Avaya Solution & Interoperability Test Lab

Application Notes for Cetis 3500IP Series and 9700IP Series SIP Telephones Version 3.0.0.53 with IP Office Server Edition Release 11.1 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Cetis 3500IP Series and 9700IP Series SIP Telephones with Avaya IP Office. The Cetis 3500IP Series and 9700IP Series are corded and cordless telephones. They were designed for the hospitality industry and register with Avaya IP Office Server Edition.

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 the Cetis 3500IP Series and 9700IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetis 3500IP Series and 9700IP Series SIP Telephones were designed for the hospitality industry. In the compliance test, Cetis SIP telephones registered with Avaya IP Office Server Edition.

In the compliance testing, Avaya IP Office Server Edition system consists of Avaya IP Office Linux based primary server running on virtualized environment and an IP500 V2 expansion.

2. General Test Approach and Test Results

This section details the general approach to the testing, what was covered, and results of the testing. If the testing was successfully concluded but it was necessary to implement workarounds or certain non-critical features did not work, it should be noted in **Section 2.2**.

Note: For compliance testing the Cetis 3500IP Series and 9700IP Series SIP telephones registered to the IP Office Server Edition server only and not with the IP500 V2 expansion.

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 these DevConnect Application Notes 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 these Application Notes, the interface between Avaya systems and the Cetis SIP telephones does not utilize TLS and secure media SRTP encryption features as requested by Cetis.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Cetis 3500IP Series and 9700IP Series SIP telephones and Avaya SIP and H.323 telephones and exercising basic telephony features, such as hold, mute, transfer and conference. In addition, hospitality features, such as call forward and do not disturb were covered. Interoperability compliance testing covered the following features and functionality:

- SIP registration of Cetis 3500IP and 9700IPSIP telephones with IP Office.
- Calls between Cetis telephones and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetis telephones and the PSTN.
- G.711 and G.729 codec support.
- Transport protocol TCP and UDP.
- Proper recognition of DTMF tones.
- Basic telephony features, including inbound/outbound, hold, mute, call forward, transfer and conference.
- Use of programmable buttons on the Cetis telephones.
- Proper system recovery after a restart of the Cetis telephones and loss of IP connectivity.

The serviceability testing focused on verifying that the Cetis 3500IP Series and 9700IP Series SIP telephones come back into service after re-connecting the Ethernet connect or rebooting the phone.

2.2. Test Results

All test cases passed with the following issue noted:

- There is an issue with blind transfer when a Cetis SIP telephone calls an Avaya SIP endpoint in the IPO Primary and the Avaya SIP endpoint makes a blind transfer to an Avaya H.323 endpoint on the Expansion server; after the transfer is completed there is no audio from both endpoints. The issue does not happen with attended transfer, or blind transfer to a SIP endpoint on the expansion server, or blind transfer within the same server. This issue is currently under investigation by Avaya and Cetis.

2.3. Support

For technical support on the Cetis 3500IP and 9700IP Telephones, contact Cetis support via phone, email, or website.

- **Phone:** + 1 (719) 638-8821
- **Email:** customerservice@cetisgroup.com **or** sipsupport@cetisgroup.com
- **Web:** http://www.cetisgroup.com/sipsupport/

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Cetis 3500IP Series and 9700IP Series SIP telephones with Avaya IP Office Server Edition. The Cetis SIP telephones registered with Avaya IP Office via SIP. For compliance testing the Cetis 3500IP Series and 9700IP Series SIP telephones registered to the IP Office Server Edition server only and not with the IP500 V2 expansion.

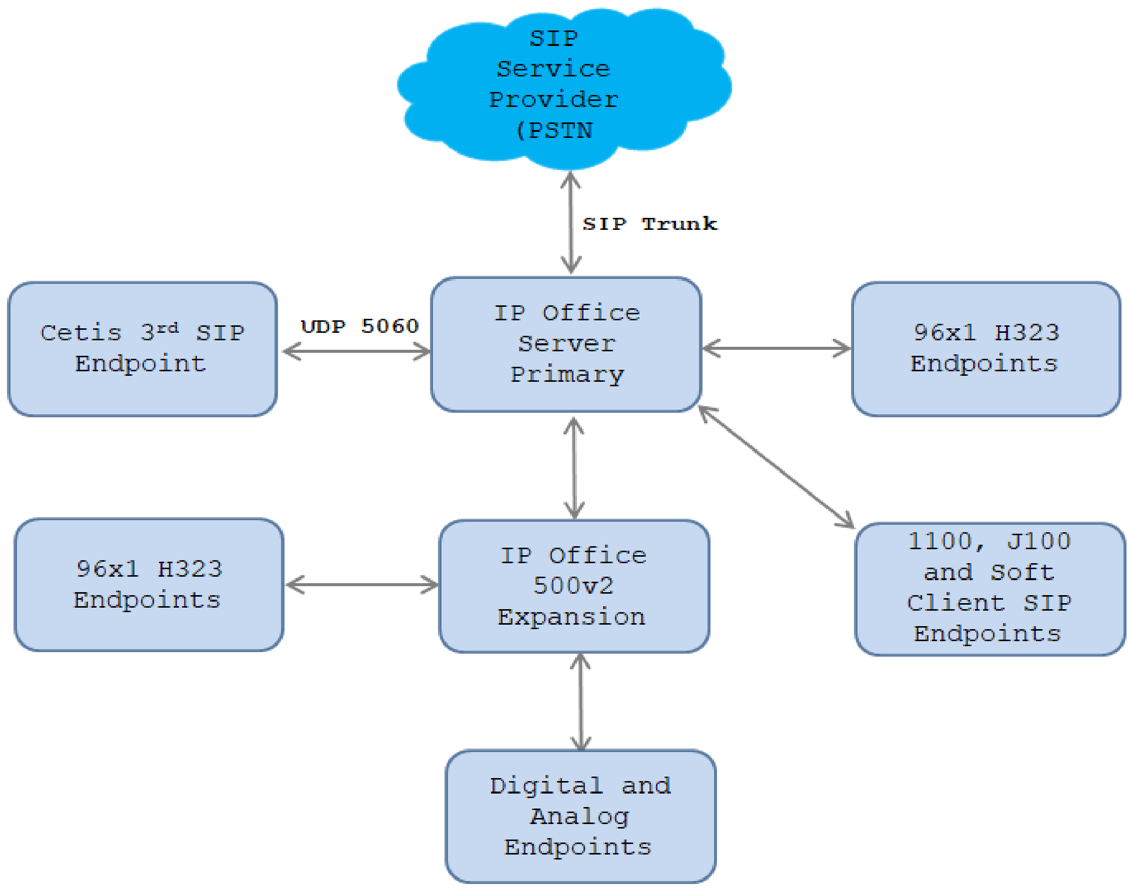


Figure 1: Test Configuration Diagram with IP Office

The following table indicates the IP addresses that were assigned to the systems in the test configuration diagram:

Description	**IP Address**
IP Office Primary	10.10.97.110
IP Office 500v2 Expansion	10.10.97.230
H323 Endpoints	10.33.5.10-11
SIP Endpoints	10.33.5.12-14
Cetis SIP Endpoints	172.16.199.5-6

4. Equipment and Software Validated

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

Equipment	Release/Version
Avaya IP Office Primary Linux running on Virtualized Environment	11.1.0.0.0237
Avaya IP Office 500V2 Expansion	11.1.0.0.0.237
Avaya IP Office Manager	11.1.0.0.0.237
Avaya 1140E SIP Deskphones	4.04.23
Avaya 96x1 IP Deskphones	6.8
Cetis 3500IP Series and 9700IP Series SIP Telephones	3.0.0.53

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

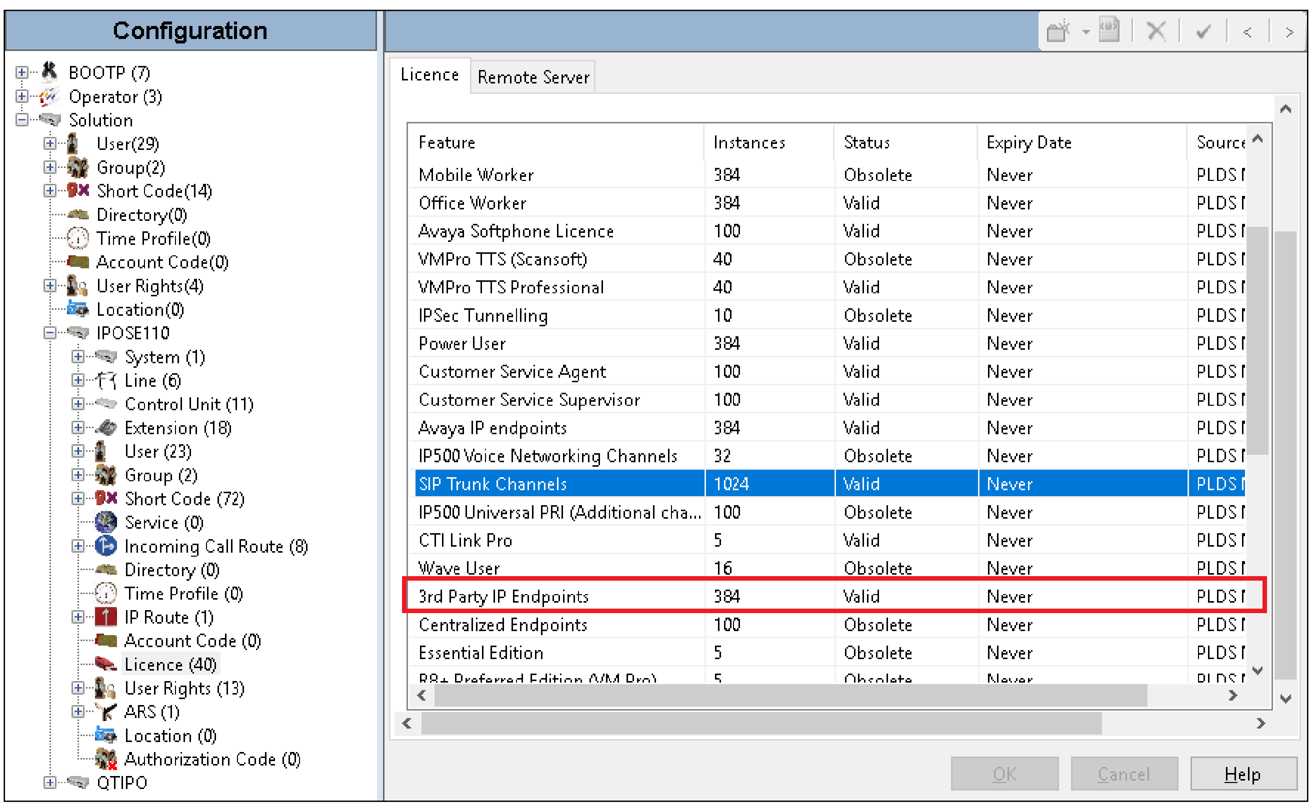
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify Avaya IP Office License
- Obtain LAN IP address
- Enable SIP Registrar
- System Telephony Settings
- Administer Codec Settings
- Administer Extension for Cetis SIP Endpoint
- Administer SIP User for Cetis SIP Endpoint

