Avaya Solution & Interoperability Test Lab

Application Notes for Cetis 3300IP Series and 9600IP Series SIP Telephones Version 3.0.0.40 with Avaya IP Office Server Edition Release 10.1 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Cetis 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office. The Cetis 3300IP Series and 9600IP Series are corded and cordless telephones that were designed for the hospitality industry and register with Avaya Aura® Session Manager.

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 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetis 3300IP Series and 9600IP Series SIP Telephones were designed for the hospitality industry. In the compliance test, Cetis SIP telephones registered with Avaya IP Office.

In the compliance testing, Avaya IP Office Server Edition system consisted of Avaya IP Office Primary Linux running on Virtualized Environment and a IP Office 500V2 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 would be noted in **Section 2.2**.

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 these Application Notes, the interface between Avaya systems and the Cetis 3300IP Series and 9600IP Series SIP Telephones do 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 3300IP Series and 9600IP Series SIP Telephones and Avaya SIP and H.323 telephones, and exercising basic telephony features, such as hold, mute, hold, 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 3200IP SIP Telephones with IP Office Server Edition.
- 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 UDP.
- Proper recognition of DTMF tones.
- Basic telephony features, including hospitality feature, inbound/outbound, hold, mute, 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 3300IP Series and 9600IP 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 observations noted:

- There is an issue with the blind transfer when Cetis 2-Line SIP telephone calls to Avaya SIP endpoint in the IPO Primary and Avaya SIP endpoint make a blind transfer to an Avaya H.323 endpoint in Expansion, after the transfer is completed there is no audio from both endpoints. This issue is currently under investigation by Cetis.

2.3. Support

For technical support on the Cetis 3300IP and 9600IP Telephones, contact Cetis Support via phone, email, or website.

- **Phone:** (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 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetis SIP telephones registered with Avaya IP Office via SIP.

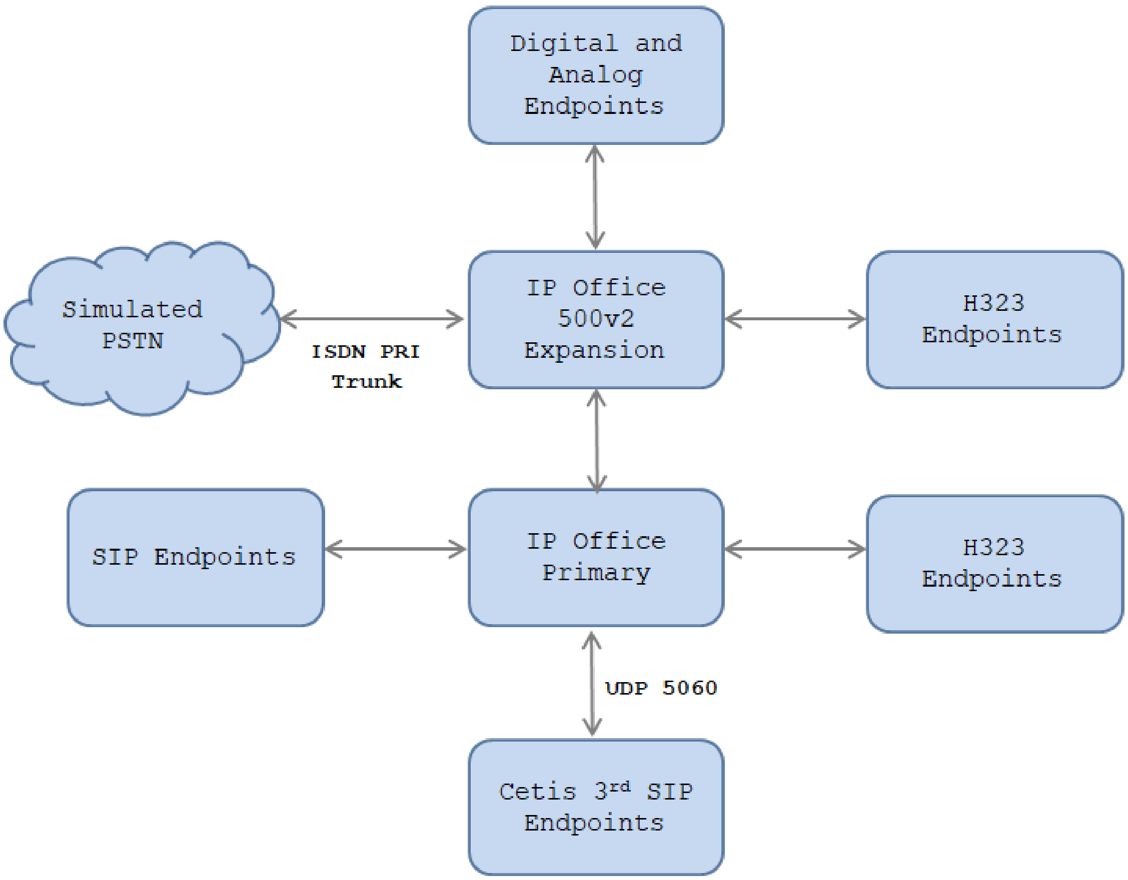


Figure 1: Test Configuration Diagram with IP Office

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

Description	IP Address
IP Office Server Edition Primary	10.10.97.110
IP Office 500v2 Expansion	10.10.97.230
H.323 Endpoints	10.33.5.10-11
SIP Endpoints	10.33.5.12-14
Cetis 3rd-PartySIP Telephones	10.33.5.42-43

4. Equipment and Software Validated

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

Equipment / Software	Release/Version
Avaya IP Office Server Edition Primary Linux running on Virtualized Environment	10.0.1
Avaya IP Office 500V2 Expansion	10.0.1
Avaya IP Office Manager	10.0.1
Avaya 1140E SIP Deskphones	4.04.23
Avaya 96x1 IP Deskphones	6.6229
Cetis 3300IP Series and 9600IP Series SIP Telephones	3.0.0.40

Note: Testing was performed with Avaya IP Office Server Edition Solution that requires an Expansion IP Office 500 V2 to support analog used by fax endpoint. Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2.

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: C32-3.0.0-040.bin is the firmware for **C**etis **C**orded **2**-line models including 3300IP and 9600IP

5. Configure Avaya IP Office

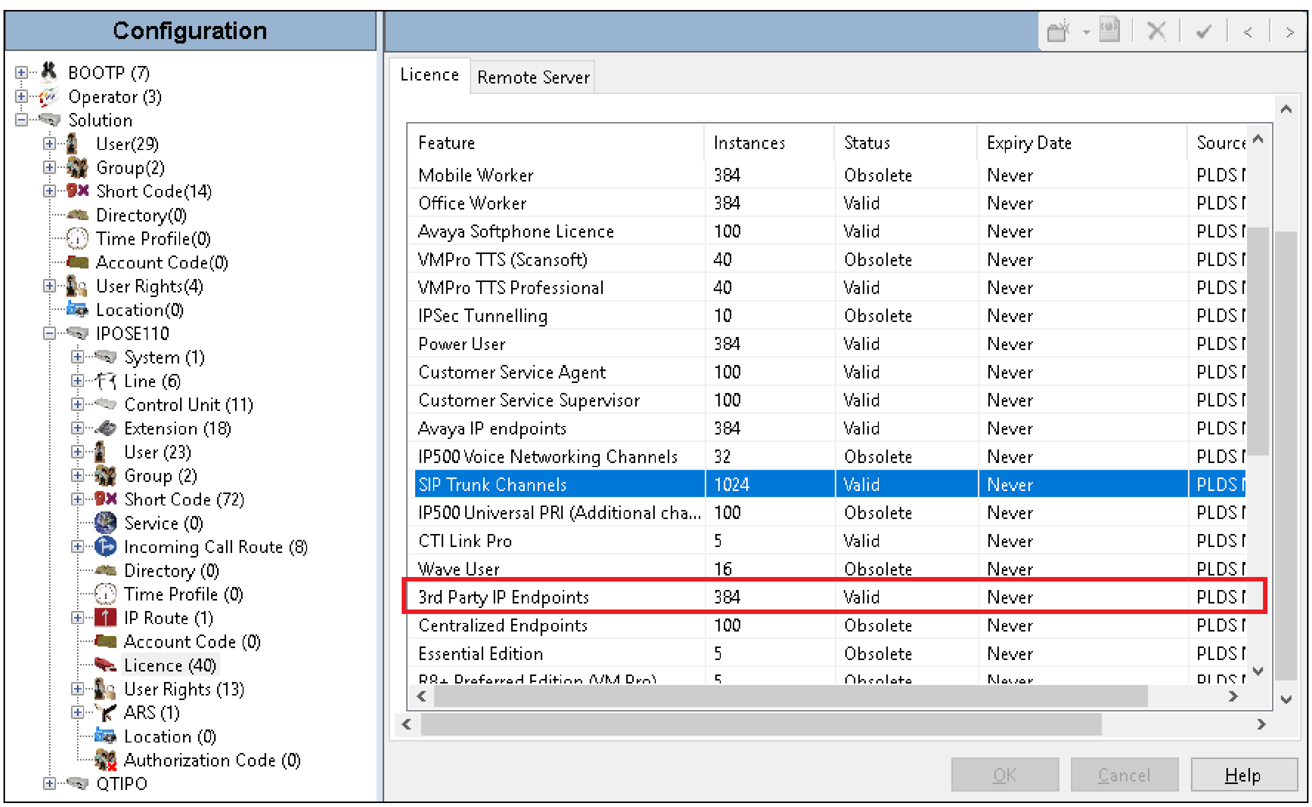
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify Avaya IP Office License
- Obtain LAN IP Address
- Enable SIP Trunks
- Administer SIP Line
- Administer Incoming Call Route
- Administer Short Code
- Administer IP Office Line

