connecting the Ethernet connection or rebooting the SIP phone.

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 this DevConnect Application Note 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 AEi Communications did not include use of any specific encryption features as requested by AEi Communications.

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/handsets that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/handsets for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of SVX-8208-SMBU IP SIP DECT Handset/Base with IP Office.
- Calls between SVX-8208-SMBU IP SIP DECT Handset/Base and Avaya SIP and H.323 telephones with Direct IP-IP Media (Shuffling) enabled and disabled (see **Section 5.2**).
- Support of multiple incoming and outgoing calls, using L1 and L2.
- G.711 MU-Law codec support.
- Proper recognition of DTMF tones.
- Long call duration and long hold duration.
- Extended telephony features using Short Codes, such as Call Forwarding and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voicemail messages.
- Proper system recovery after a restart of the SVX-8208-SMBU IP SIP DECT Handset/Base and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- The SVX-8208-SMBU IP SIP DECT Handset/Base Telephone does not support conference.
- The SVX-8208-SMBU IP SIP DECT Handset/Base Telephone does not support transfer however a call can be transferred to it.

2.3. Support

For technical support on the AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone, contact AEi Communications Support via phone, email, or website.

- **Phone:** +1 (650) 552-9416
- **Email:** sales@aeicommunications.com
- **Web:** http://www.aeicommunications.com/contact.html

3. Reference Configuration

Error! Reference source not found. illustrates a sample configuration with an Avaya IP Office network that includes the following Avaya products:

- IP Office Server Edition with an IP Office IP500 V2 connected as an expansion box.
- Avaya H.323 and SIP telephones.
- SVX-8208-SMBU IP SIP DECT Handset/Base Telephone interfaced to IP Office as a SIP Endpoint.

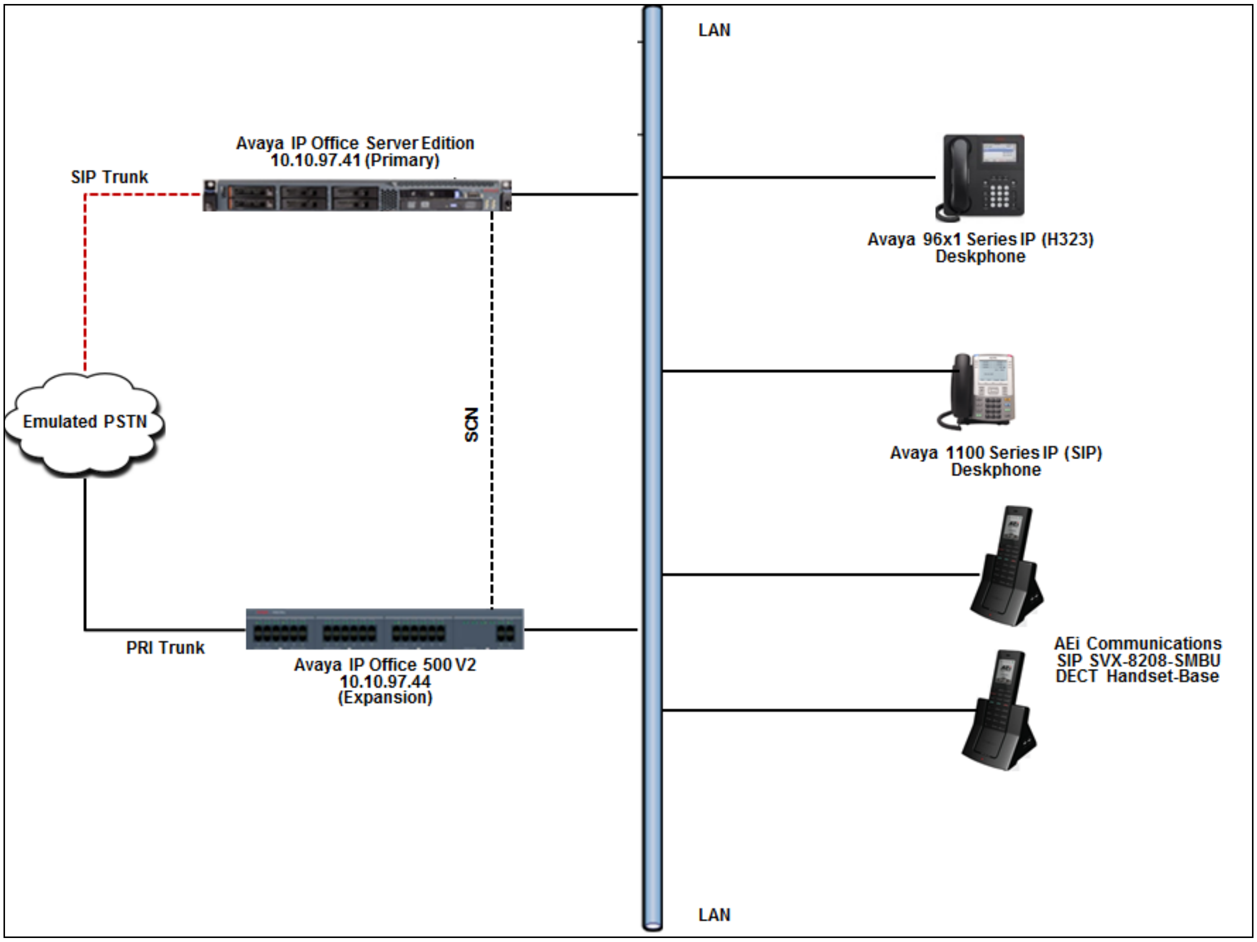


Figure 1: Avaya IP Office Network with AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephones

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Linux (Primary)	10.0 SP3
Avaya IP Office IP500 V2 (Expansion)	10.0 SP3
Avaya Telephones: • 9650 IP (H323) Deskphone • 1140 IP (SIP) Deskphone • 9641 IP (H323) Deskphone	 3.270B 04.04.23.00 6.6401
AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone	SVX8210_C13

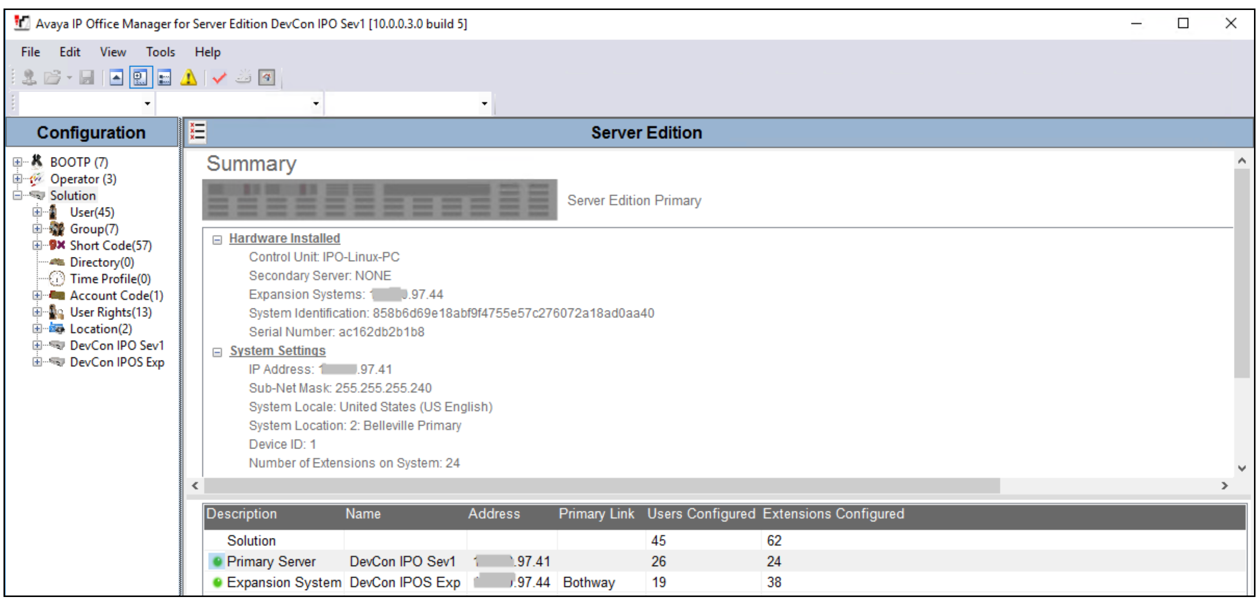
Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office IP500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office

Configuration and verification operations on the Avaya IP Office illustrated in this section were all performed using Avaya IP Office Manager. The information provided in this section describes the configuration done on the Primary (Linux server) system and the same configuration applies to the Expansion (IP500 V2) system too. It is implied a working system is already in place with the necessary licensing. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**. The configuration operations described in this section can be summarized as follows:

- Configure System
- Administer SIP Extensions for SVX-8208-SMBU IP SIP DECT Handset/Base Telephone
- Administer SIP Users for SVX-8208-SMBU IP SIP DECT Handset/Base Telephone
- Save Configuration

From a PC running the IP Office Manager application, select **Start → Programs →IP Office → Manager** to launch the Manager application. Select the proper IP Office system, and log in using the appropriate credentials. The Avaya IP Office Manager for Server Edition screen is displayed as shown below.