5.1. Verify Avaya IP Office License

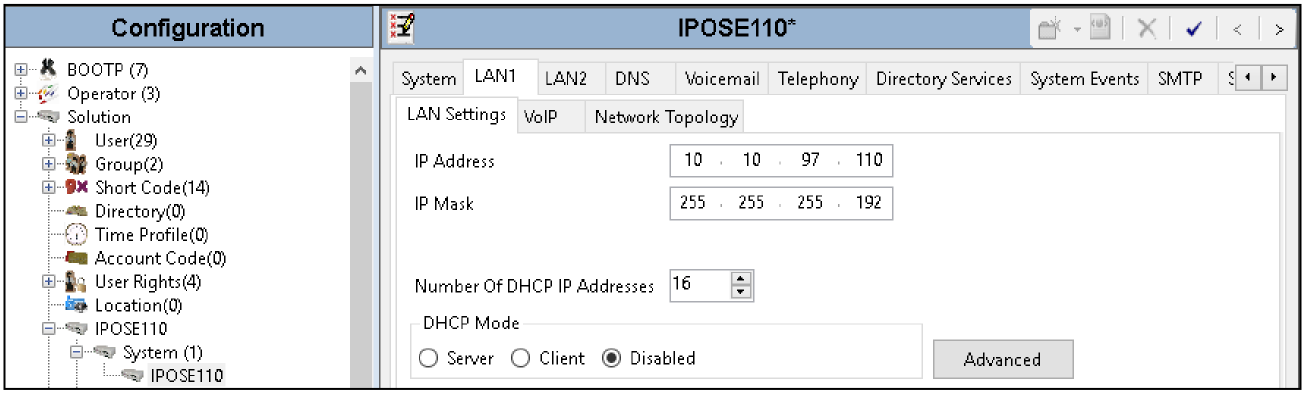
From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the Manager application. Select the correct IP Office system and log in with the appropriate credentials.

The **Avaya IP Office Manager** screen is displayed. From the configuration tree in the left pane, select **License**. Verify that the **3rd Party IP Endpoints** license is "Valid", and that the **Instances** value is sufficient for the desired maximum number of simultaneous registrations.



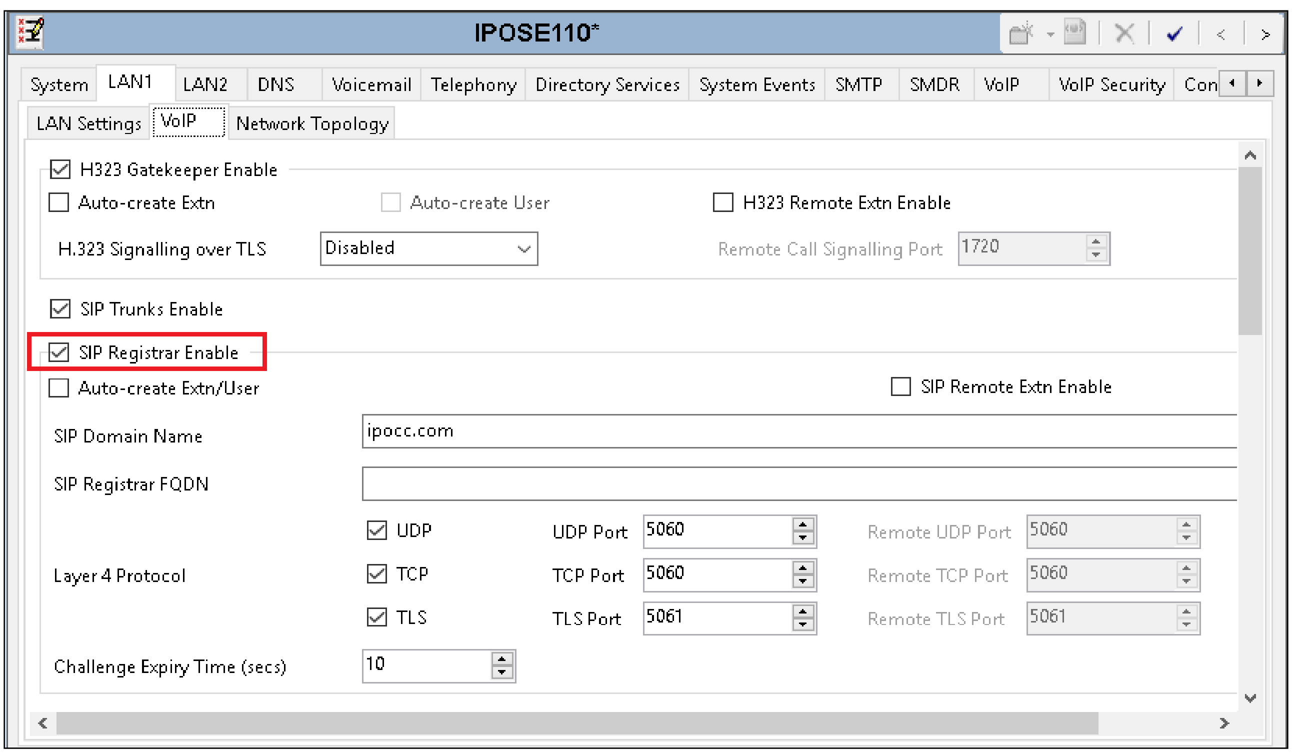
5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **System** screen for the **IPOSE110** in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure the Cetis SIP endpoints.

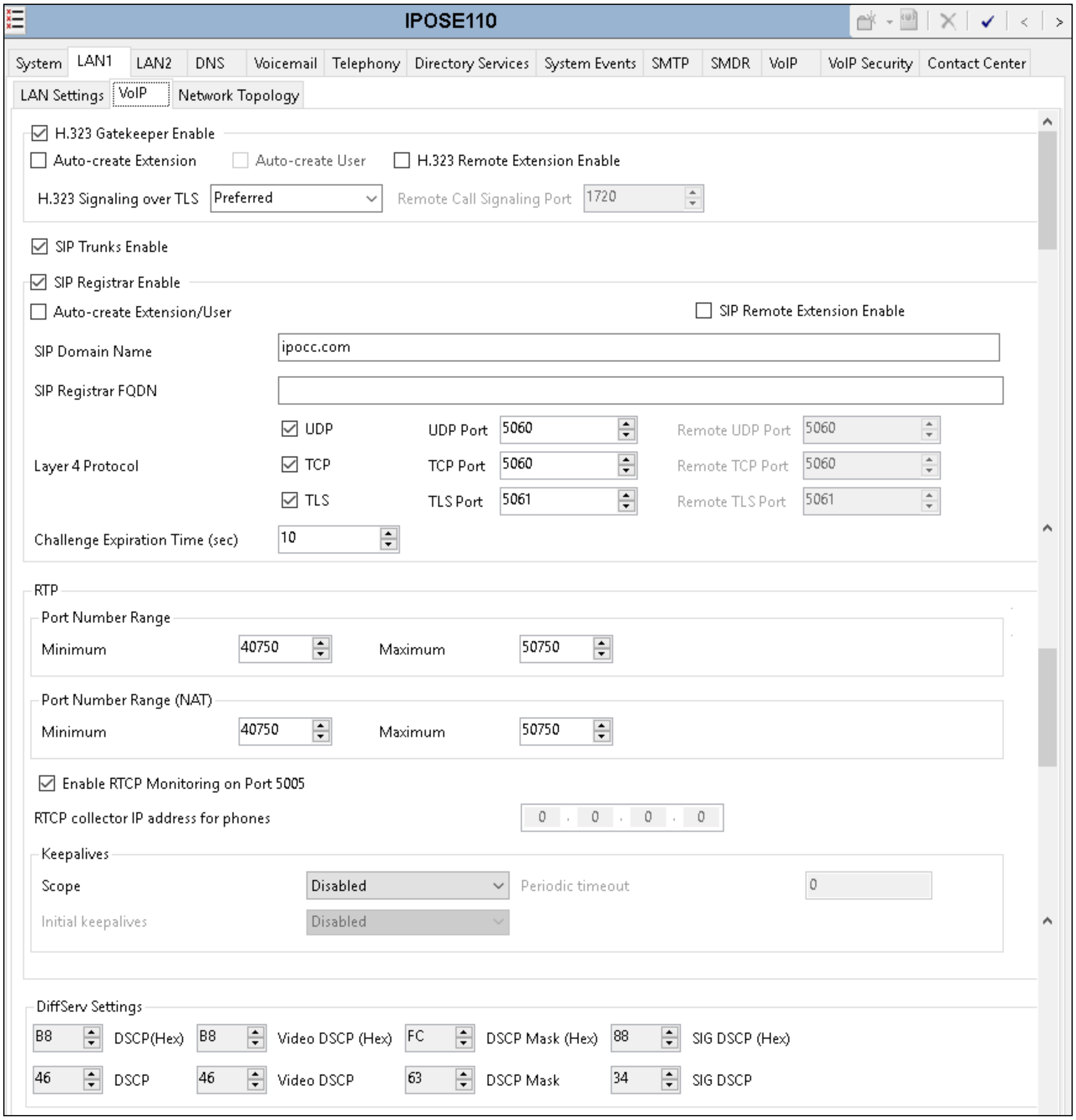


5.3. Enable SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** is checked as shown below. Define the port to be used for the signaling transport, in the test environment **TCP**, **UDP** and **TLS** were used, and the port number was left at the default value.