5.1. Verify Avaya IP Office License

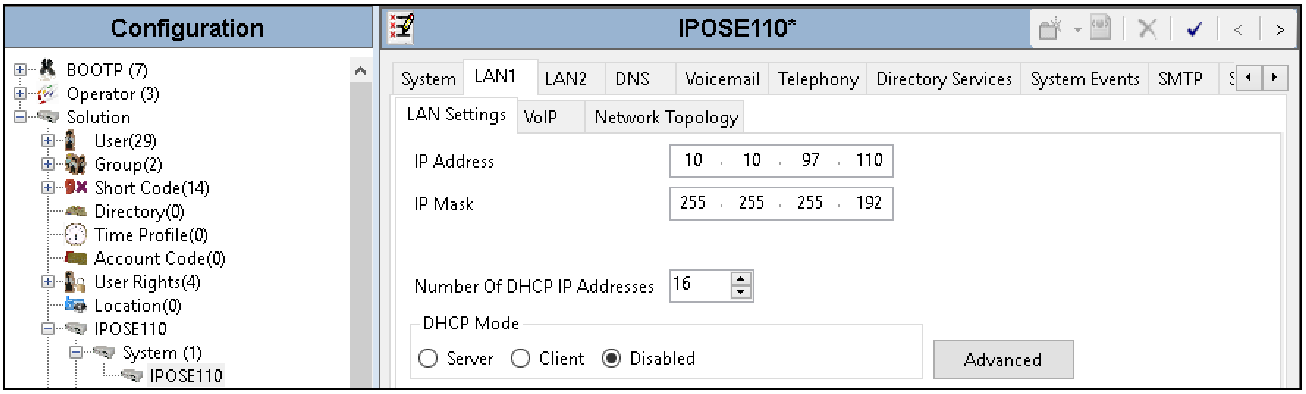
From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the Manager application. Select the correct IP Office system and log in with the appropriate credentials.

The **Avaya IP Office for Server Edition 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.



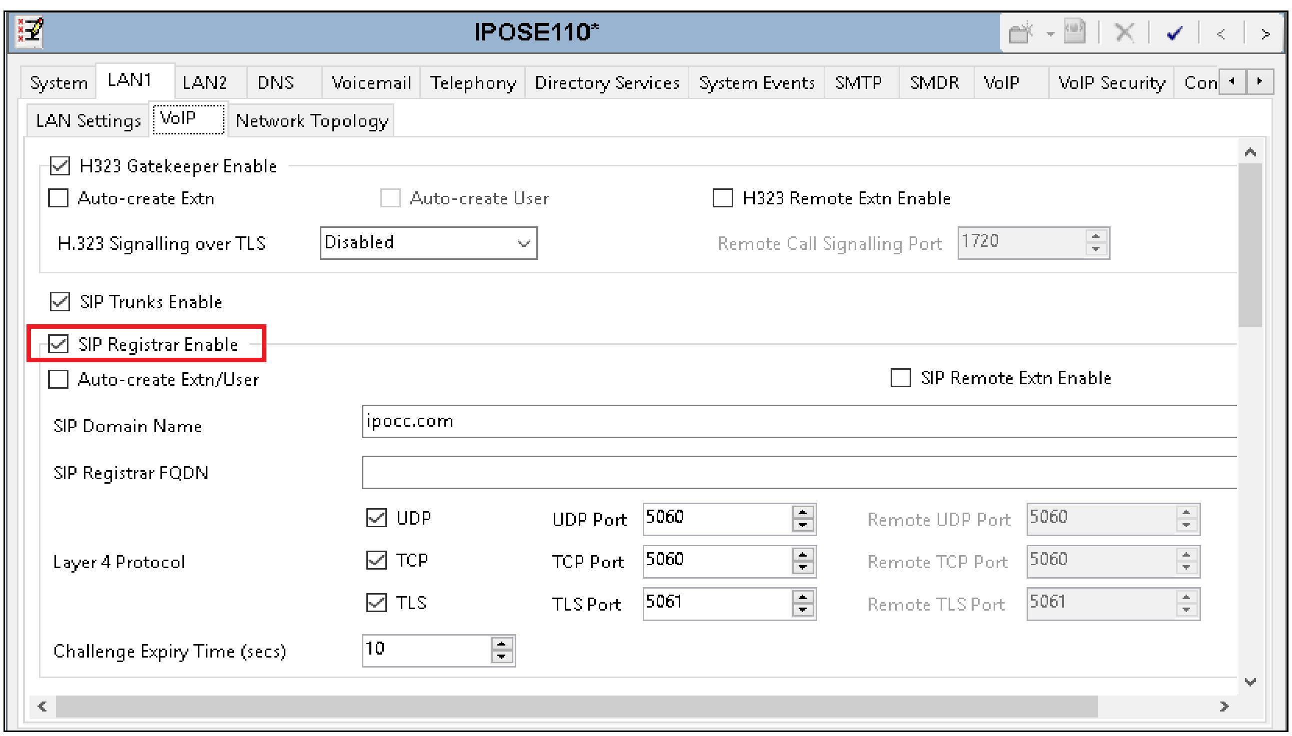
5.2. Obtain LAN IP Address

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 telephone.