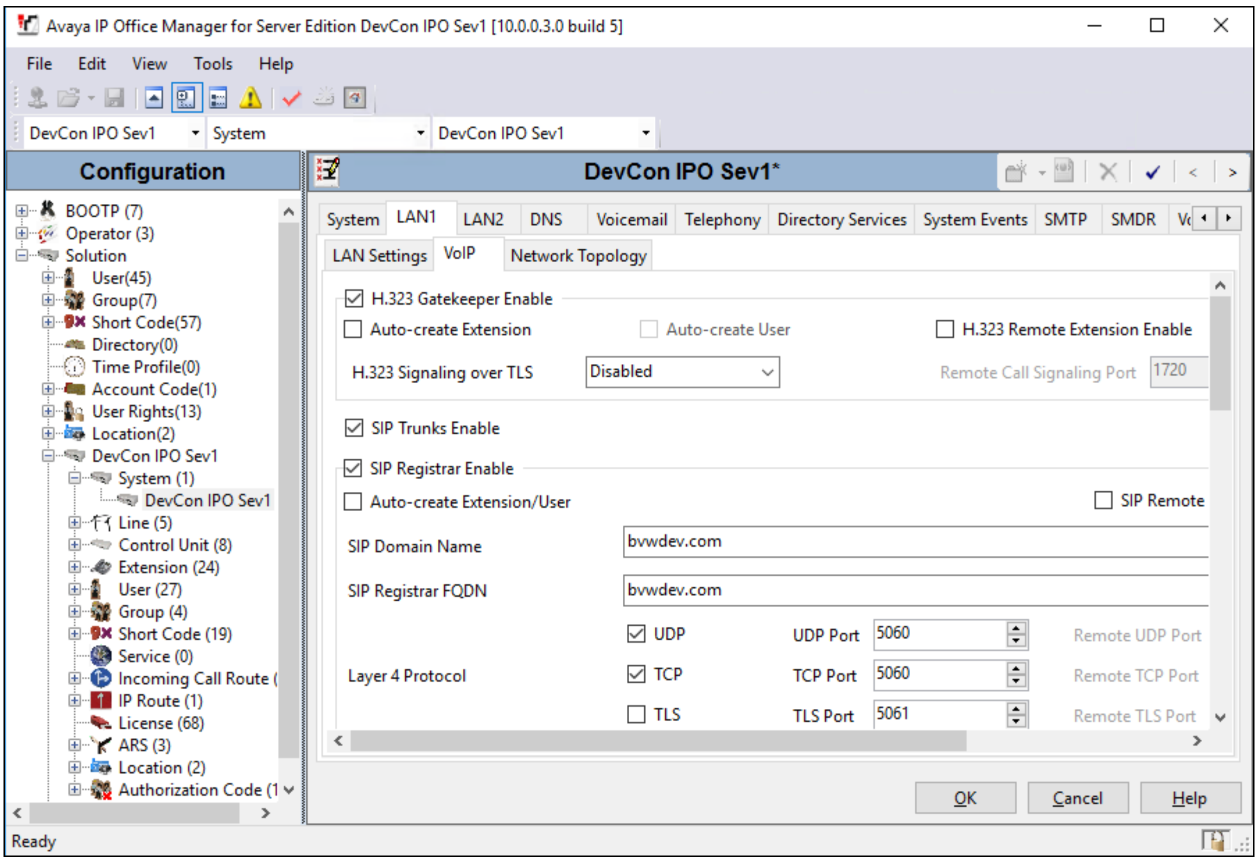


5.1. Configure System

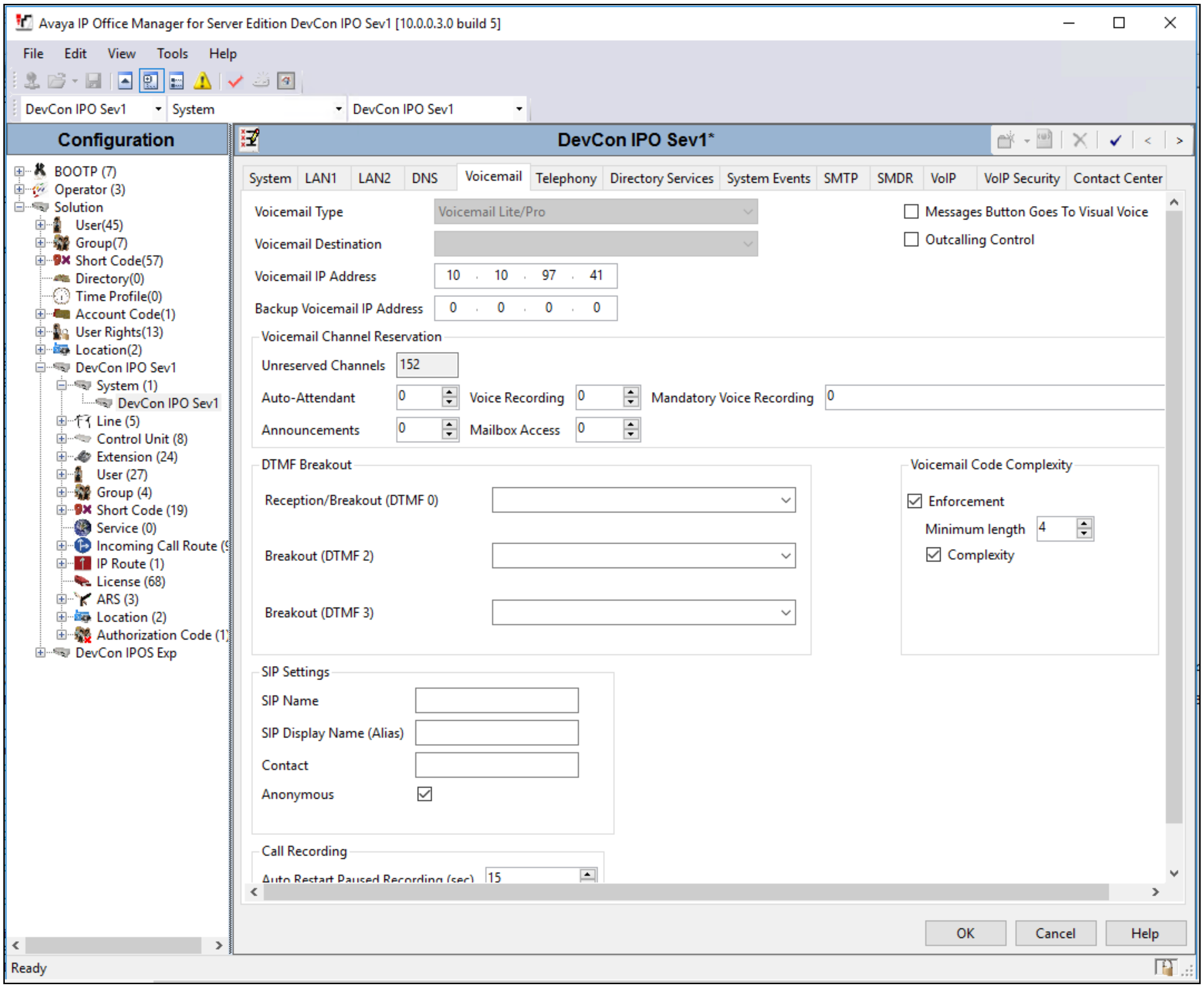
From the configuration tree in the left pane, select **DevCon IPO Sev1 → System → DevCon IPO Sev1** to display the screen in the right pane, where **DevCon IPO Sev1** is the name of the IP Office system.

Select the **LAN1** tab, IP Office can support SIP on the LAN1 and/or LAN2 interfaces, however during compliance testing the LAN1 interface was used.

Select the **VoIP** sub-tab. Make certain that **SIP Registrar Enable** is checked, as shown below. Enter a valid **SIP Domain Name** for SIP endpoints to use for registration with IP Office or this field can be left blank. During compliance testing, the **SIP Domain Name** of *bvwdev.com* was configured. Also, ensure that depending on the setup either **UDP** or **TCP** is enabled. During this compliance testing UDP protocol was used to communicate with SVX-8208-SMBU IP SIP DECT Handset/Base Telephone.