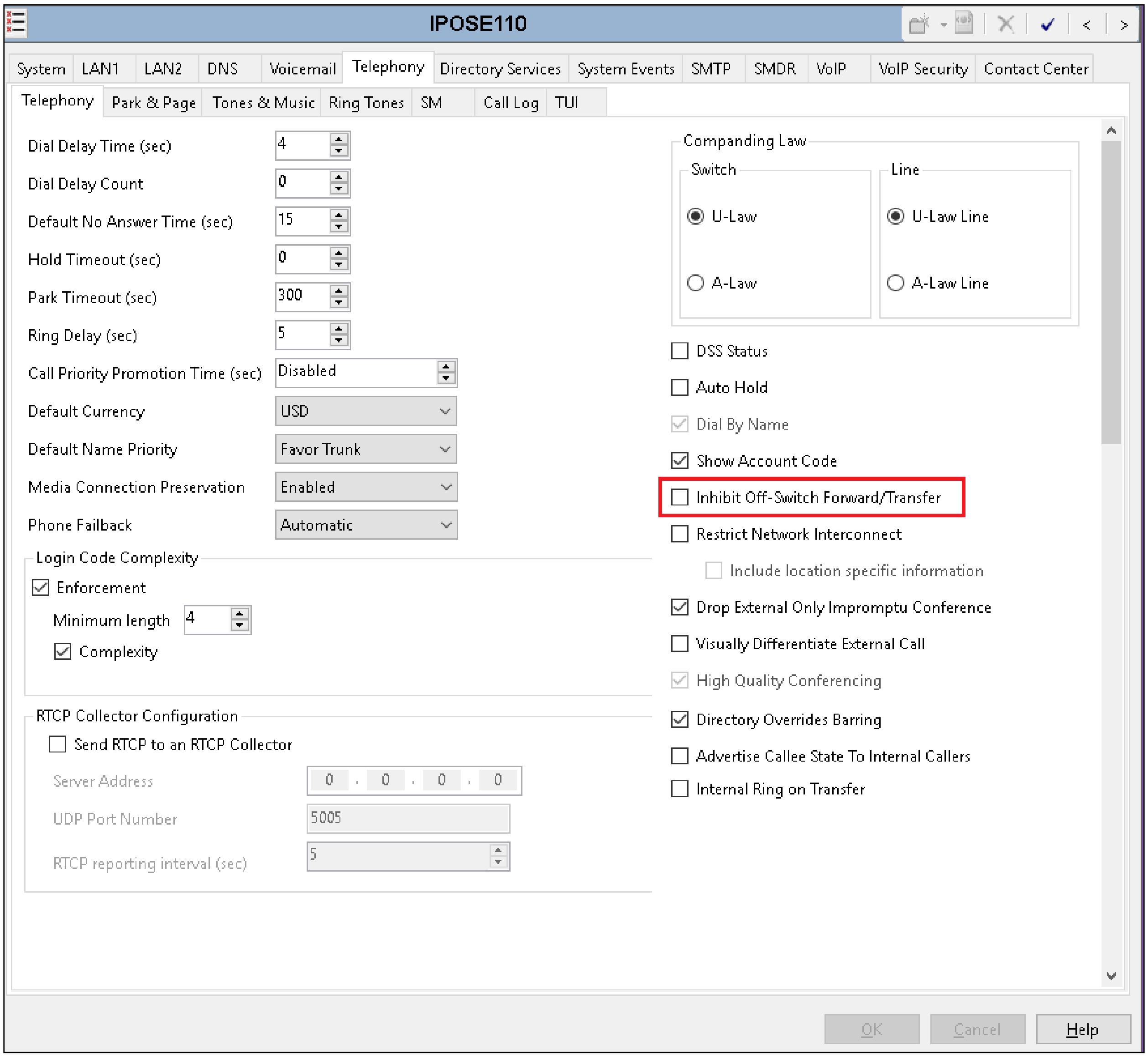


Scroll down for further configuration. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office requests RTP media to be sent to a UDP port in the configurable range for calls using LAN1. The range used for testing was the Linux default setting of **40750** to **50750**.



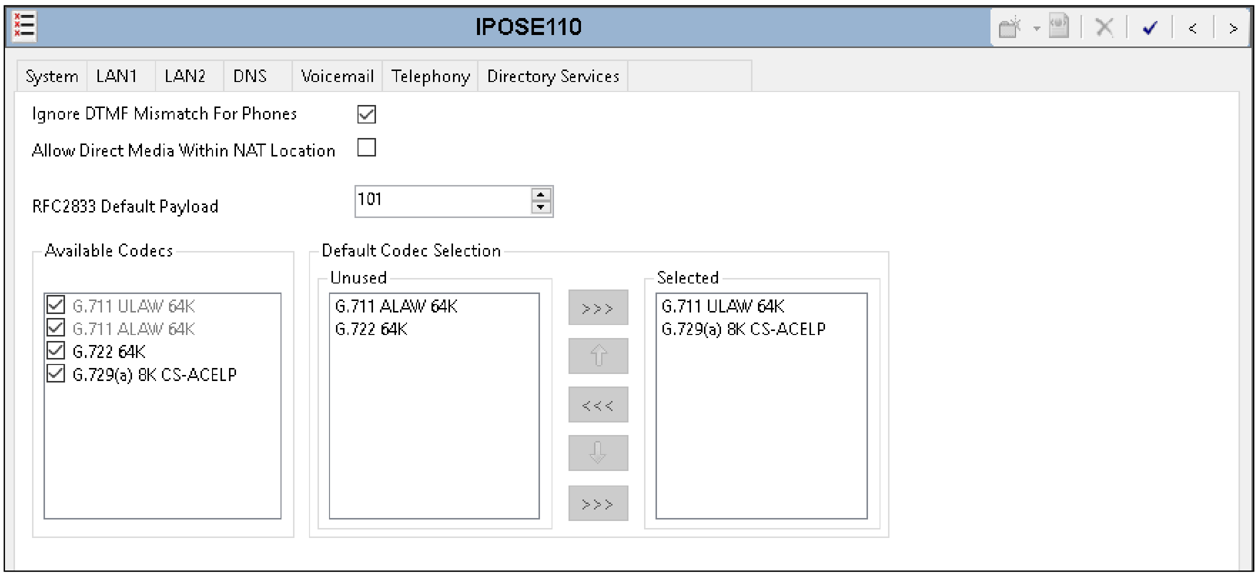
5.4. System Telephony Settings

Navigate to the **Telephony** → **Telephony** tab on 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 transfer to the PSTN. On completion, click the **OK** button (not shown).



5.5. Administer Codec Settings

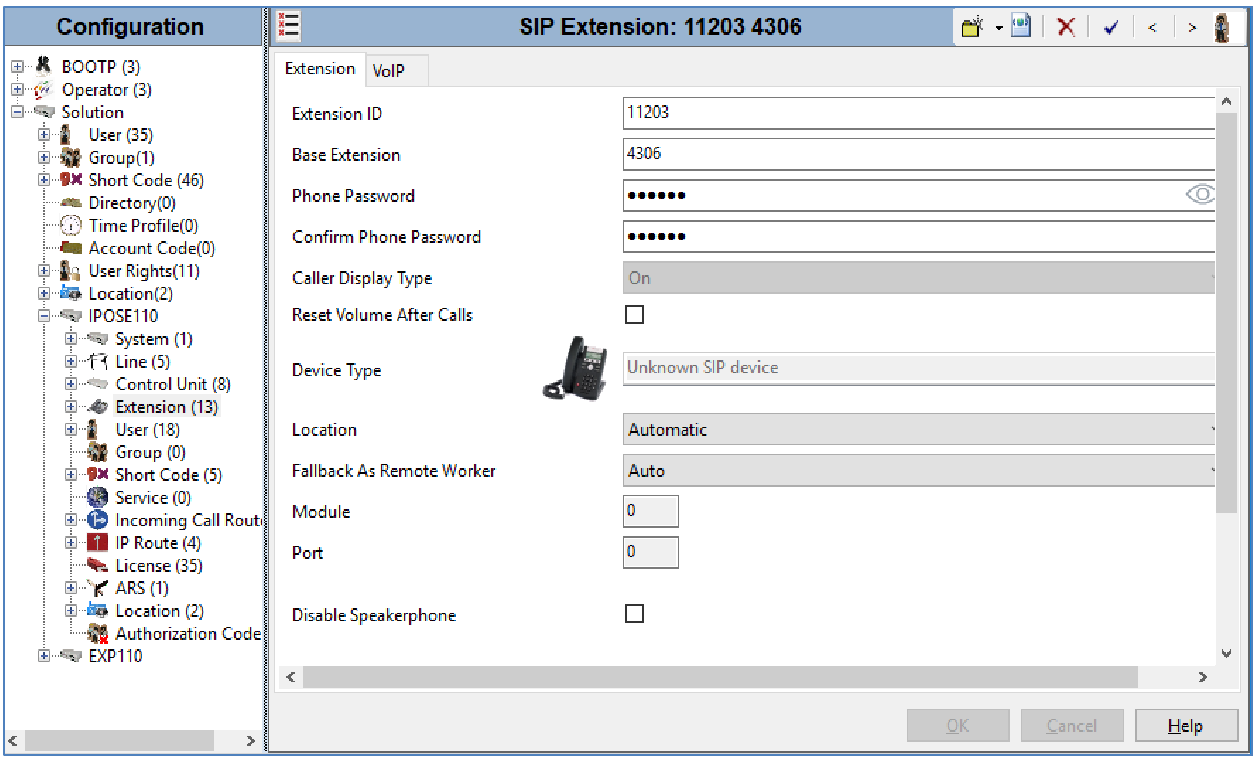
Navigate to the **VoIP** tab on the Details Pane. Check the **Available Codecs** boxes as required for the IP endpoints. Note that **G.711 ULAW 64K** and **G.711 ALAW 64K** are greyed out and always available. Once available codecs are selected, they can be used or unused by using the horizontal arrows as required. Note that in test **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** were used as the default codecs. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).



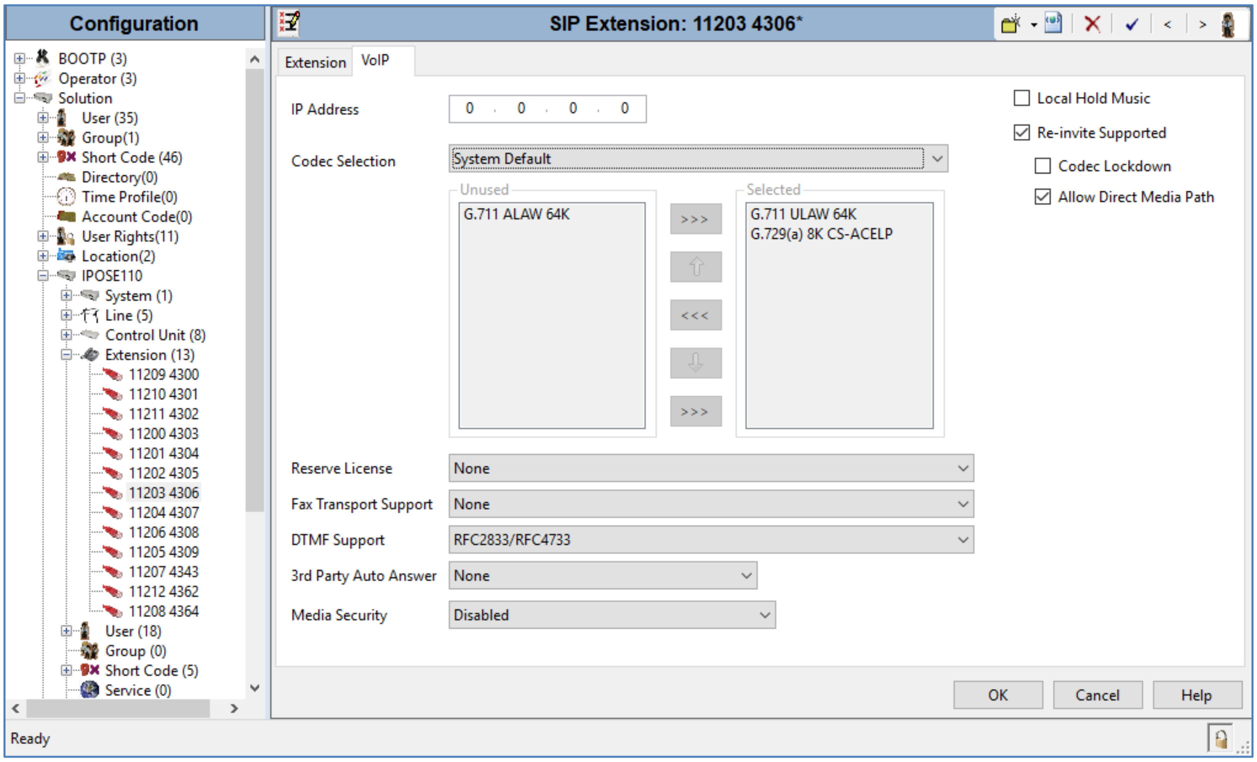
5.6. Administer Extension for Cetis SIP Endpoint

From the configuration tree in the left pane, right-click on **Extension** and select **New SIP** (not shown) from the pop-up list to add a new SIP extension. Enter the desired extension for the **Base Extension** field, a password in **Phone Password** and **Confirm Phone Password** fields as shown below.