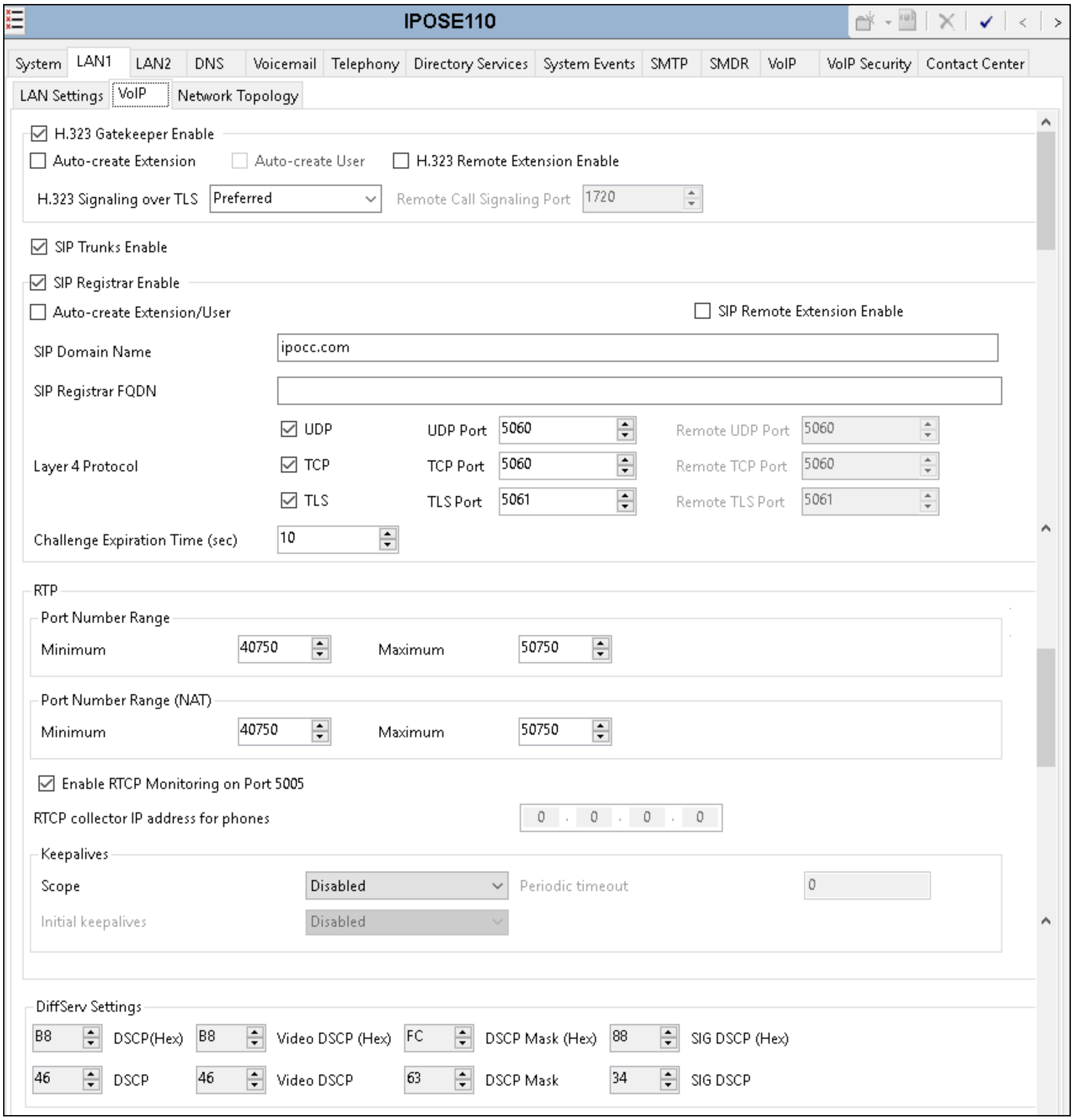


5.3. Enable SIP Registrar

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. Note that Cetis SIP telephone only uses UDP.

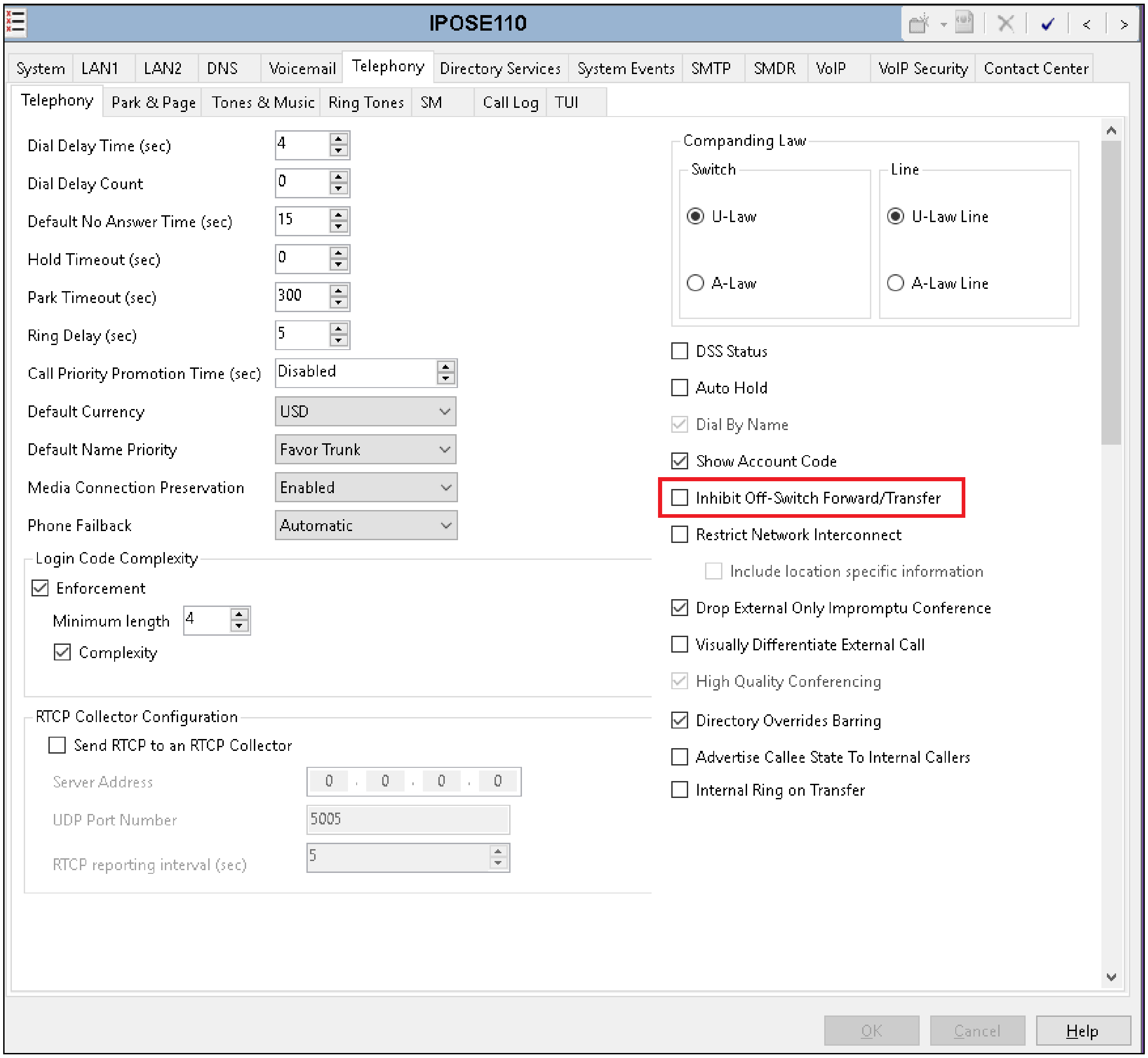


Scroll down for further configuration. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office requests RTP media to be sent to a UDP port in the configurable range for calls using LAN1. The range used for testing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



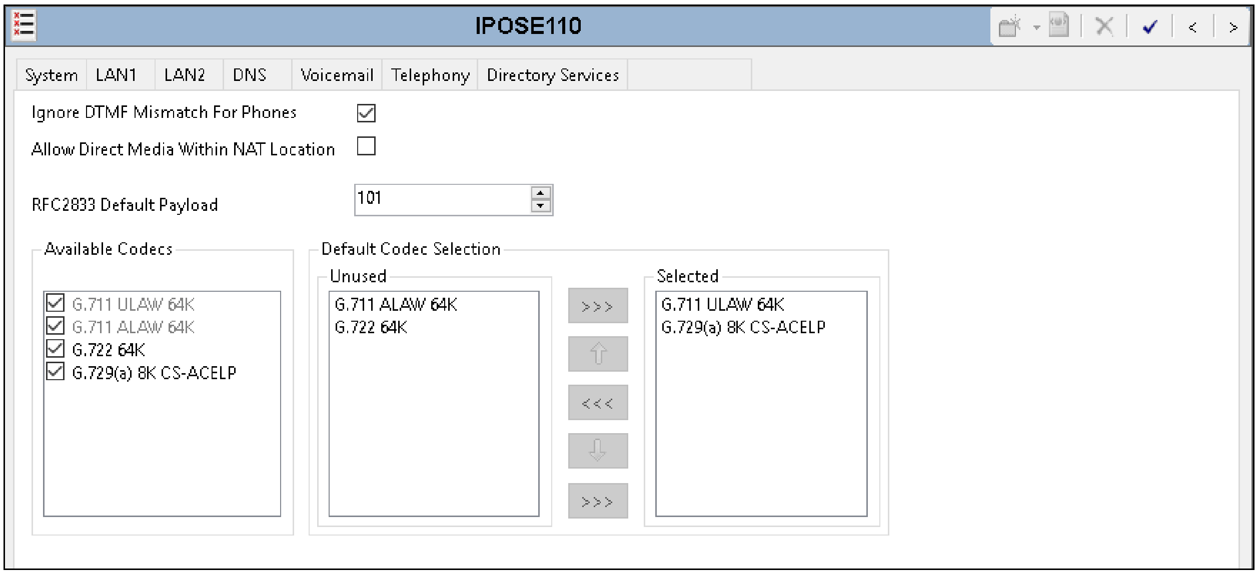
5.4. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Europe, **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).