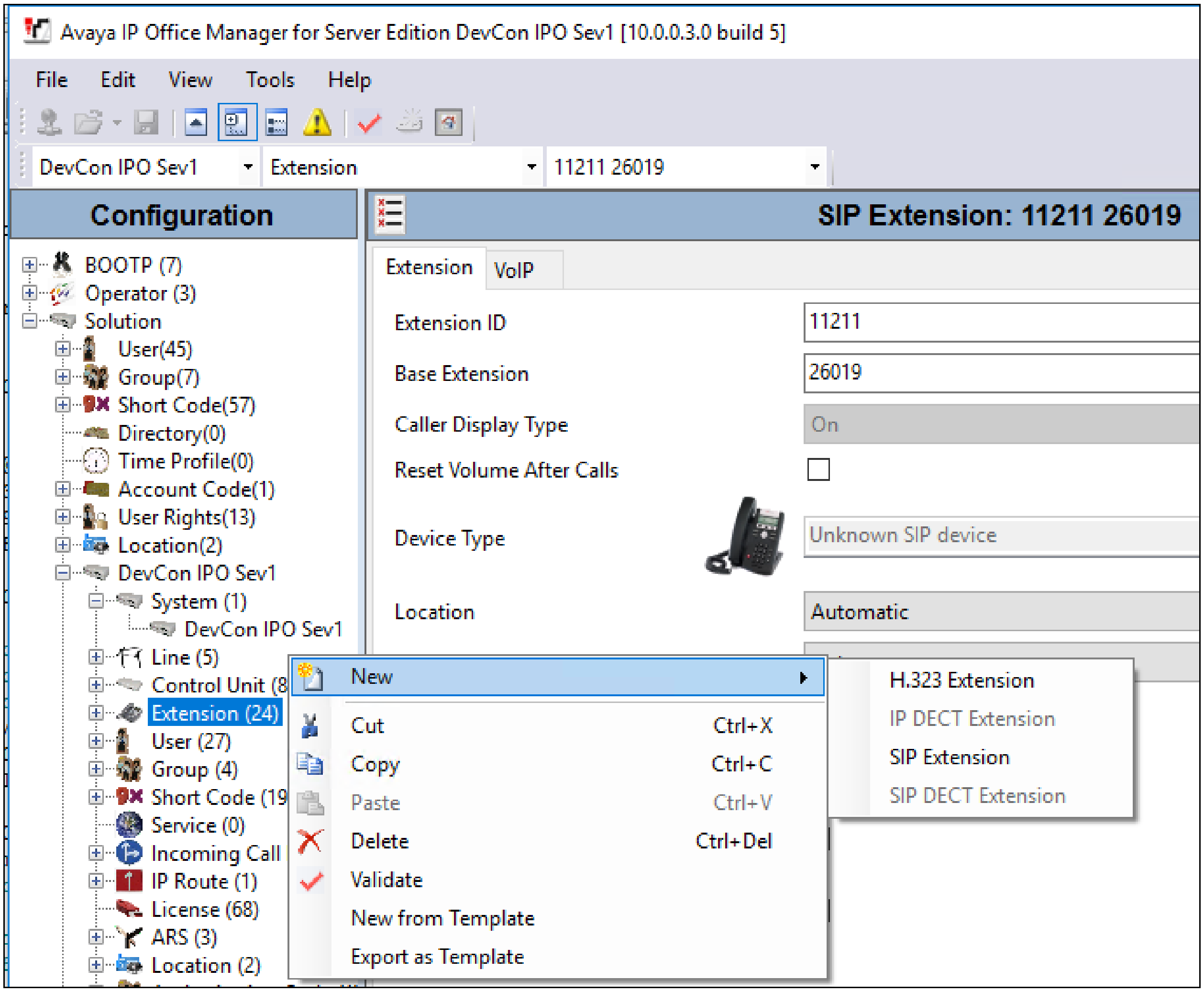


Retain default values for all fields in the **Vociemail** tab as shown in the screen below.



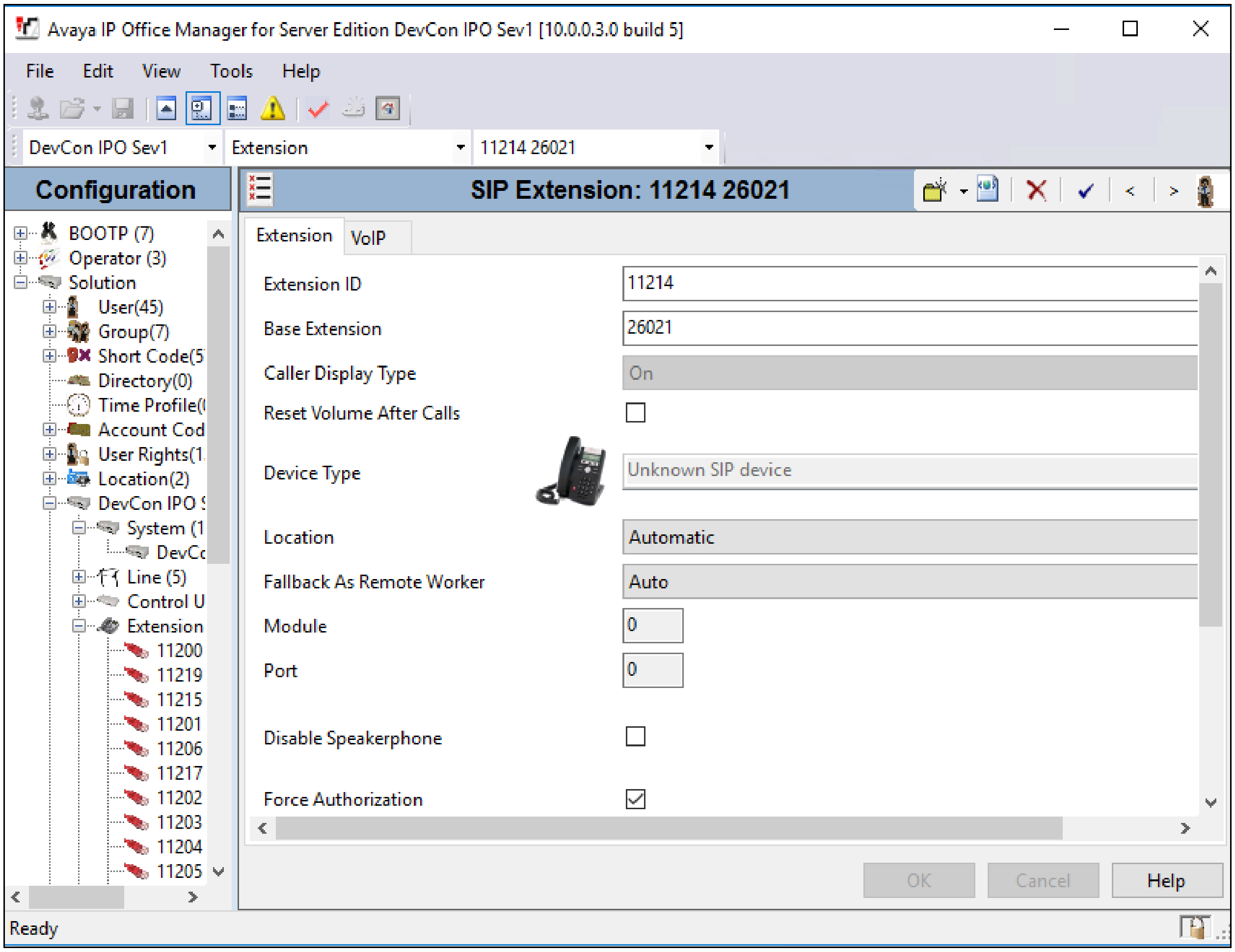
5.2. Administer SIP Extensions for SVX-8208-SMBU IP SIP DECT Handset/Base Telephone

From the configuration tree in the left pane, navigate to IP Office primary server **DevCon IPO Sev1 → Extension** and right-click on **Extension**. Select **New → SIP Extension** from the pop-up list to add a new SIP extension as shown in the screen below.