Note that this is the password that Cetis SIP phone will be used to register to IP Office, if the **Phone Password** field is left blank, the login code in the **Telephony** (**Supervisor Settings**) tab of SIP User will be used to register to IP Office. The Phone Password is more secure than the login code because they combine number and character while the login code accepts only the number.



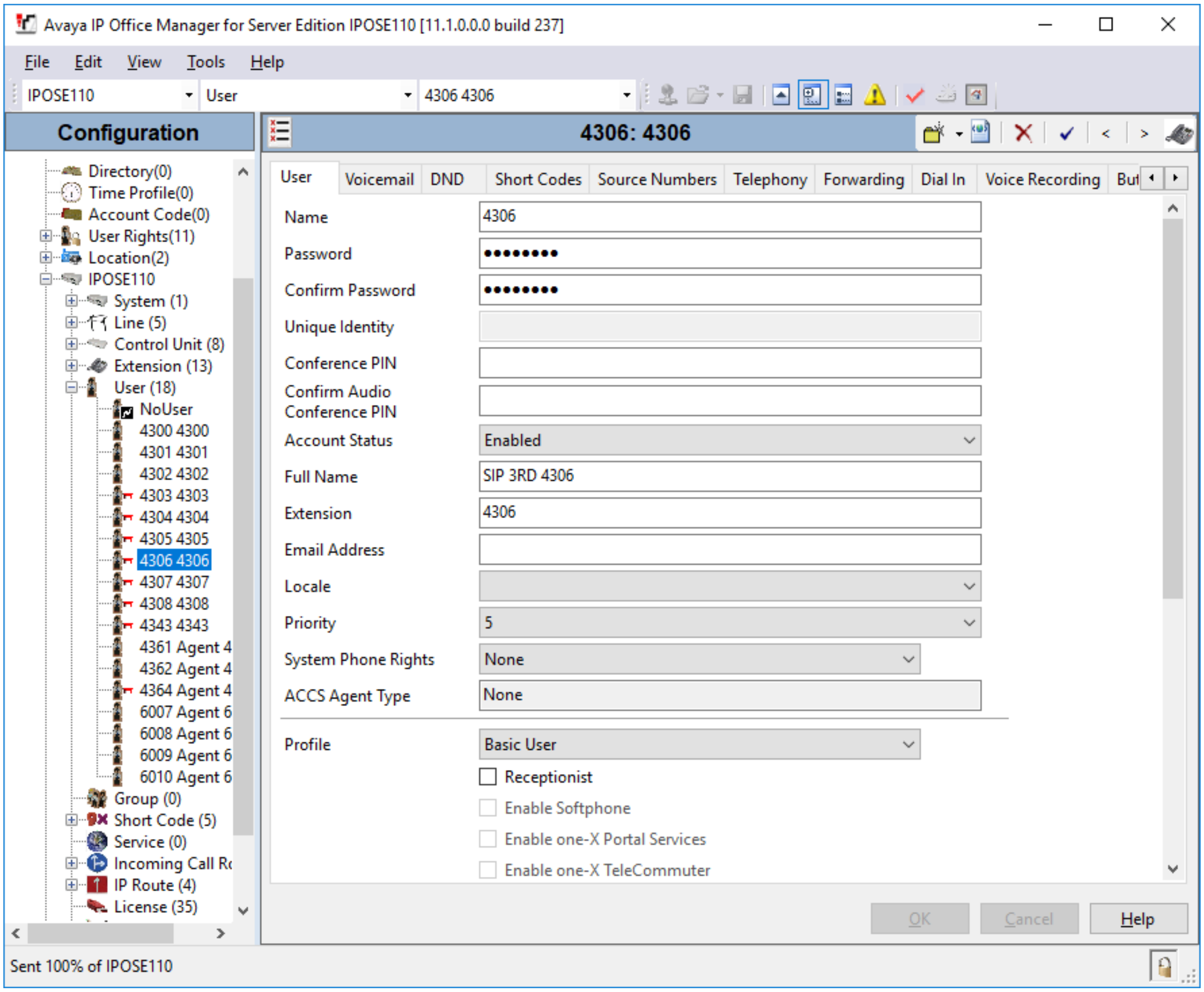
Select the **VoIP** tab and retain the default values in the all fields. During the compliance test, Cetis SIP endpoint was tested with G.711 and G.729 codecs. Enable **Allow Direct Media Path** so that audio/RTP flows directly between two SIP endpoints without using media resources in Avaya IP Office Server Edition. Note that **Media Security** should be set to "Disabled" for Cetis SIP endpoint that does not support the media security.



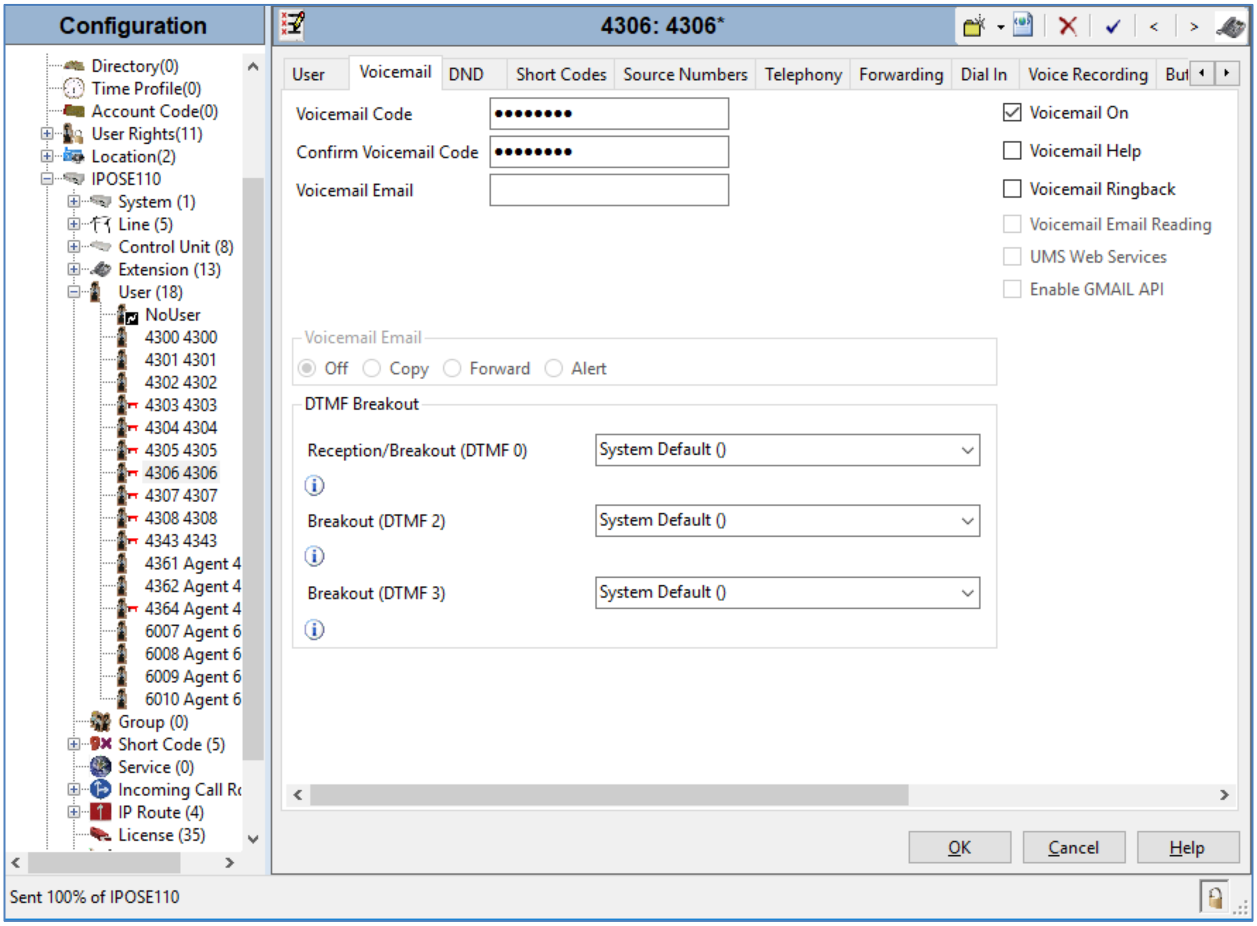
Repeat the same procedure to create another extension 4307 for second Cetis SIP phone.

5.7. Administer SIP User for Cetis SIP Endpoint

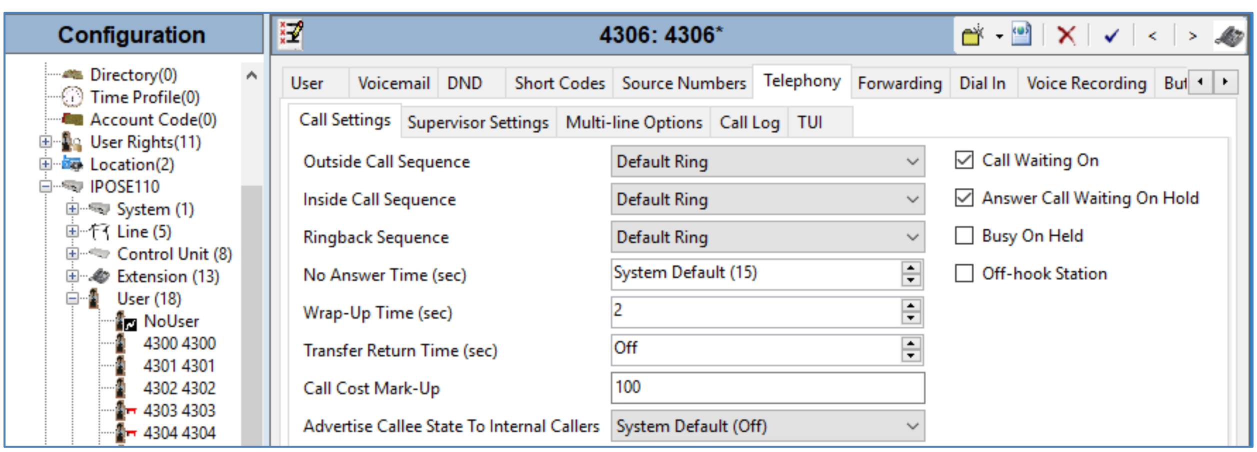
From the configuration tree in the left pane, do a right-click on **User** and select **New** from the pop-up window (now shown). Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension number as created above.