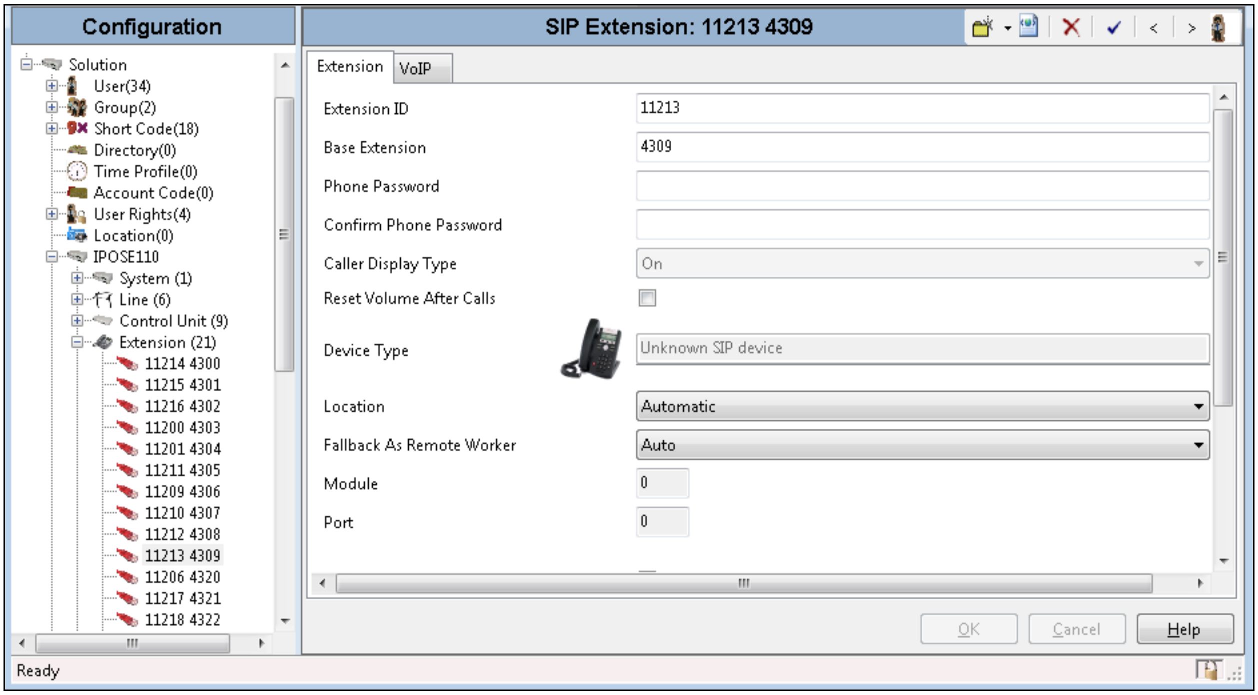
5.5. Administer Codec Settings

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).

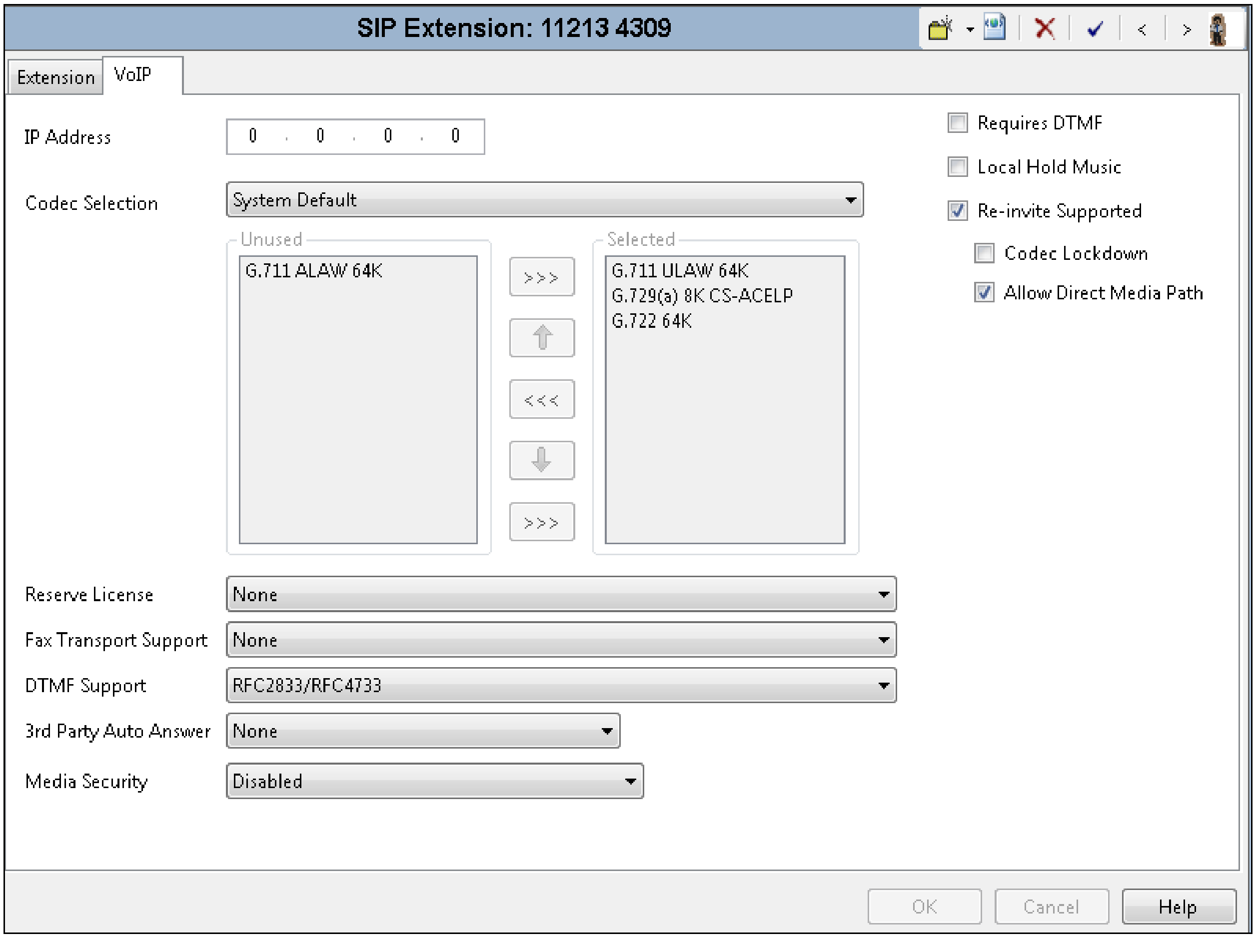


5.6. Administer Extension for Cetis SIP Telephone

From the configuration tree in the left pane, right-click on **Extension** and select **New SIP** from the pop-up list to add a new SIP extension (not shown). Enter the desired extension for the **Base Extension** field as shown below. In this example, Cetis SIP telephone was assigned extension 4309. This is the extension that Cetis SIP telephone will use to register with IP Office Server Edition.