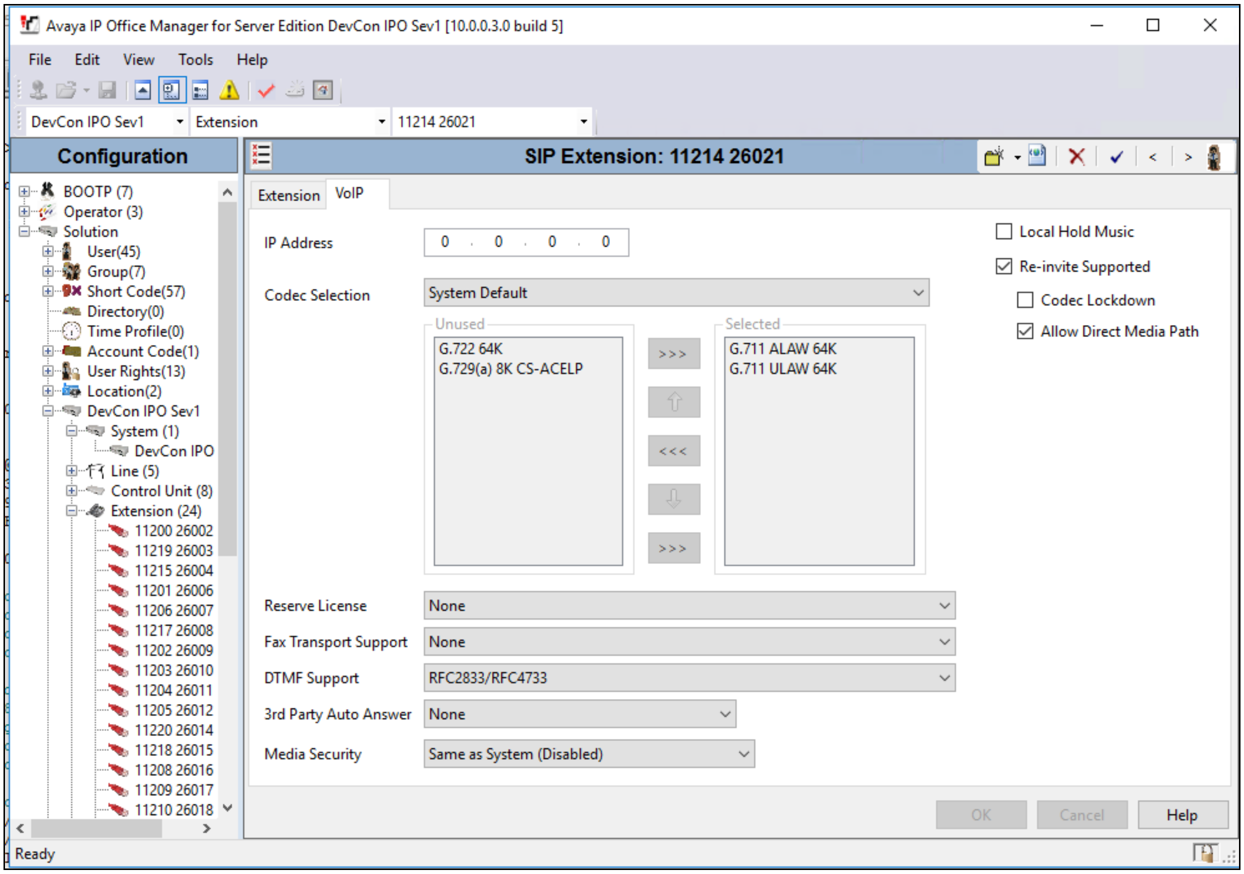


In the following window, under the **Extension** tab, enter the desired digits for **Base Extension** as shown below. Retain default values for all other remaining fields.

Repeat this section to add the desired number of SIP extensions. In the example below, a SIP extension with base extension of *26021* was created.

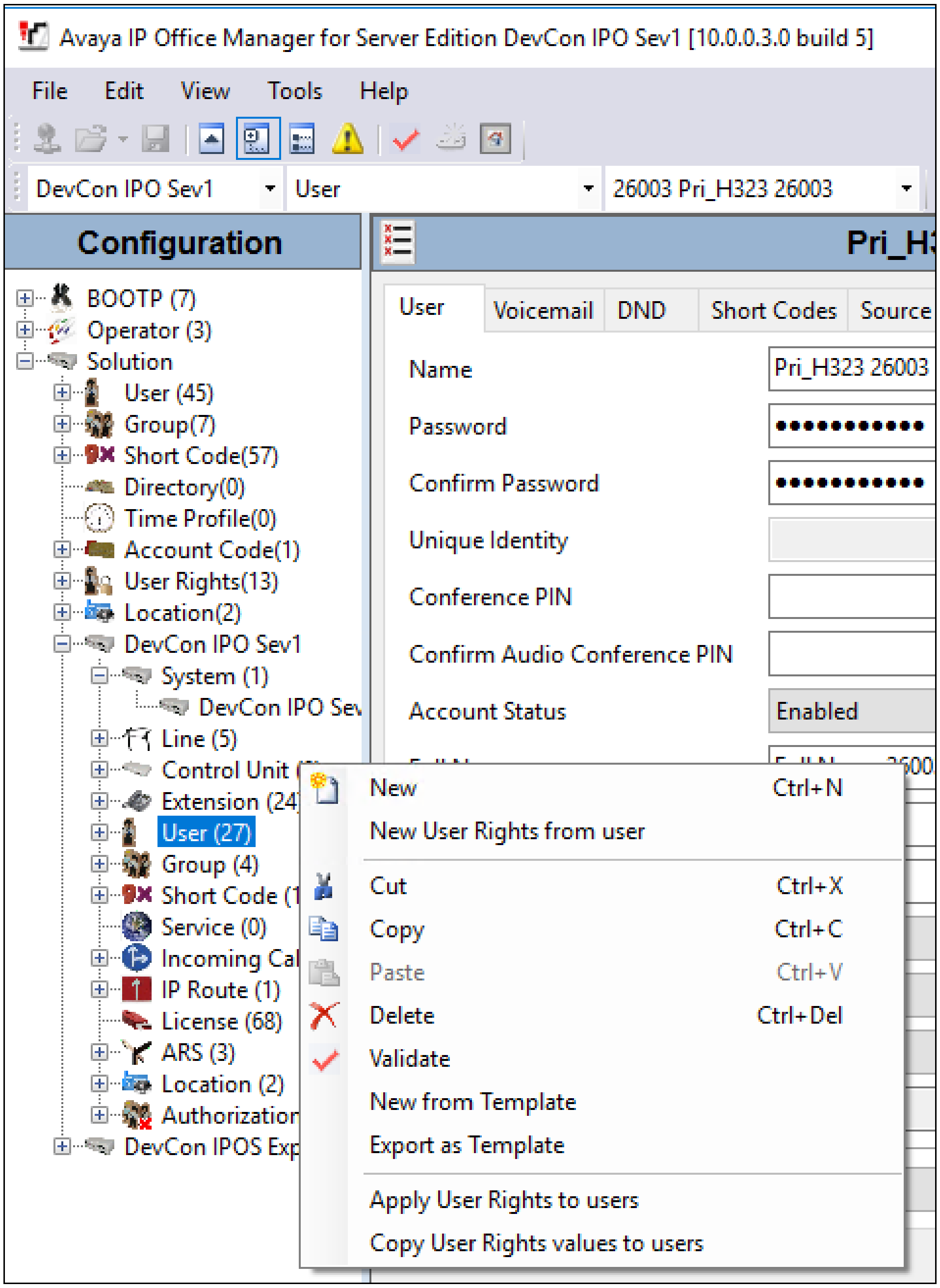


Default values were retained for all fields under the **VoIP** tab. During compliance testing only *G.711ULAW* codec was tested. By default the **Allow Direct Media Path** box is selected. This allows the SIP phone to establish direct media path (shuffling) with other SIP phones that also support this feature. If required this feature can be disabled by unchecking the box.