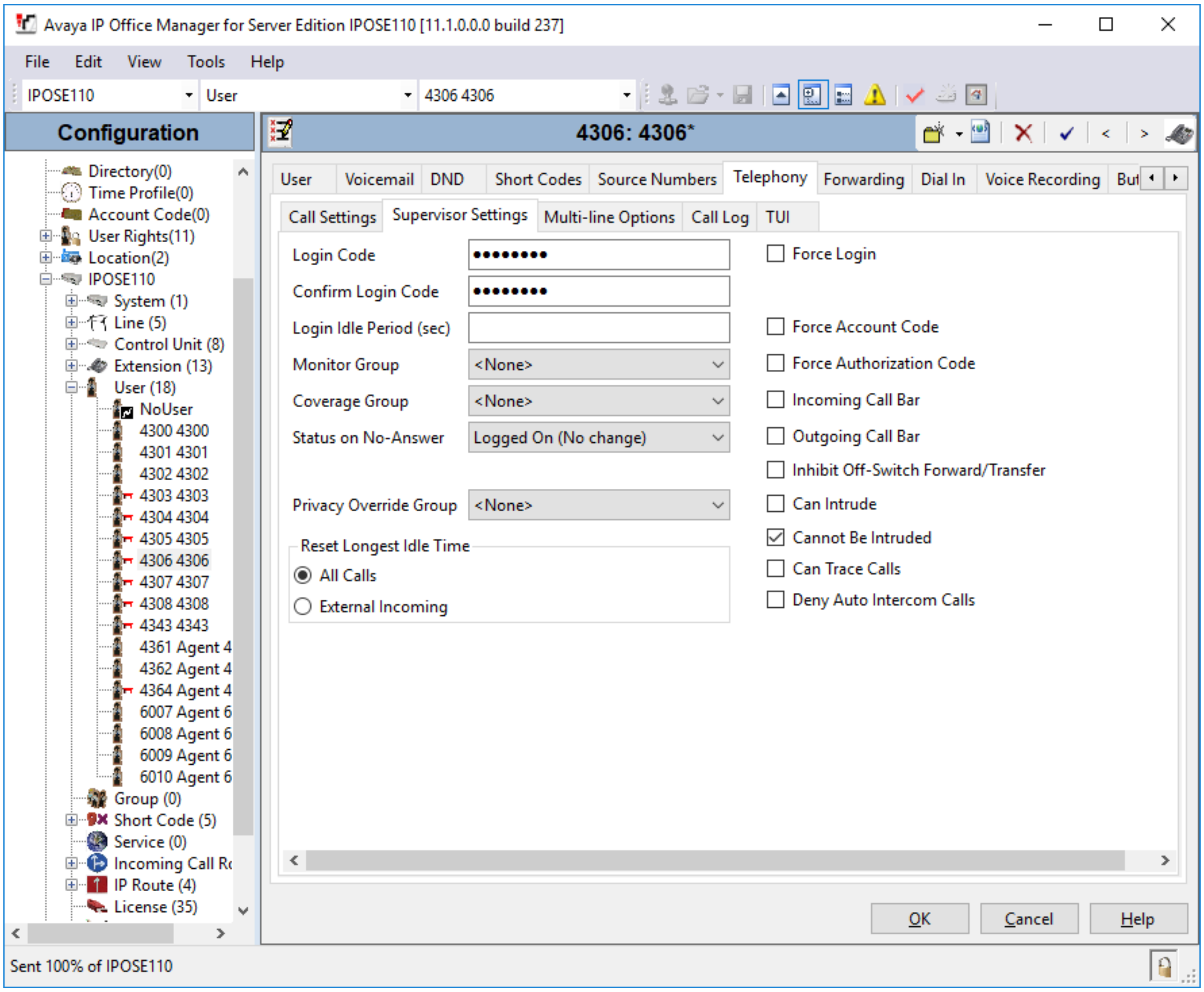
Select the **Voicemail** tab and select **Voicemail On** to enable voicemail and enter a passcode in the **Voicemail Code** and **Confirm Voicemail Code** field.



Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note that **Call Waiting On** is required to allow a secondary incoming call to Cetis SIP endpoint; otherwise, a second incoming call would be denied.

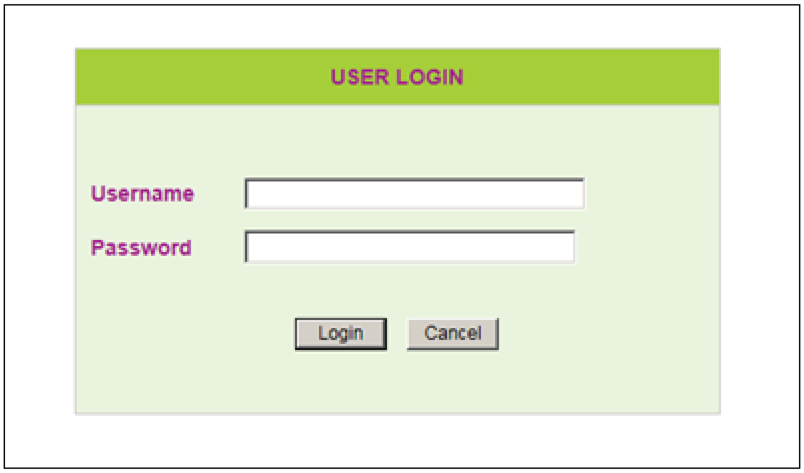


Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by Cetis SIP endpoint to register with IP Office Server Edition.



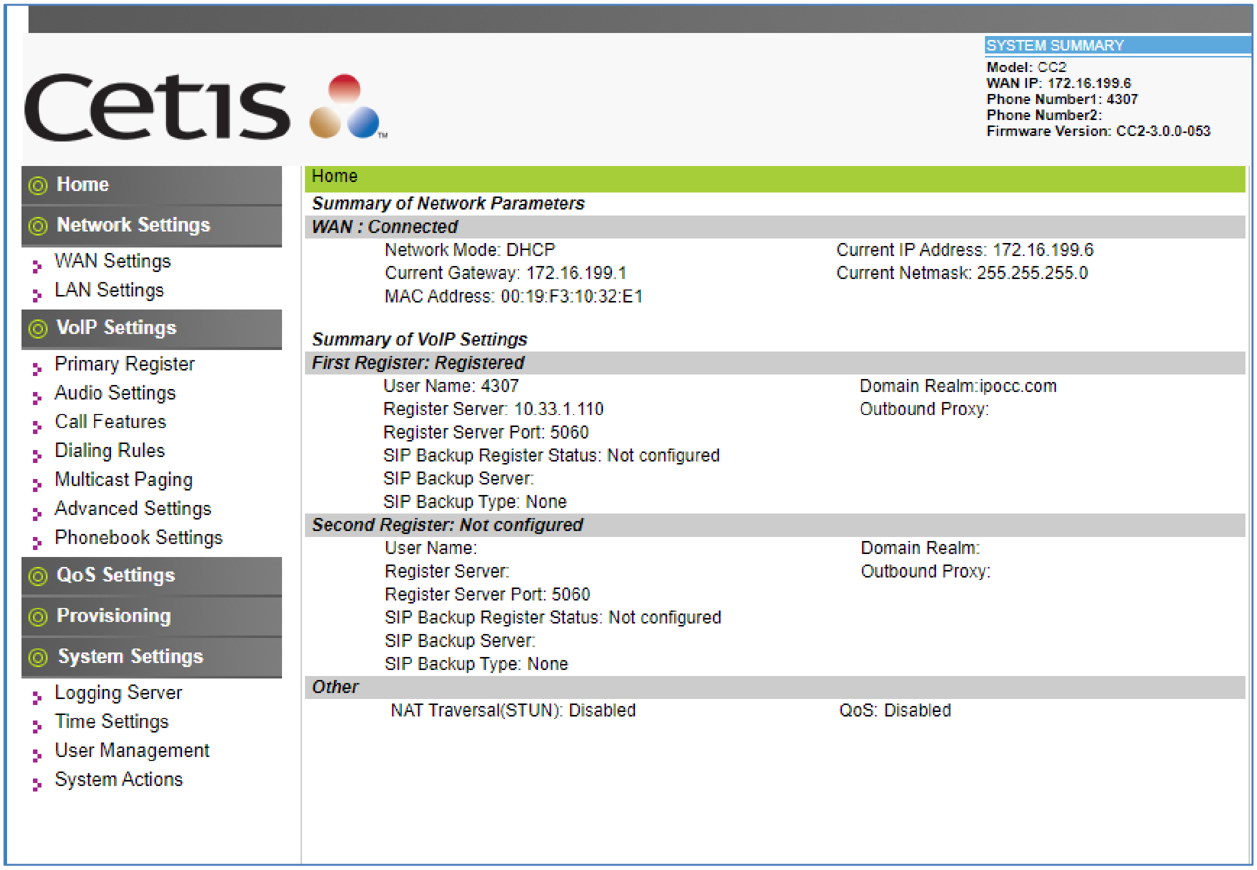
6. Configure Cetis SIP Telephones

Access the Cetis SIP telephones web interface using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the Cetis telephone. By default, DHCP is enabled on the Cetis telephones. For this compliance test, a dynamic IP address was assigned to the Cetis telephone. To determine the IP address assigned to the Cetis telephone, enter **47# on the telephone to hear the IP address.



6.1. Network Settings

To view the network configuration, select the **WAN Settings** under the **Network Settings** section.