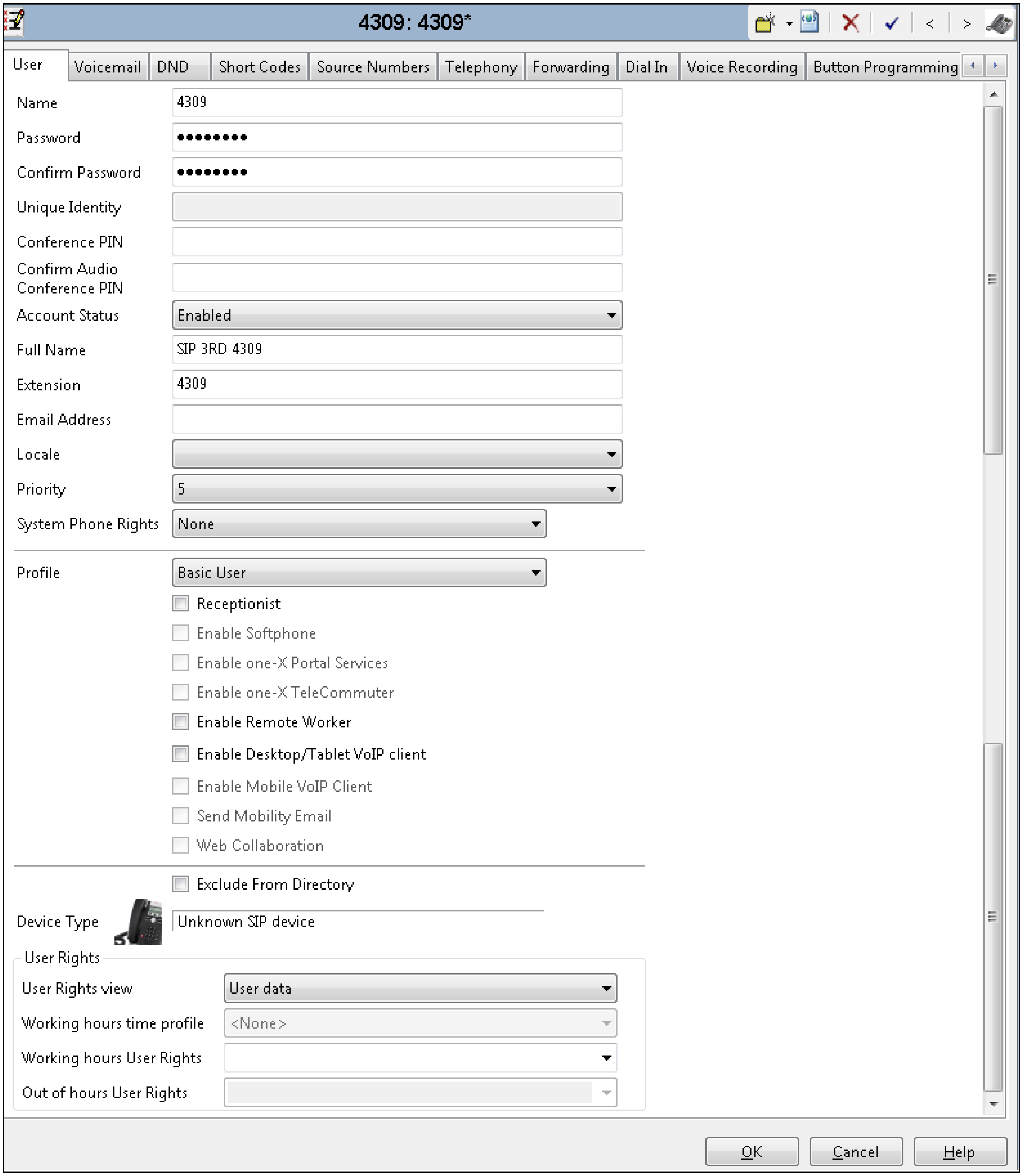


Select the **VoIP** tab and retain the default values in the all fields. During the compliance test, Cetis SIP telephone was tested with G.711and G.729 codecs. Enable **Allow Direct Media Path** so that audio/RTP flows directly between two SIP endpoints without using media resources in Avaya IP Office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

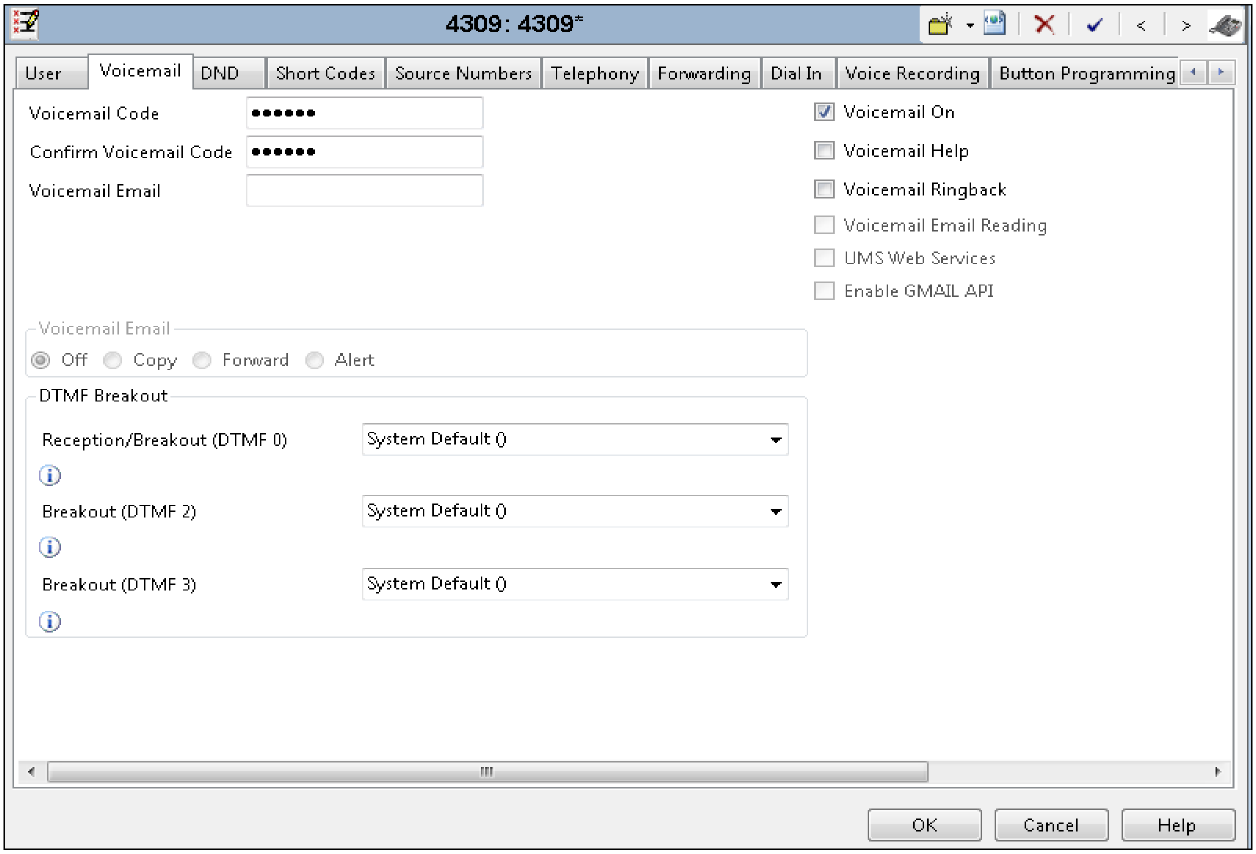


5.7. Administer SIP User for Cetis SIP Telephone

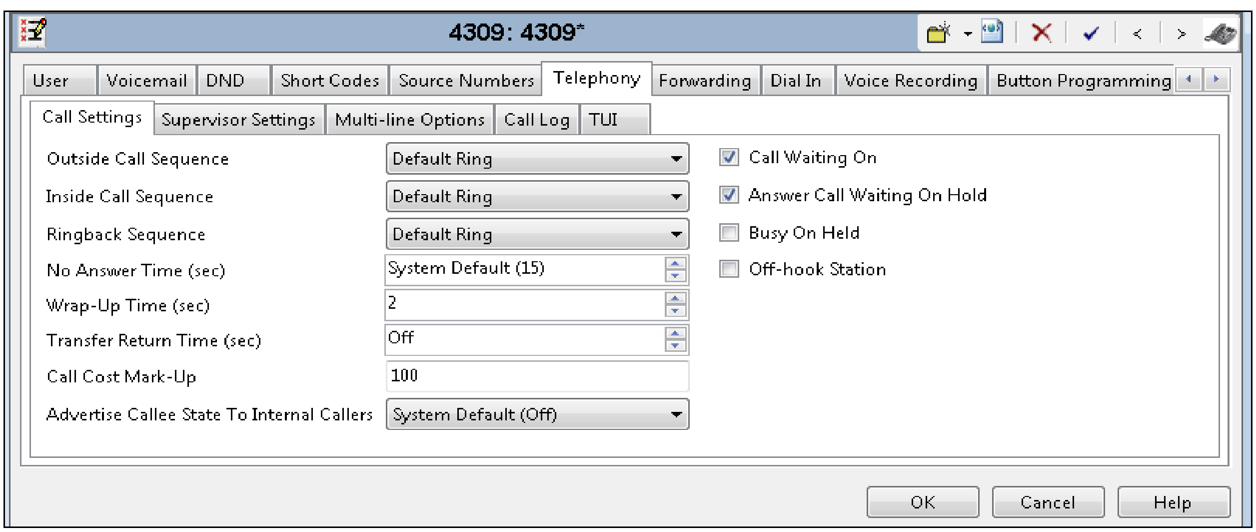
From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list (not shown). Enter desired values for the Name and Full Name fields. For the Extension field, enter the SIP extension created above.