5.3. Administer SIP Users for SVX-8208-SMBU IP SIP DECT Handset/Base Telephone

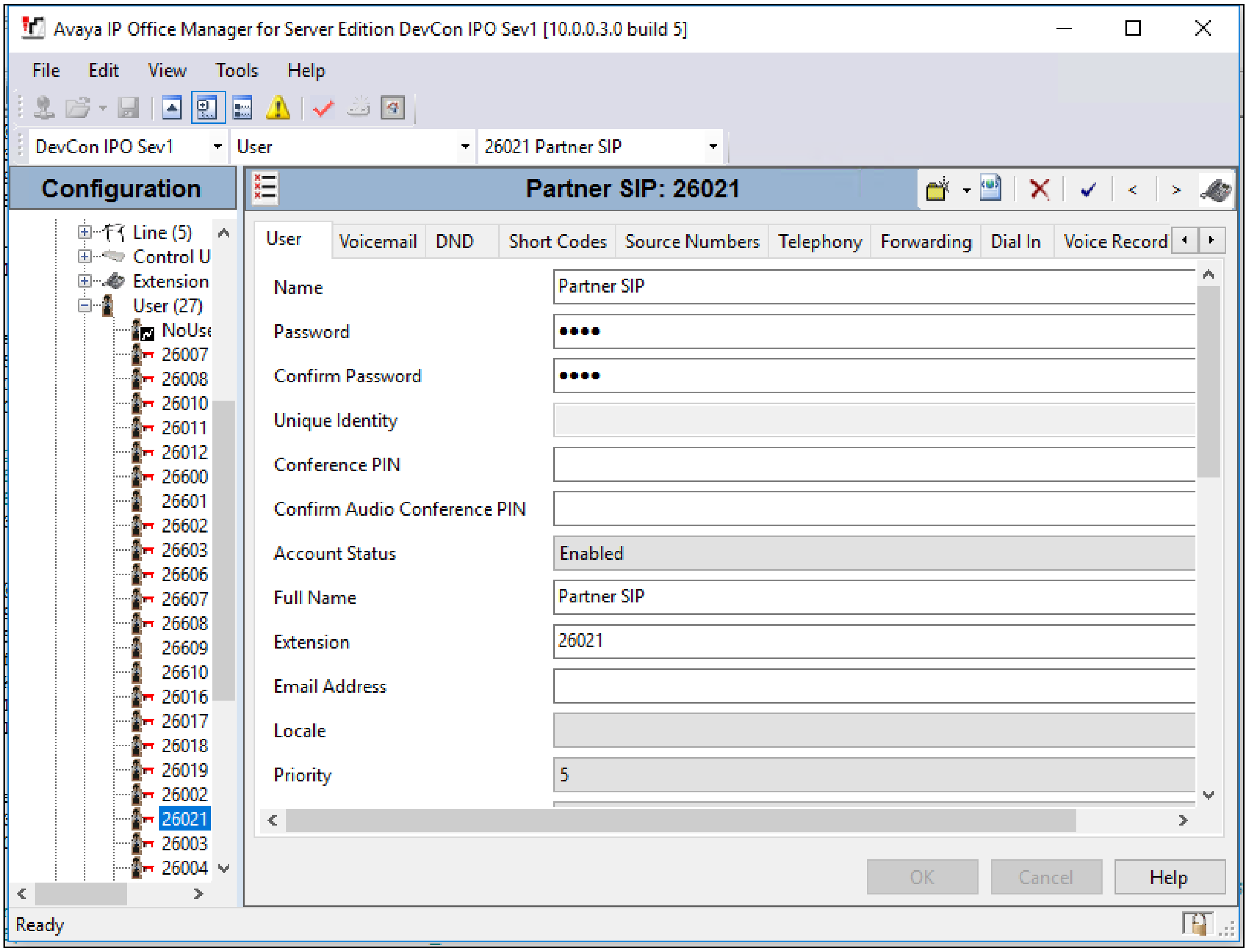
From the configuration tree in the left pane, navigate to IP Office primary server **DevCon IPO Sev1** → **User** and right-click on **User**. Select **New** from the pop-up list to add a new user.



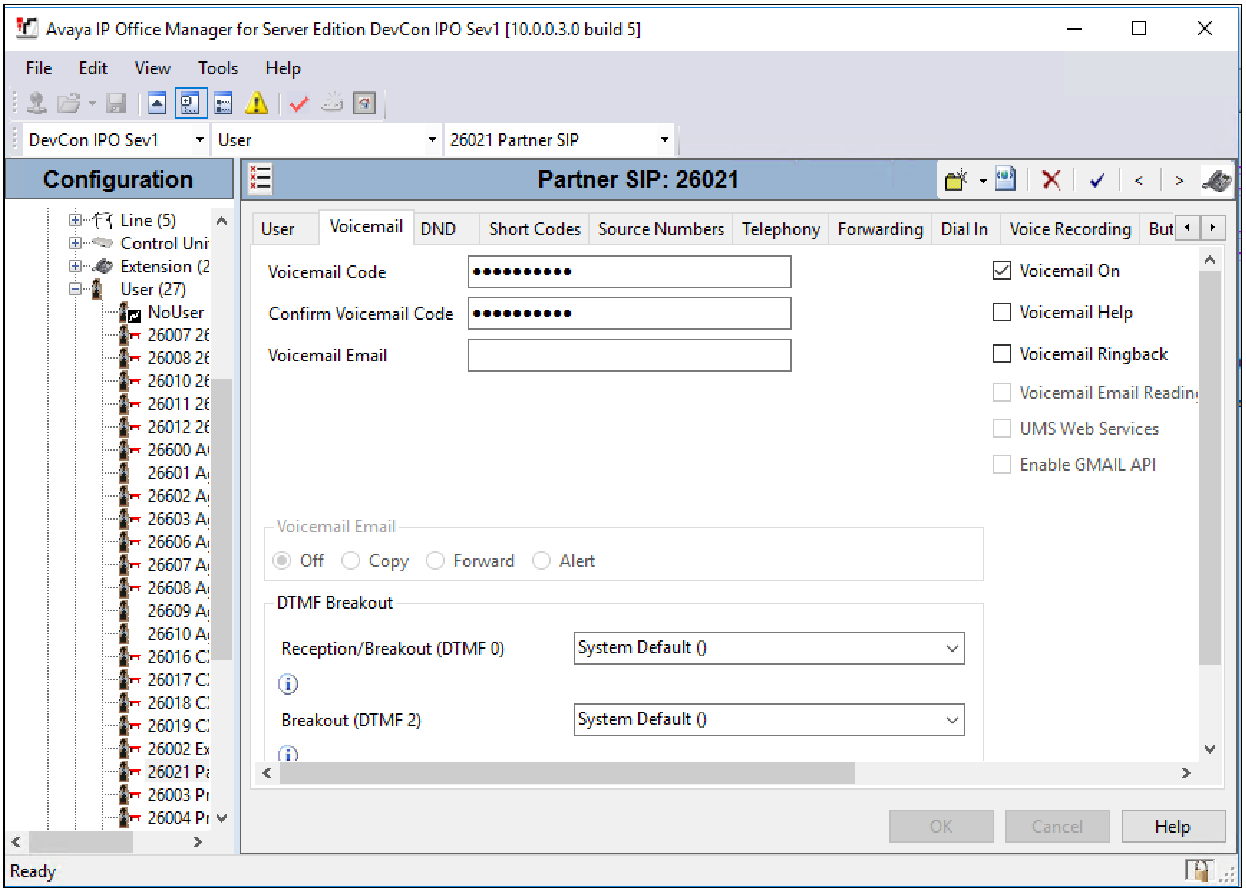
In the following window, configure the values for the **User** tab as shown in the screen below.

- **Name:** A descriptive name
- **Password and Confirm Password:** A valid numeric password
- **Full Name:** A descriptive name
- **Extension:** The extension configured in **Section 5.2**

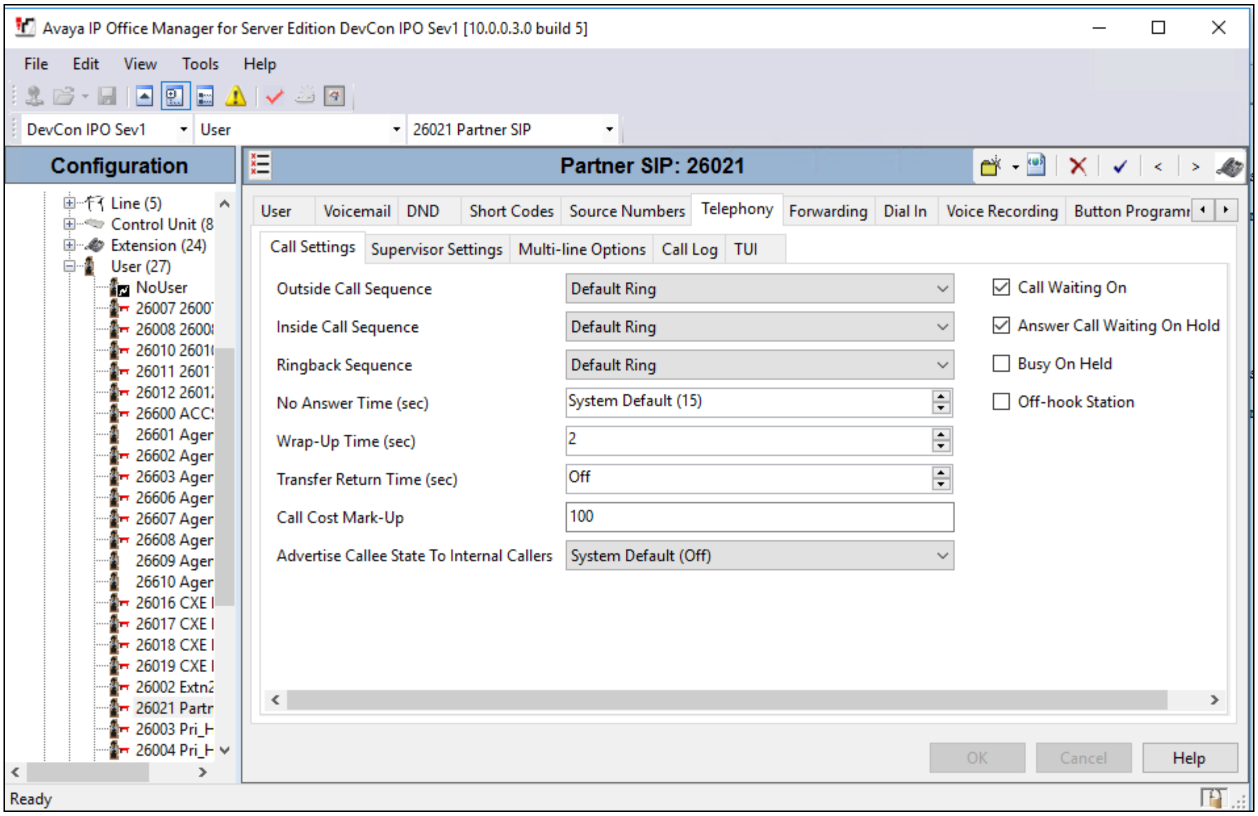
Retain default values for all other fields.