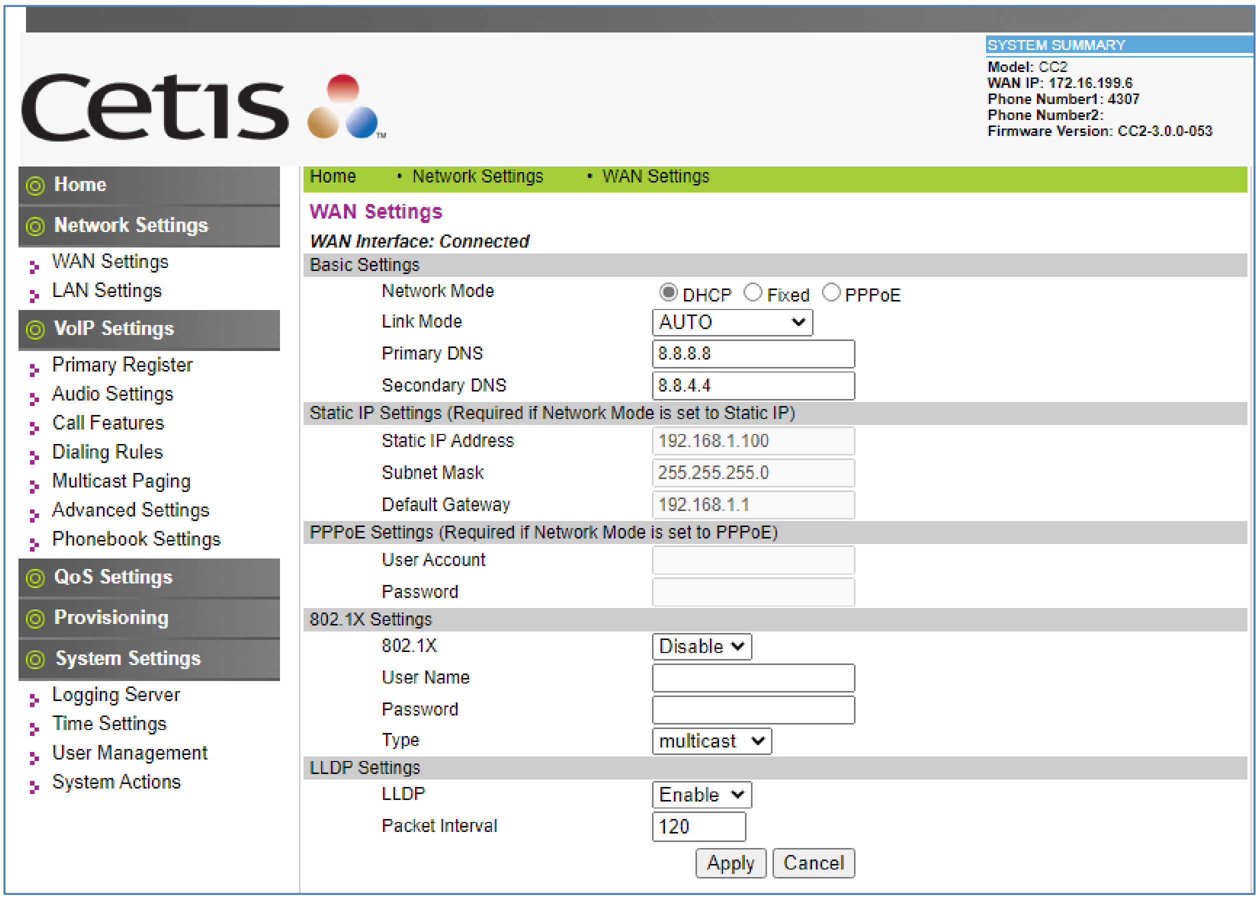
***Note*:** Cetis SIP firmware follows a naming convention based on model.

All Cetis IP phones share the same base chipset and firmware, meaning that models using the same number firmware version share the same traits and compatibility. Server registrations, SIP messaging, and call control are all the same. The different model prefixed versions are to accommodate variances in single vs. 2-line capability, corded vs. cordless radio handsets and LCD display screen sizes. Example: C32-3.0.0-050.bin is the firmware for Cetis Corded 2-line models including 3500IP and 9700IP.

In the **WAN Settings** page, provide the following information:

- **Basic Settings**
- **Static IP Settings**
- **PPPoE Settings**
- **802.1X Settings**
- **LLDP Settings**

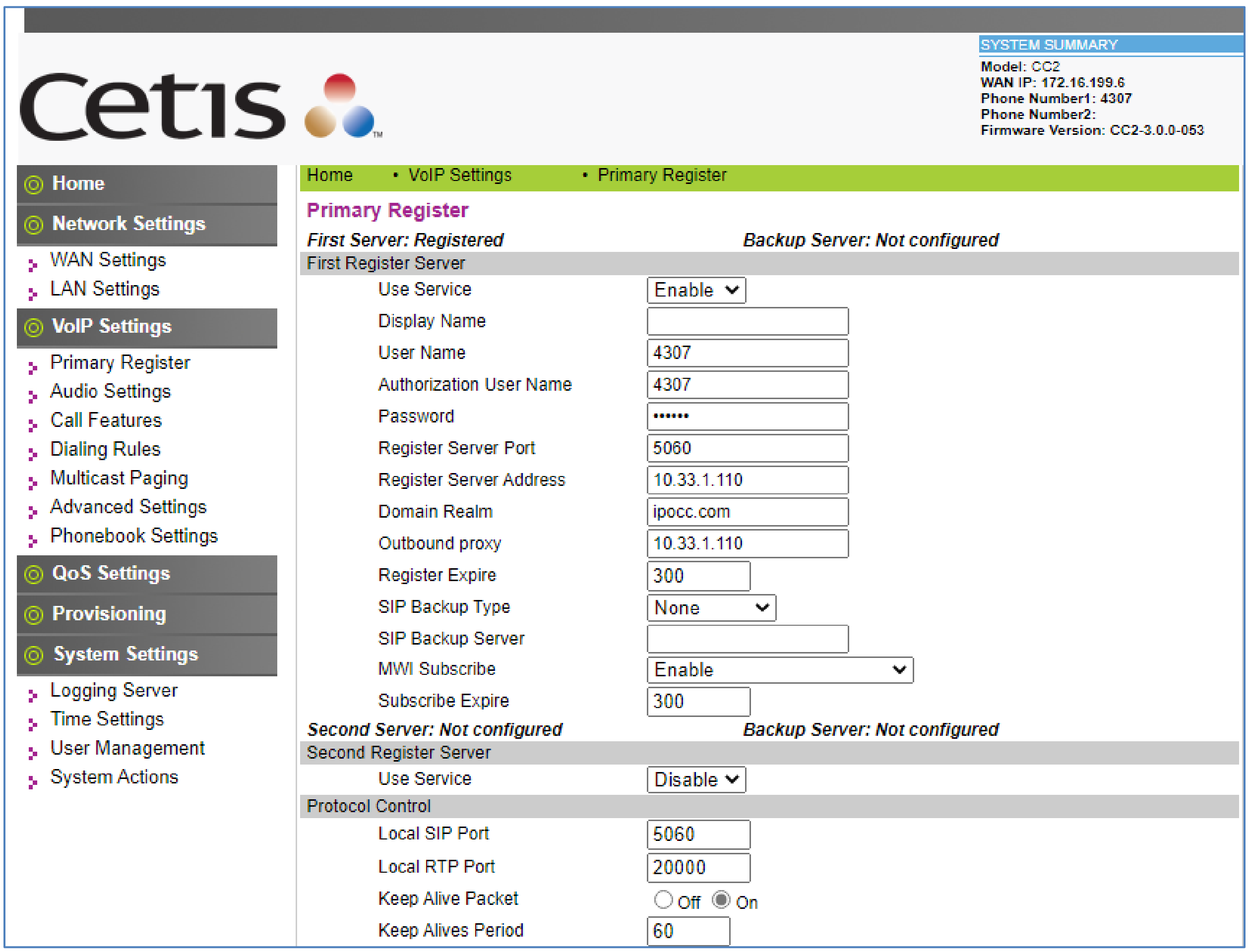
During the compliance test, dynamic IP address was utilized. The following screen show what was configured and used.



6.2. VoIP Settings

Select **Primary Register** under the **VoIP Settings** section. In the **Register Server** section, provide the following information:

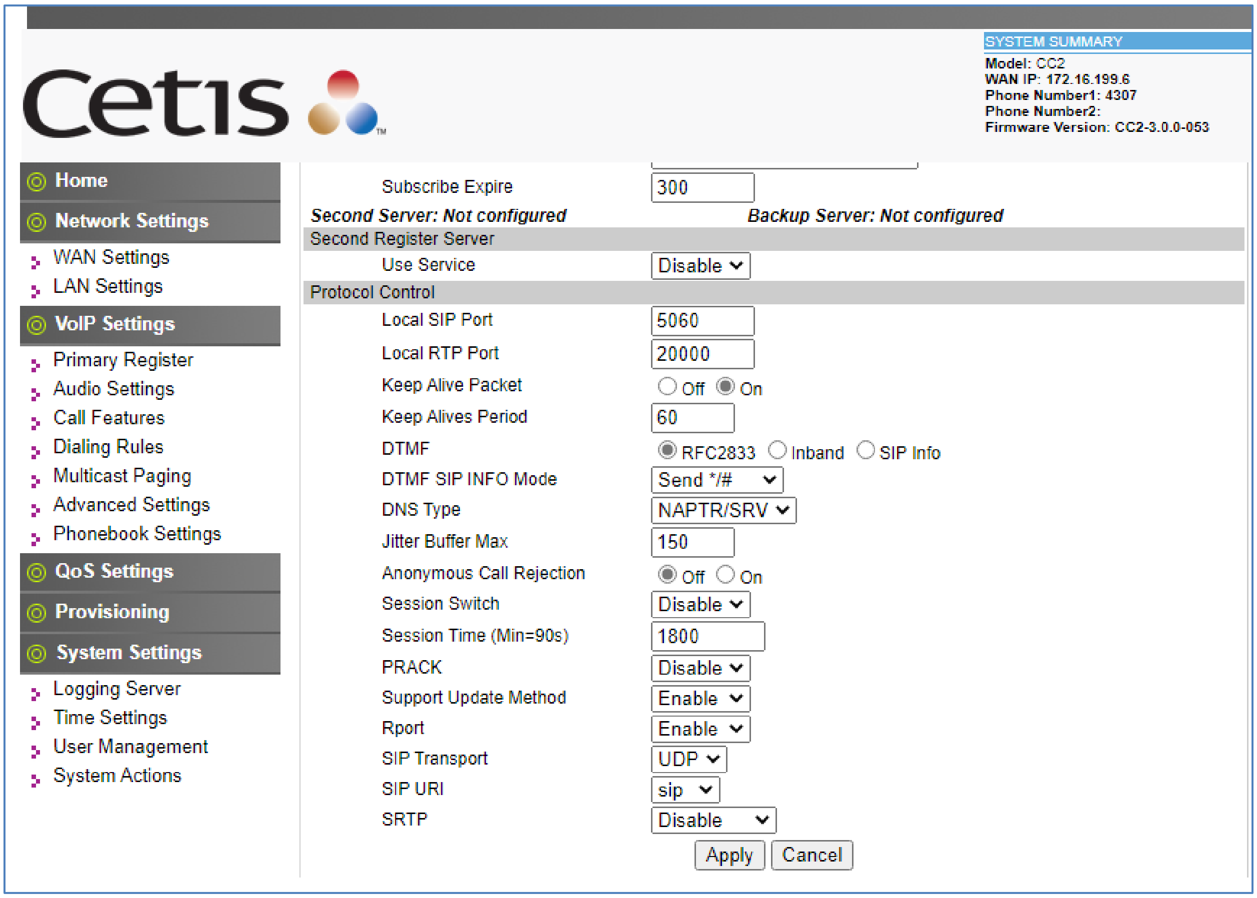
- **Use Service** – Select **Enable**.
- **Display Name** – Enter a descriptive name.
- **Register Server Address** – Enter the LAN1 IP address of IP Office.
- **Register Server Port** – Enter **5060** for UDP.
- **User Name -** Enter the user name created in **Section 5.7**.
- **Authorization User Name -** Enter the user name as configured in **Section 5.7**.
- **Password -** Enter the password created in **Section 5.6**.
- **Domain Realm** – Used **ipocc.com** during the test.
- Leave other fields at default value.