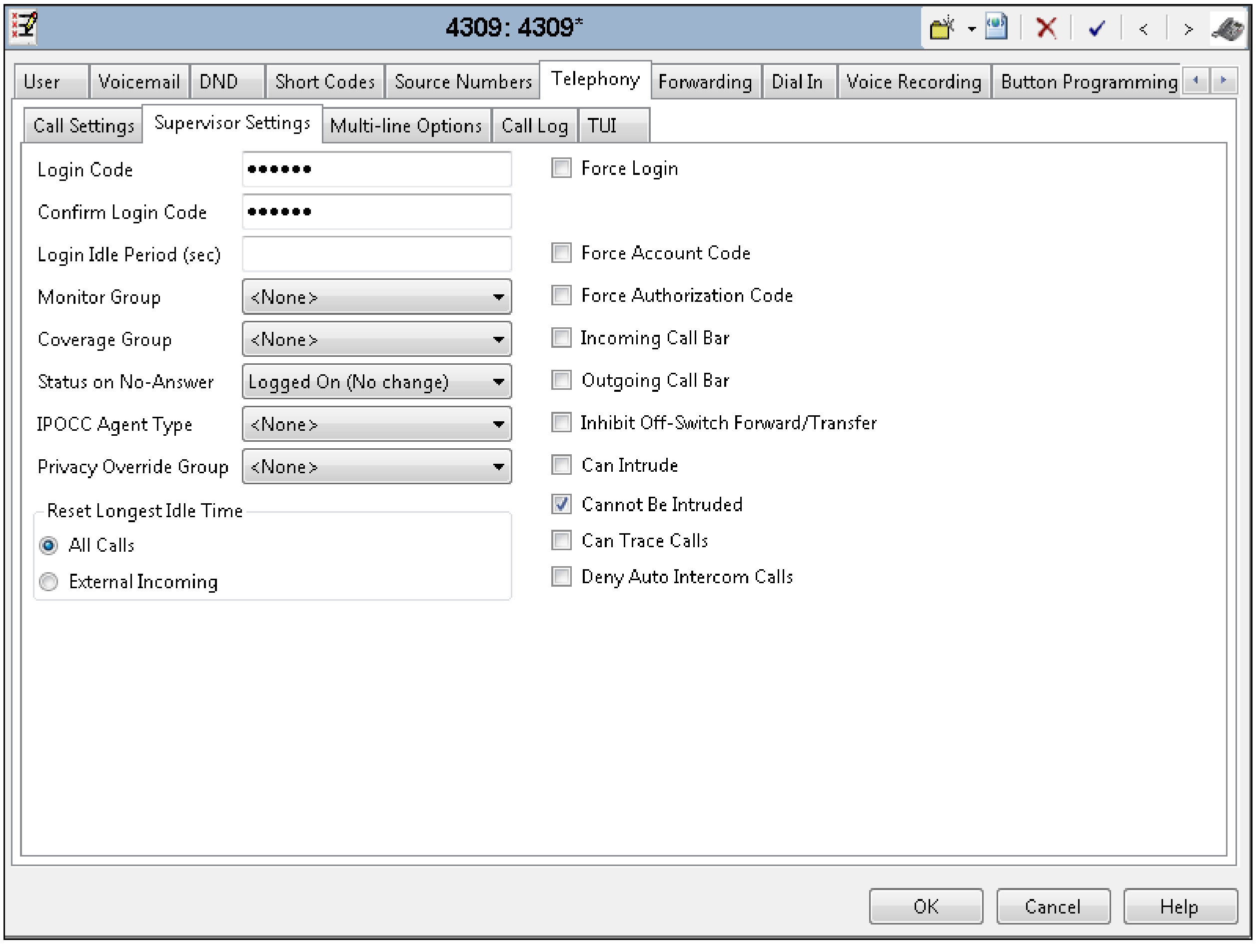
Select the **Voicemail** tab and select **Voicemail On** to enable voicemail for Cetis SIP telephone.



Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note the settings below for the user. Note: Call Waiting is required to allow a secondary incoming call to Cetis SIP telephone; otherwise, a second incoming call would be denied.

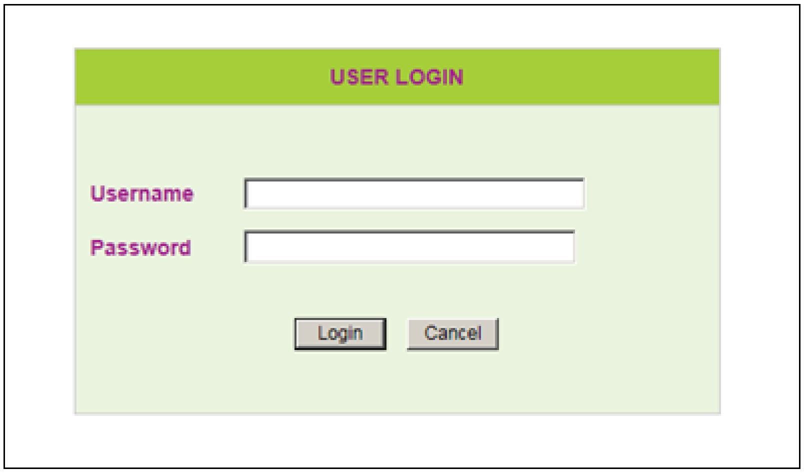


Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by Cetis SIP telephone to register with IP Office Server Edition.



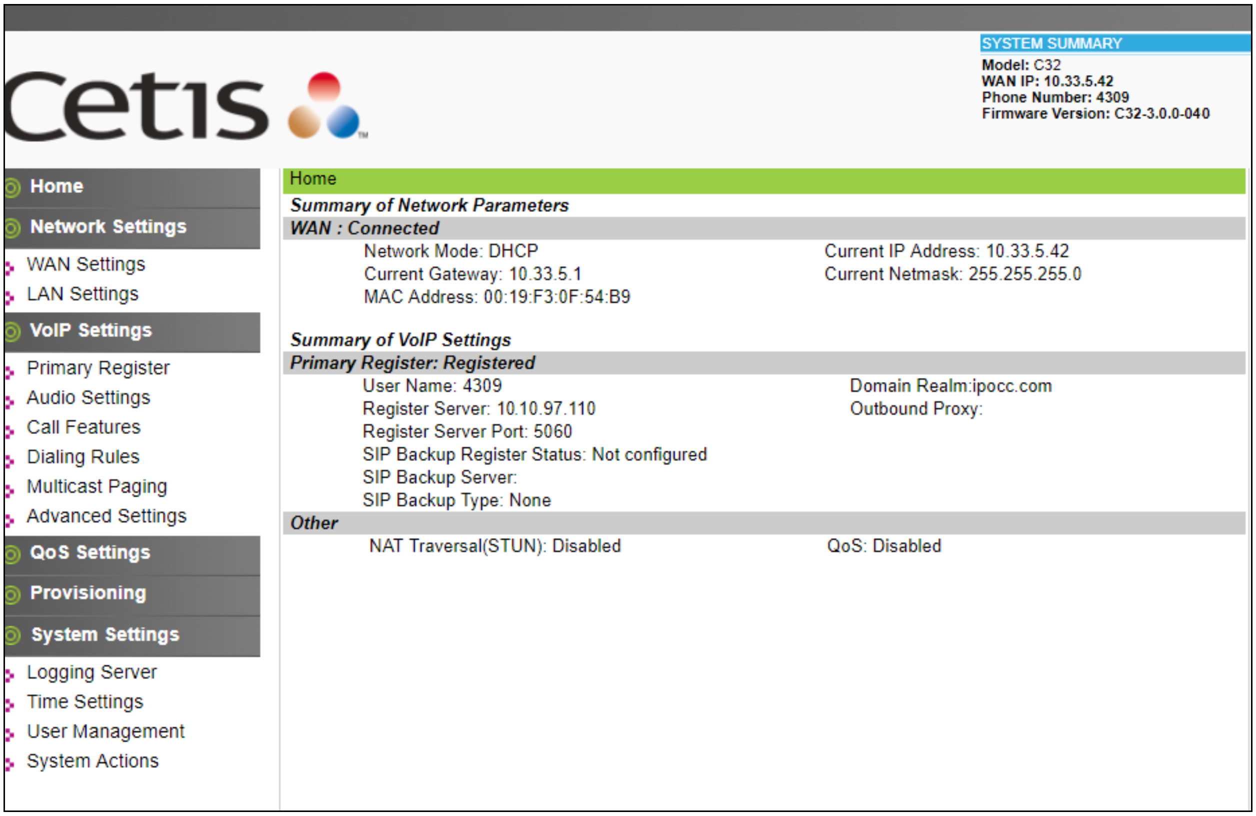
6. Configure Cetis SIP Telephones

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 static 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. Default **Username**/**Password** are admin/admin.



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: C32-3.0.0-040.bin is the firmware for **C**etis **C**orded **2**-line models including 3300IP and 9600IP.

Common Firmware on Cetis, Inc. SIP Phones

Cetis' current SIP firmware follows a naming convention based on and mated to the phone model name. The newest Cetis SIP phones all share the same base chipset and firmware, meaning that models using the same number firmware version share the same traits and compatibility. Server registration, SIP messaging, and call control are all the same. The different prefixes are to accommodate variances in single vs. 2-line capability, corded vs. cordless radio handsets and LCD display screen sizes.

Example: CC1-3.0.0-040.bin is a firmware file for the models associated with that CC1 prefix. Firmware number 3.0.0-040 could have any of the below prefixes tying it to the associated models

Prefix	Model	Features
CC1	**M100IP, ND2100IP, E100IP**	1-line, corded
CC2	**M200IP, ND2200IP, E200IP**	2-line, corded
CD1	**9600IP, M103IP, NDC2100IP, E103IP**	No LCD display, 1-line, cordless
CD2	**9602IP, M203IP, NDC2200IP, E203IP**	No LCD display, 2-line, cordless
C31	**3300IP**	2-Line LCD display, 1-line, corded
C32	**3302IP**	2-Line LCD display, 2-line, corded
CT1	**3300IP-TRM, M100IP-TRM**	1-line, corded, Trimline form
CT2	**3302IP-TRM, M100IP-TRM**	2-line, corded, Trimline form
CM1	**E100IP-TRM**	1-line, corded, Trimline form
CM2	**E200IP-TRM**	2-line, corded, Trimline form

CC = Cetis Corded I CD = Cetis DECT/Cordless I CT/CM = Cetis Trimline I C3 = Cetis 3300 series

The current SIP phone firmware (3.x) is NOT compatible with the SIP phones using (1.x) firmware or (2.x) firmware. Each of these SIP endpoints are distinct and separate hardware technologies, although they will have the same physical form factor and physical aesthetic characteristics in many cases.