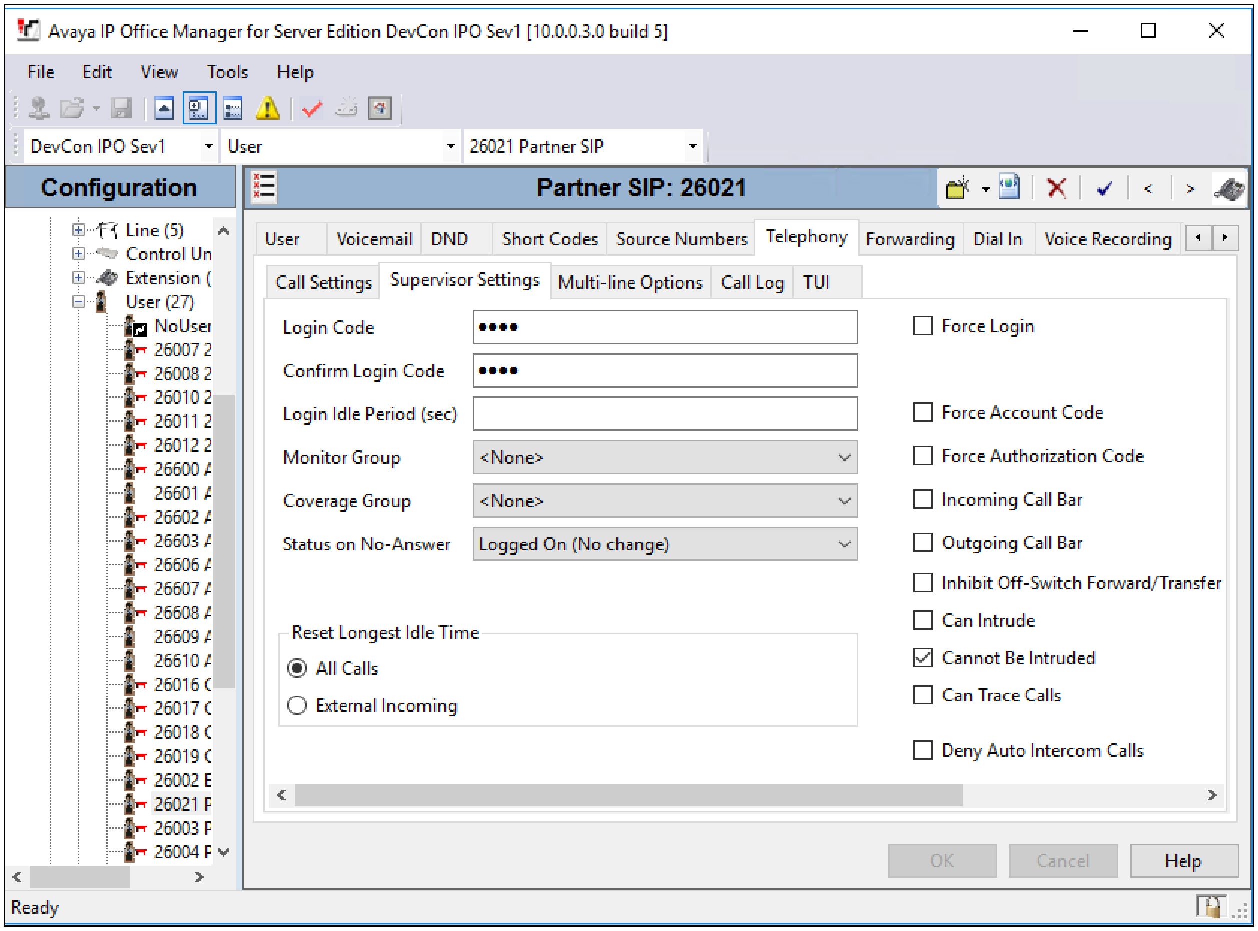
Under the **Voicemail** tab, retain default values for all fields and ensure that the **Voicemail On** box is checked and a numeric value is configured in the **Voicemail Code** and **Confirm Voicemail Code** as shown in the screen below. Retain default values for all other fields.



Under the **Telephony → Call Settings** tab, retain all default values and ensure that the **Call Waiting On** box is checked as shown below.

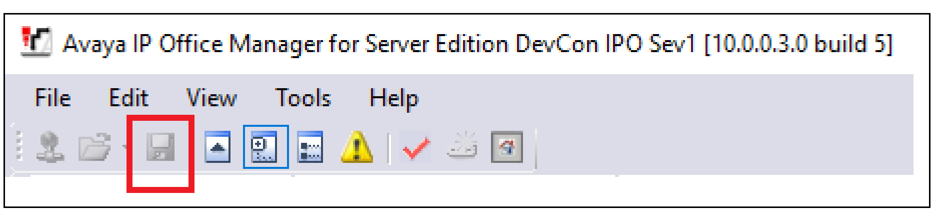


Under the **Telephony → Supervisor Settings** tab, retain all default values and ensure that the **Login Code and Confirm Login Code** fields are configured as shown below. This login code will be used while configuring SVX-8208-SMBU IP SIP DECT Handset/Base Telephone in **Section 6.2**.



5.4. Save Configuration

Once all the configurations are complete, the changes need to be saved on the IP Office System. Click on the Save icon as shown in the screen below to save the changes. A subsequent window will appear (not shown) asking the user to proceed with the changes made to the IP Office system/s or not. Click on the **OK** button to confirm.

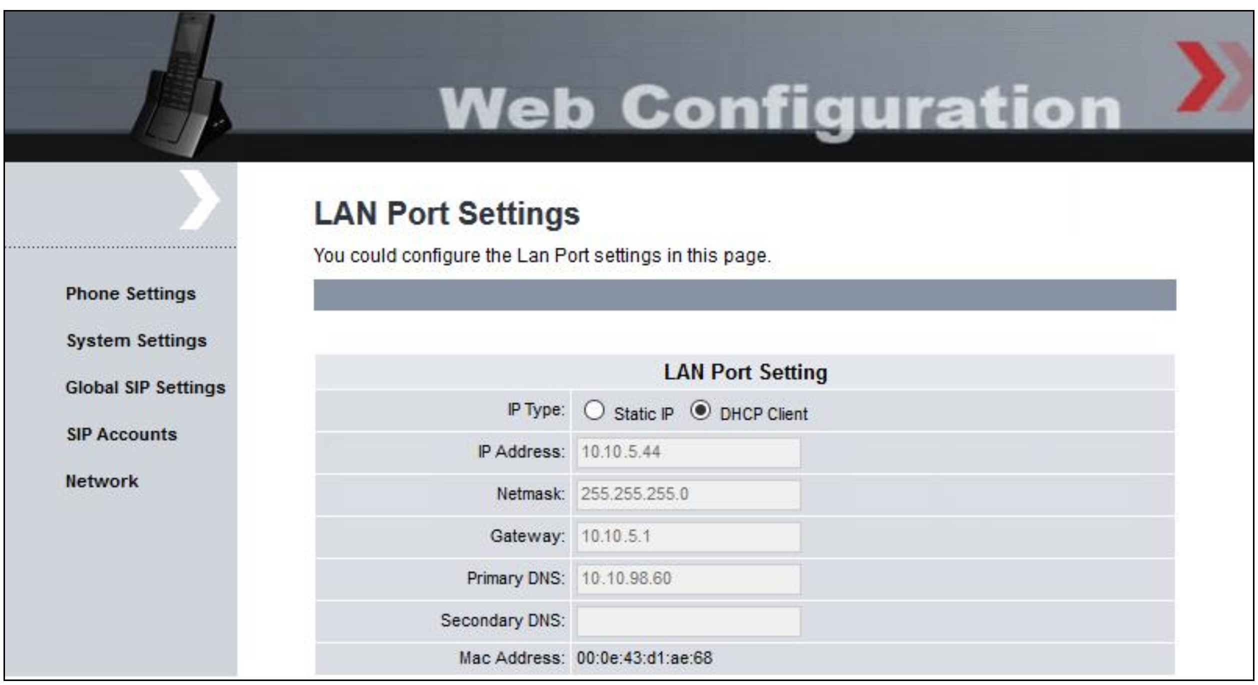


6. Configure AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone

Access the SVX-8208-SMBU IP SIP DECT Handset/Base web interface by using the URL "https://ip-address:8000" in an Internet browser window, where "ip-address" is the IP address of the SIP phone. Log in using the appropriate credentials and then click **Login**.