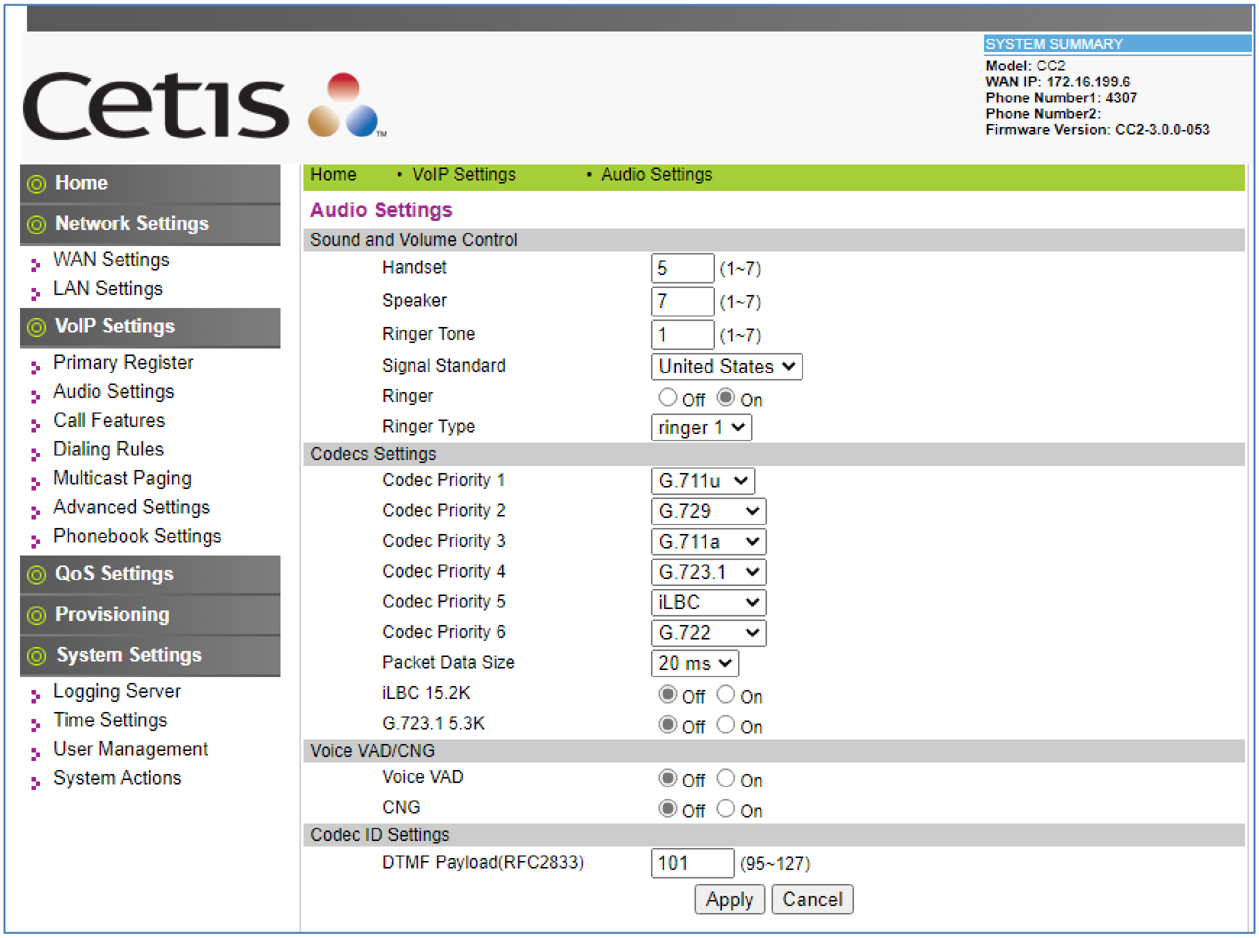
In the **Protocol Control** section, provide the folllowing values.

- **DTMF** – Select the RFC2833 option.
- **SIP Tranport** – Select **UDP** from the dropdown menu.
- Leave other fields at default value.

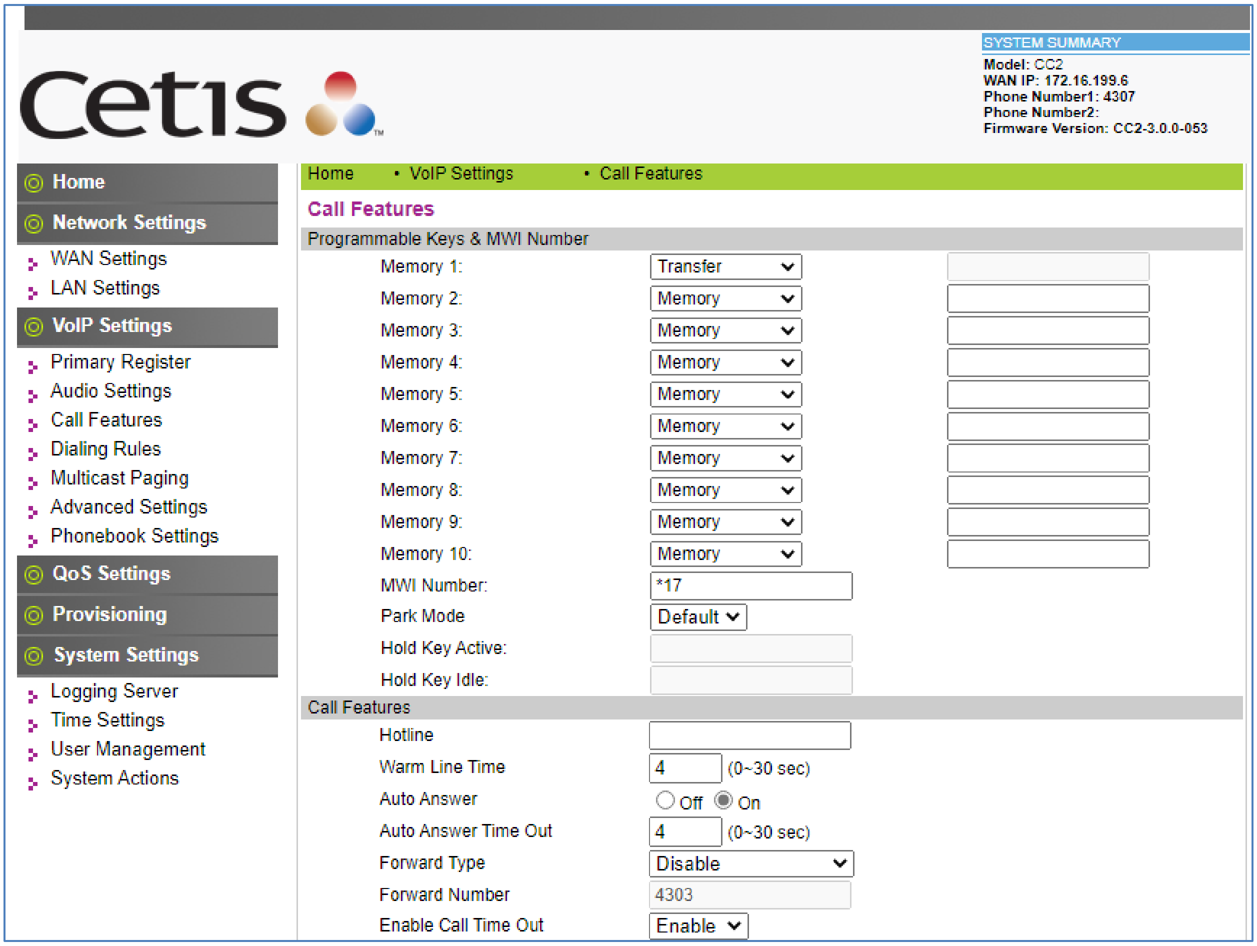
Click **Apply** button to save the changes.



Select **Audio Settings** under the **VoIP Settings** section. In this page, a customer can prioritize codec settings. The picture below shows the list of codecs used for the compliance test.

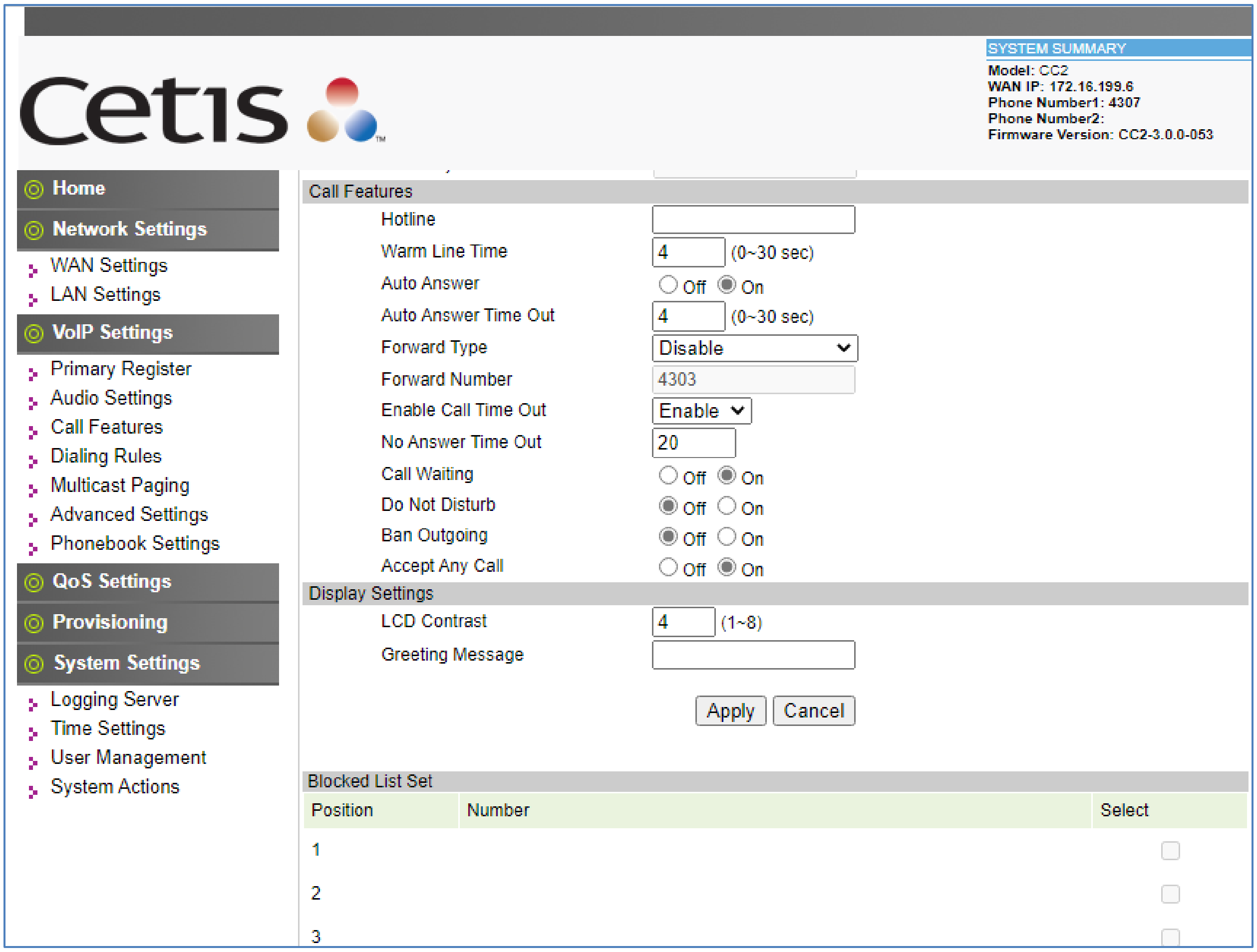


Select **Call Features** under the **VoIP Settings** section. In this page, a customer can program the memory buttons. For Cetis SIP phone comes with 10 memory buttons. Enter the voicemail short code of IP Office in the **MWI Number** box this setting allows user to access to the voicemail system by press **Message** button the phone.