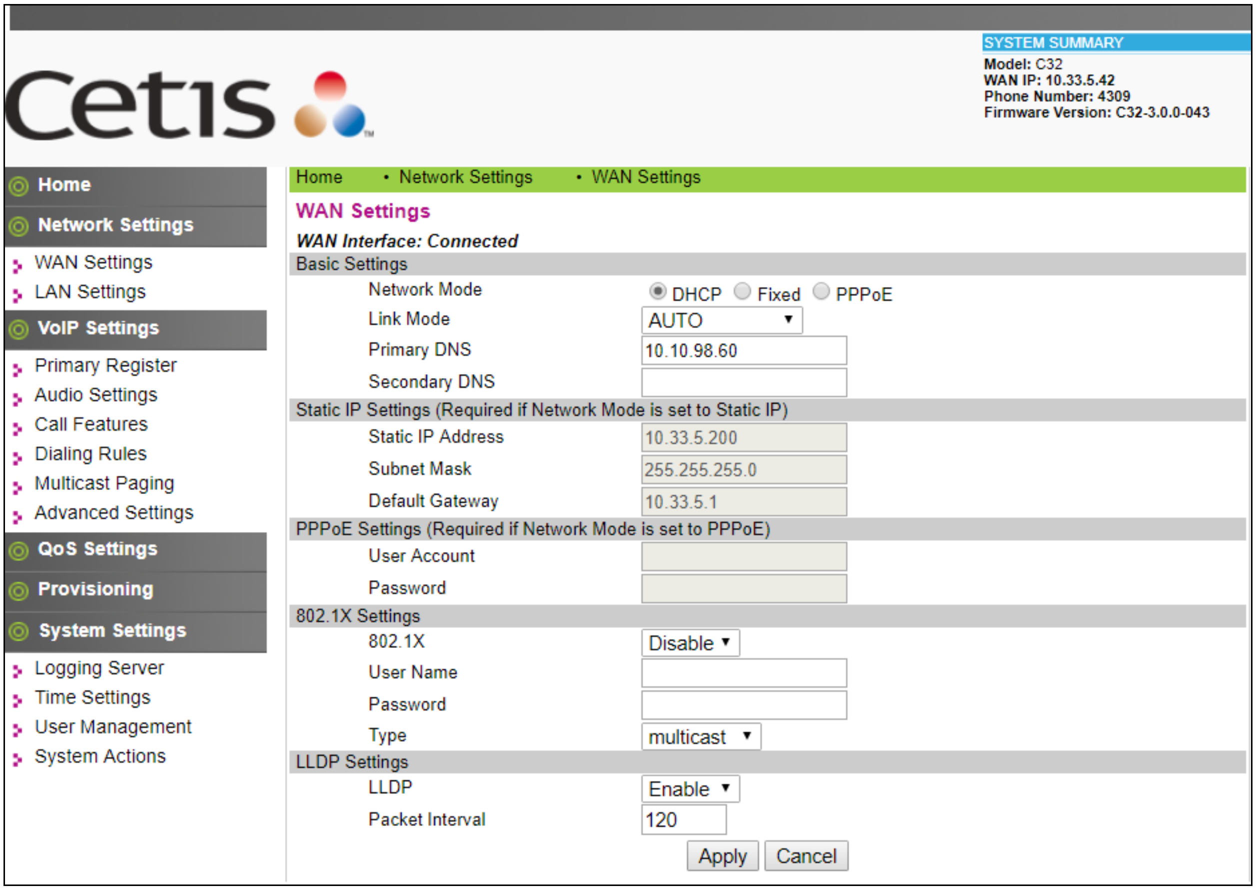
Notable additional features in the newest phones are:

Support of LLDP-MED protocols in network deployment I Support of macaddress named configuration files in network deployment. More sophisticated provisioning methods and re-direction server for cloud-based deployment is also supported.

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

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

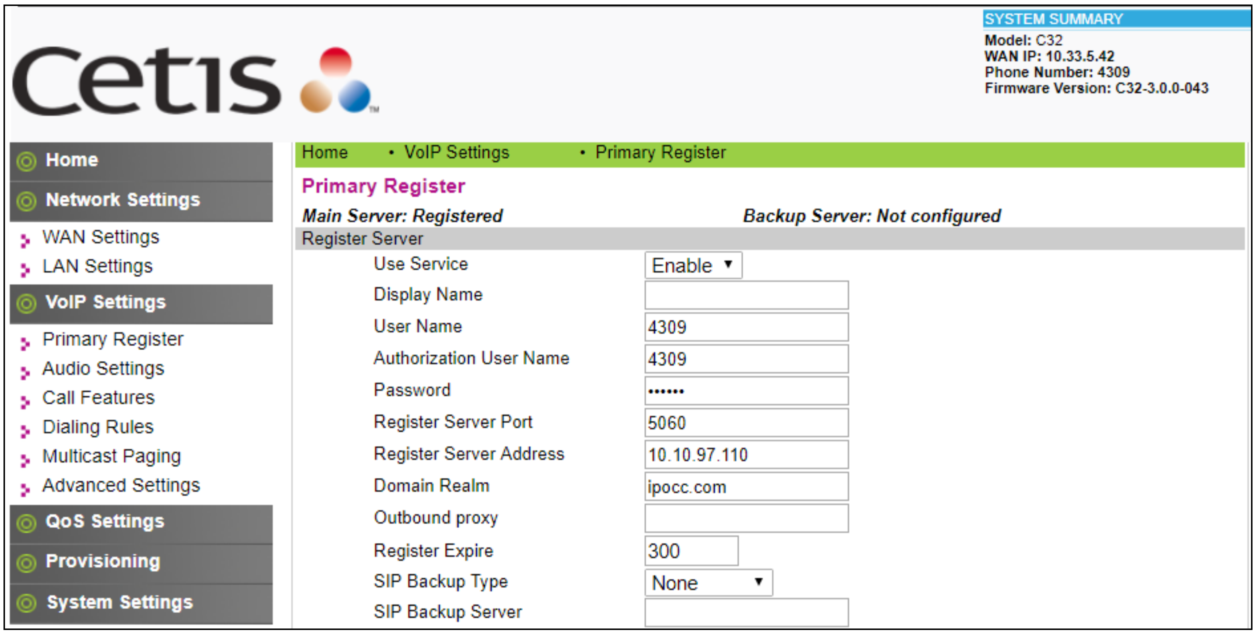
During the compliance test, DHCP was used. The following screen show what was configured and used.