6.1. Administer LAN Port Settings

Select **Network → LAN Port Settings** in the left pane and configure the SIP phone's network settings as shown below. During the compliance test, DHCP was utilized.



6.2. Administer SIP Accounts

Navigate to **SIP Accounts** in the left pane and click **Add** to add a SIP account.

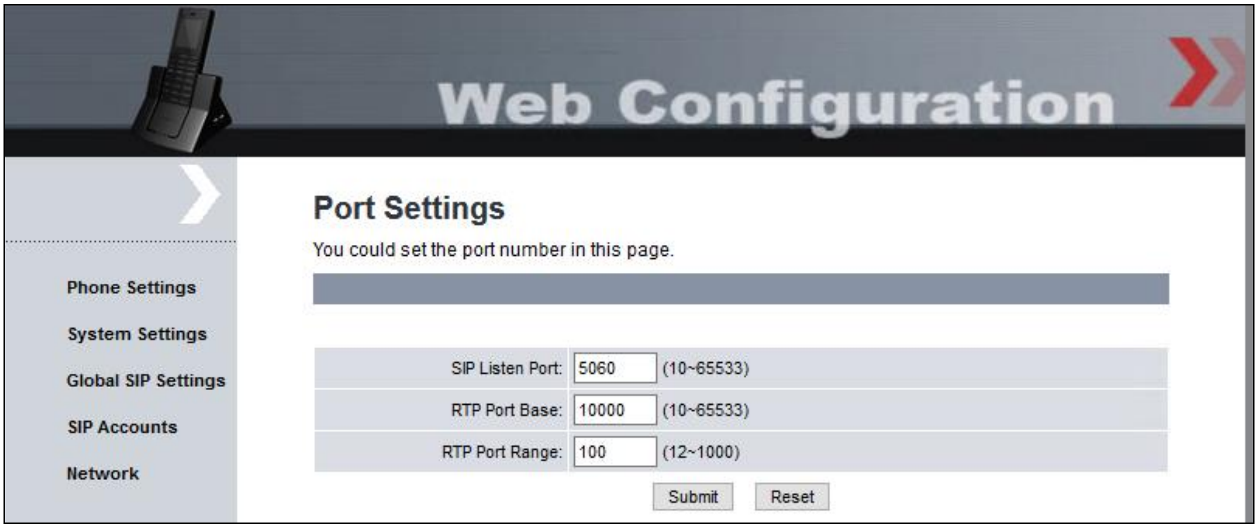
Navigate to the **SIP Proxy** webpage as shown below. Under the **Basic SIP Proxy Settings** section, configure the following parameters.

- **Registration ID:** Specify the Registration ID (e.g., *26021*, the SIP extension).
- **Display Name:** Specify the Display Name (e.g., *26021*, the SIP extension).
- **Authentication Name:** Specify the SIP extension of the SVX-8208-SMBU IP SIP DECT Handset/Base Telephone (e.g., *26021*).
- **Password:** Specify the SIP password configured in **Section 5.3**.
- **Registration Server:** Set to the Primary server IP Address and port (e.g., *10.10.97.41:5060*).
- **Proxy Server:** Set to the Primary server IP address and port (e.g., *10.10.97.41:5060*).
- **Voice Mail:** Specify the voicemail pilot number if required.
- **MWI:** Set to *Enable*.
- Retain the default values in the remaining fields.

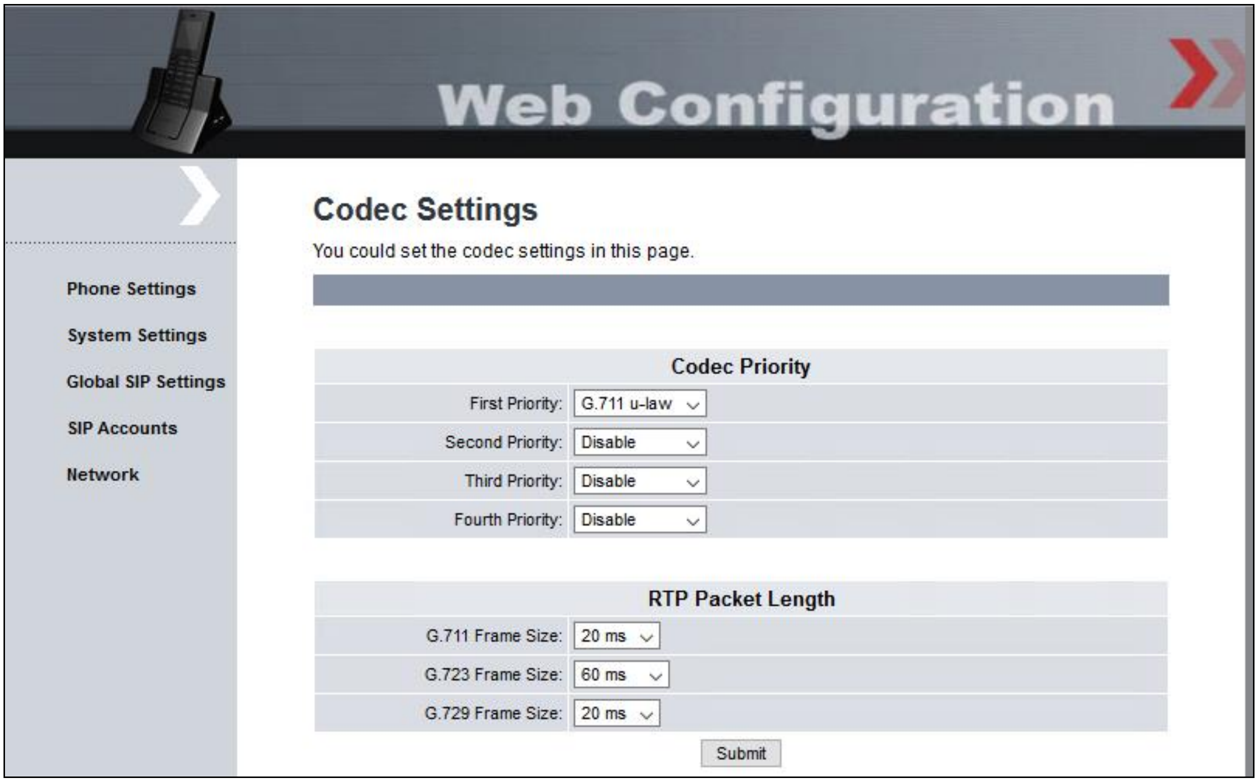
Notice at the bottom of the screen that the status is *registered* with IP Office Primary server.