Under the **Call Features** section in the right pane, two features (Auto Answer, Do Not Disturb and Call Forward) are tested.

After the configuration is completed, click **Apply**.

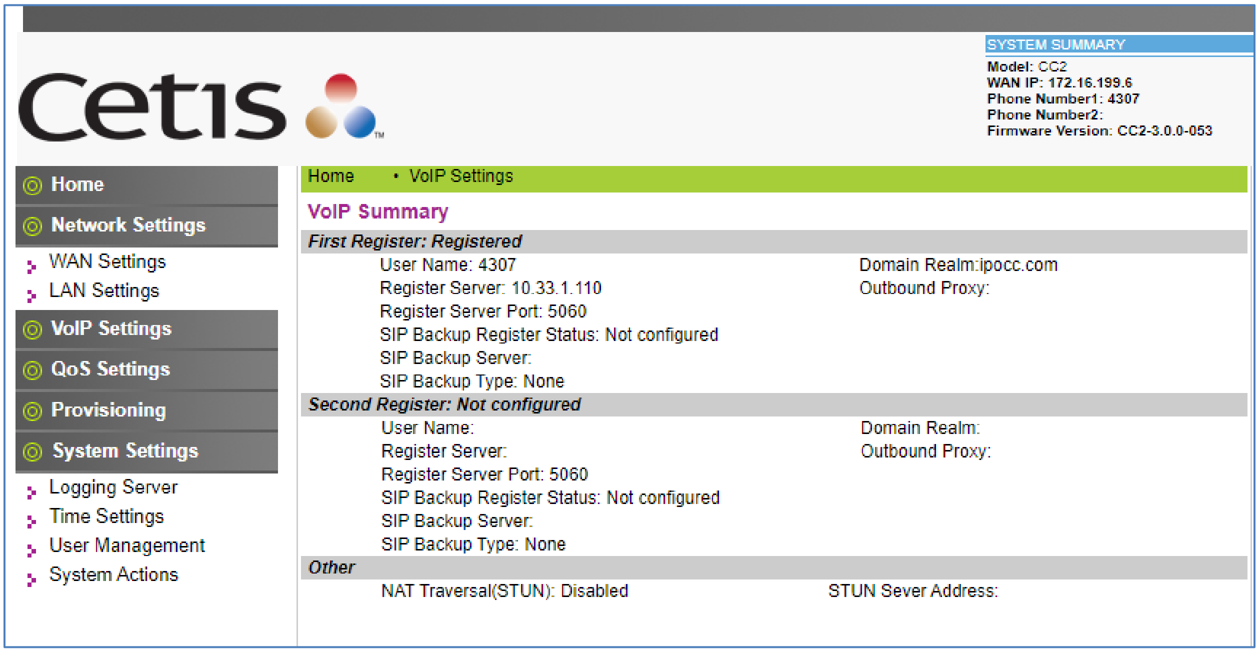


7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of IP Office and Cetis SIP Telephones.

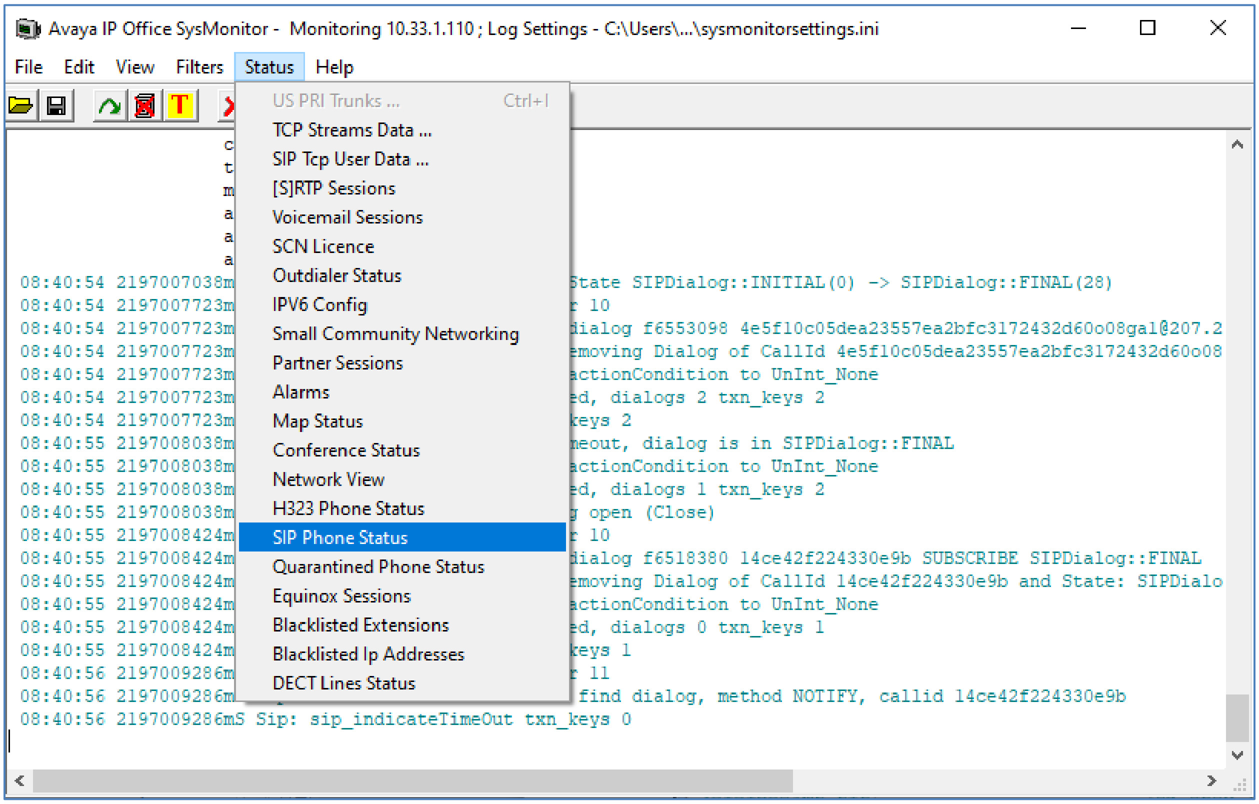
7.1. Verify Cetis SIP Telephones

Select **VOIP Settings** in the left pane to display the **VoIP Summary** page. Verify that the **Primary Register** is set to *Registered*.

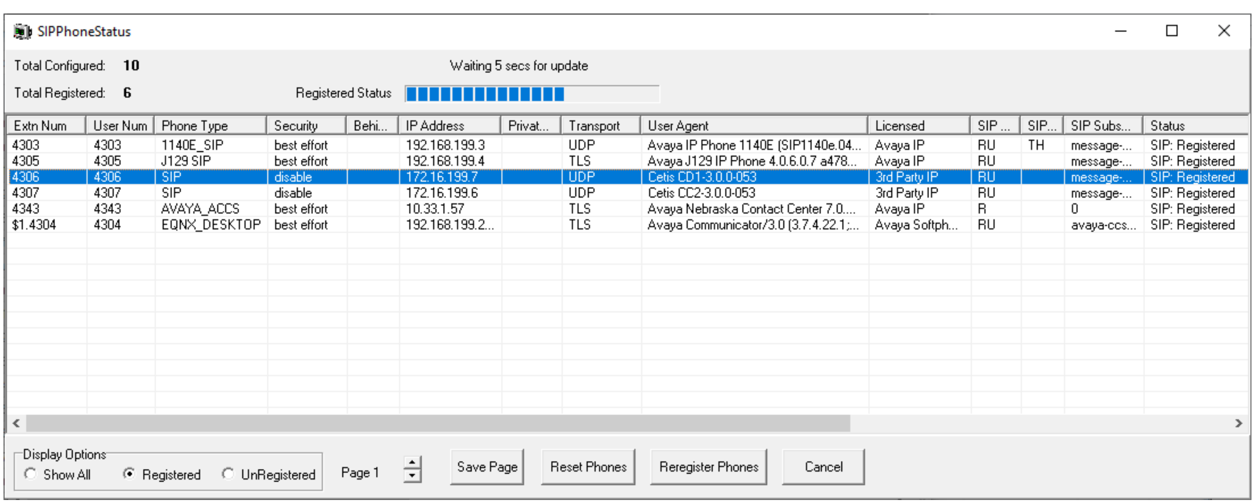


7.2. Verify Avaya IP Office

From a PC running the Avaya IP Office Monitor application, select **Start → Programs→ IP Office → System Monitor** to launch the application. The **Avaya IP Office SysMonitor** screen is displayed, as shown below. Select **Status → SIP Phone Status** from the top menu.



The **SIPPhoneStatus** screen is displayed and select the **Registered** radio button in the **Display Options** area it displays all SIP users currently register to IP Office. Verify that there is an entry for the Cetis C32.3.0.0.53 in the list.



8. Conclusion

These Application Notes have described the administration steps required to integrate the Cetis 3500IP Series and 9700IP Series SIP telephones SIP with Avaya IP Office Server Edition. The Cetis SIP telephones registered successfully with Avaya IP Office via SIP. Incoming and outgoing calls were placed to/from the Cetis SIP telephones and basic telephony and hospitality features were exercised. All test cases passed with observations noted in **Section 2.2**.

9. References

This section references the documentation relevant to these Application Notes. Product documentation for Avaya IP Office, including the following, is available at:
http://support.avaya.com/

[1] *Deploying IP Office Server Edition*, Release 11.1, Issue 14, April 2020.
[2] *IP Office Platform 11.0, Deploying Avaya IP Office Servers as Virtual Machines,* 15-601011 Issue 07d, June 9, 2020.
[3] *IP Office Platform 11.0, Deploying Avaya IP Office Essential Edition (IP500 V2)*, 15-601042, Issue 35f, January 2020.
[4] *Administering Avaya IP Office Platform with Manager, Release 11.1 Issue 2,* May 2020.
[5] *Administering Avaya IP Office™ Platform with Web Manager*, *Release 11.1 Issue 2*, May 2020.
[6] *Planning for and Administering Avaya IX*™ **Workplace Client** *for Android, iOS, Mac and Windows, Issue 1, Release 3.9, June 2020.*
[7] *Using Avaya IX*™ ***Workplace Client*** *for IP Office, Release 11.1 Issue 9, June 2020.*

Additional Avaya IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

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.