6.3. Administer Global SIP Settings

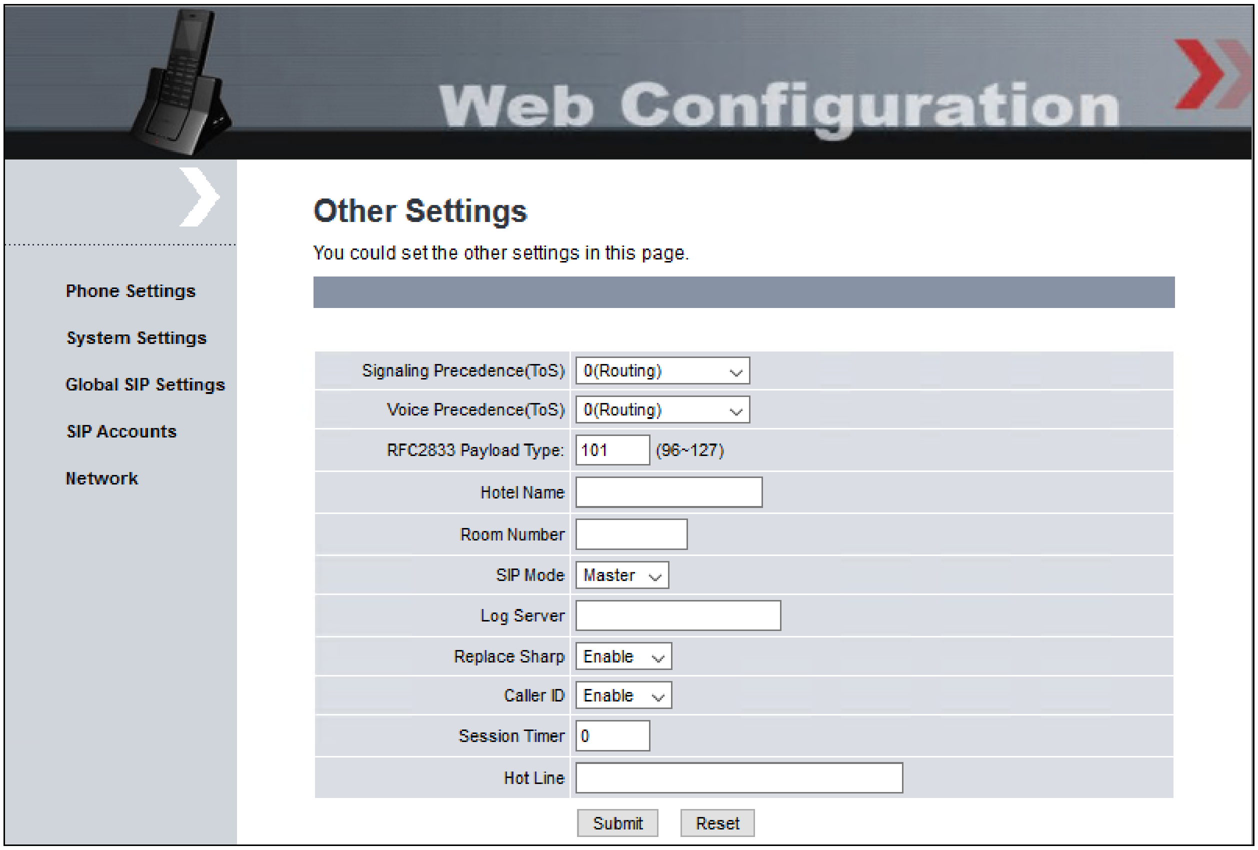
Navigate to **Global SIP Settings → Port Settings** and verify the SIP Listen Port being used (e.g., *5060*).



Navigate to **Global SIP Settings → Codec Settings** to verify the codec priority. In this example, the first priority is *G.711u-law*. AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephones supports G.711.



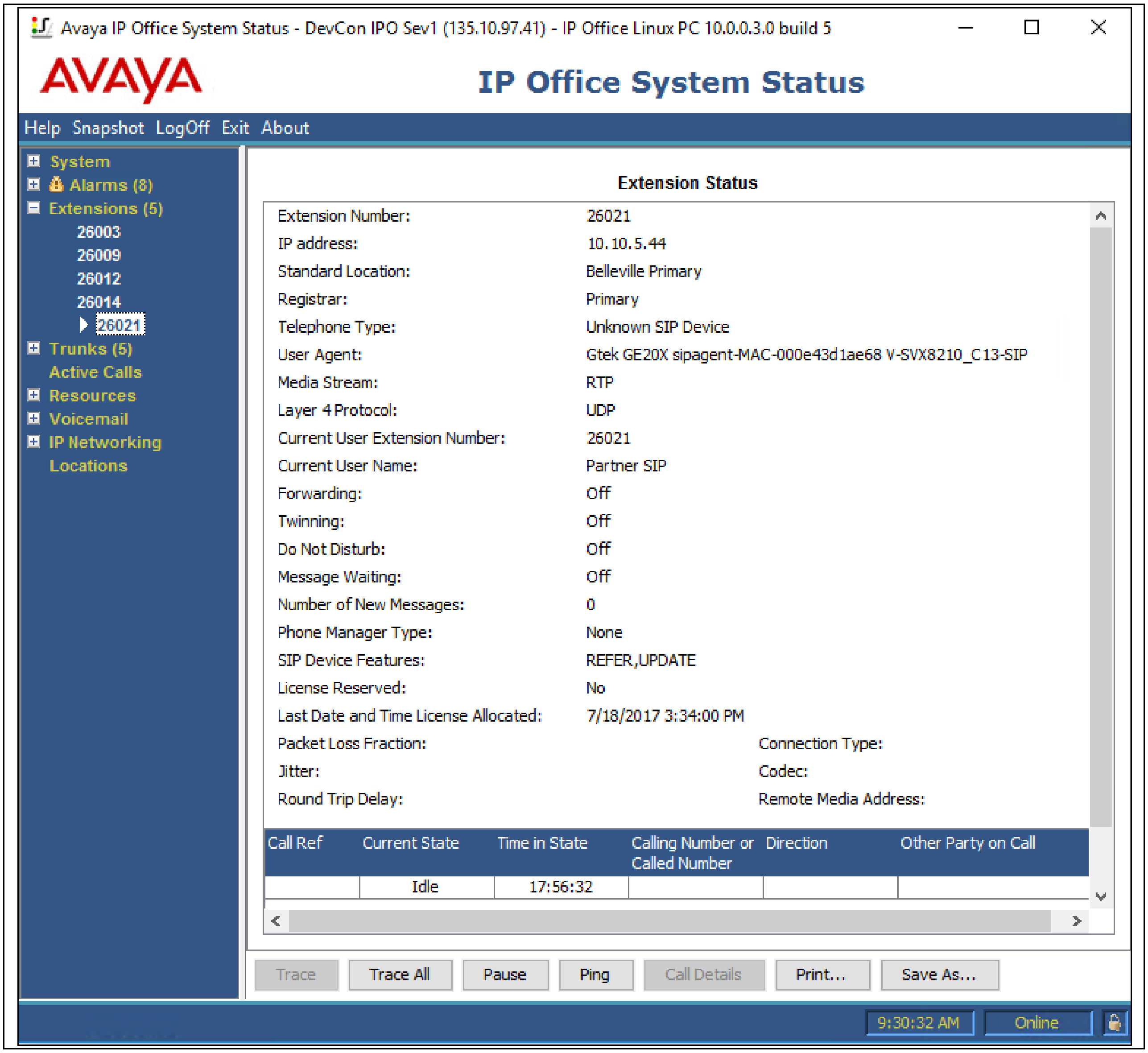
Navigate to **Global SIP Settings → Other Settings**, and verify the Caller ID was set to *Enable*, so that the calling party ID can be displayed during the conversation. Retain default value for all other fields.



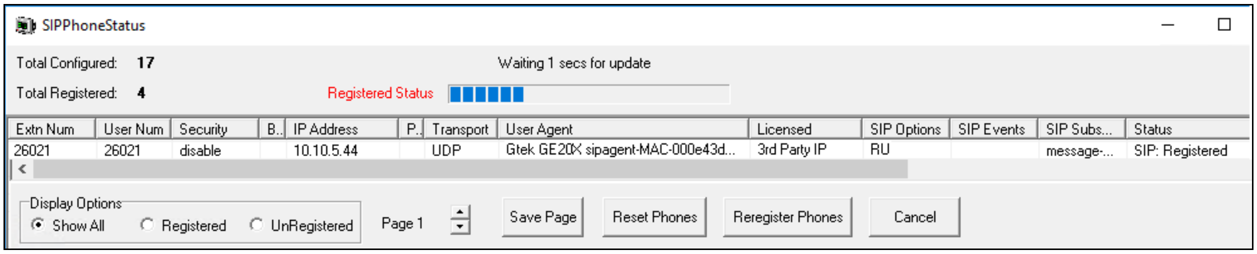
7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone with Avaya IP Office.

1. From the Avaya IP Office System Status window for the Primary server, verify that the SIP extension (endpoint) is **Registrar** to *Primary*.



2. Registration can also be verified from the IP Office System Monitor window as shown below. The screen below shows the endpoint that was configured for SVX-8208-SMBU IP SIP DECT Handset/Base Telephone as registered.



3. The SIP registration status can also be seen in the SIP Account page of the SVX-8208-SMBU IP SIP DECT Handset/Base Telephone web interface seen in **Section 6.2**.

4. Verify basic telephony features by establishing calls between a SVX-8208-SMBU IP SIP DECT Handset/Base Telephone with another SVX-8208-SMBU IP SIP DECT Handset/Base phone and with Avaya SIP and H.323 endpoints.

8. Conclusion

These Application Notes have described the administration steps required to integrate the AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone with Avaya IP Office Server Edition. The AEi Communications SVX-8208-SMBU IP SIP DECT Handset/Base Telephone successfully registered with Avaya IP Office as a SIP endpoint and basic telephony and hospitality features were verified. All test cases passed with any observations noted in **Section 2.2**.

9. References

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

1. *Deploying IP Office ™ Platform Server Edition Solution,* Release 10.0.
2. *Administering Avaya IP Office™ Platform with Manager,* Release 10.0.
3. *Deploying Avaya IP Office™ Platform IP500 V2,* 15-601042 Issue 31I.

The following document was provided by AEi Communications.

1. *Configuring Hospitality SVM-8x08-SMKG IP SIP Phone,* Version 1.1, Date: 28/07/16.

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.