6.2. VoIP Settings

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

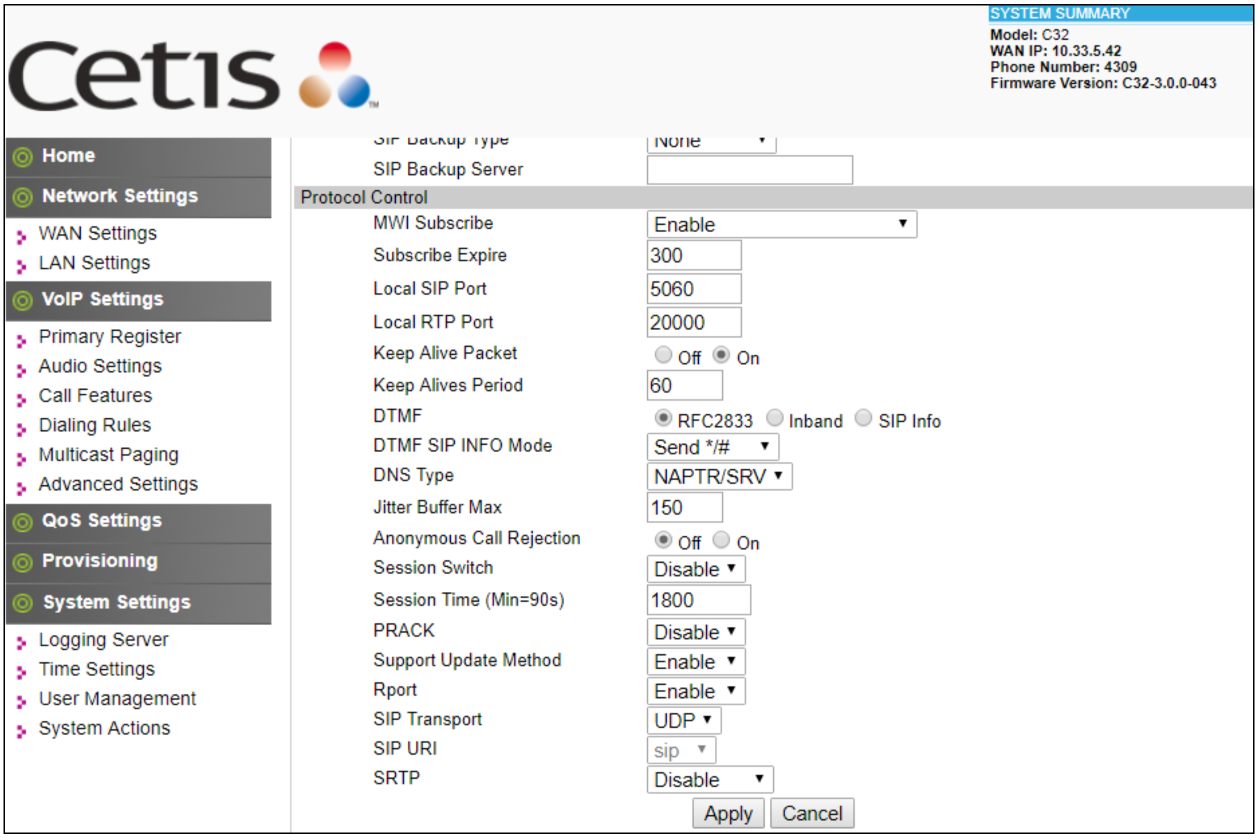
- **Use Service** – Select **Enable**.
- **Display Name** – Enter a descriptive name.
- **Register Server Address** – Enter the LAN1 IP address of IP Office.
- **Register Server Port** – Enter **5060** for UDP.
- **User Name** - Enter the user name created in **Section 5.2**.
- **Authorization User Name** - Enter the user name as configured in **Section 5.2**.
- **Password -** Enter the password created in **Section 5.2**.
- **Domain Realm** – Used **ipocc.com** during the test.
- Leave other fields at default value.



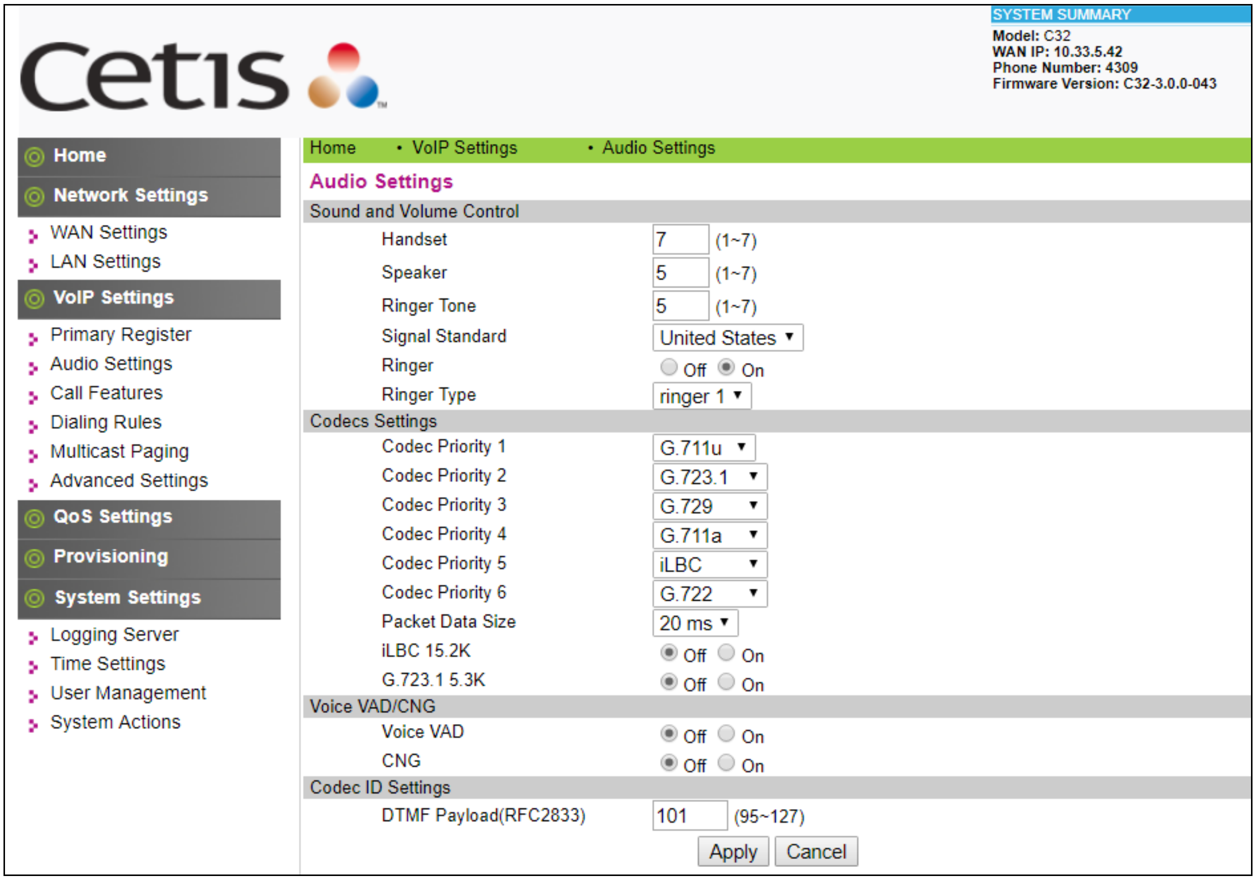
In the **Protocol Control** section, provide the folllowing values.

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

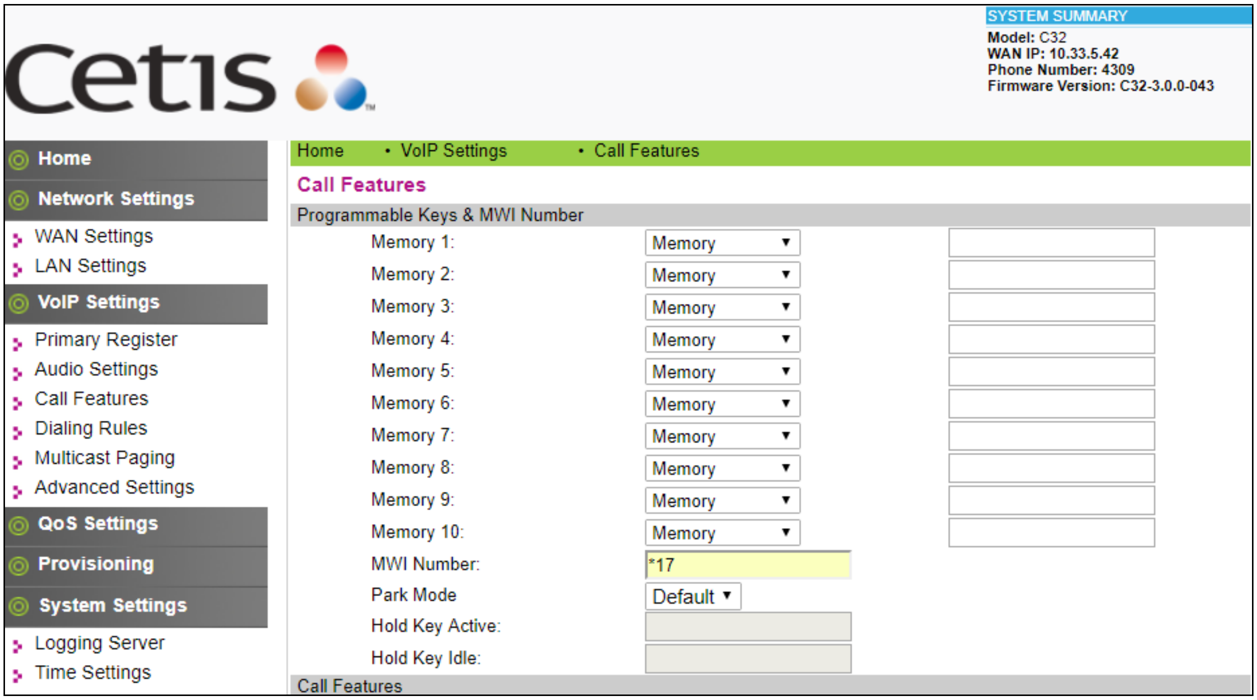
Click **Apply** button to save the changes.



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 supported by the Cetis SIP telephone.

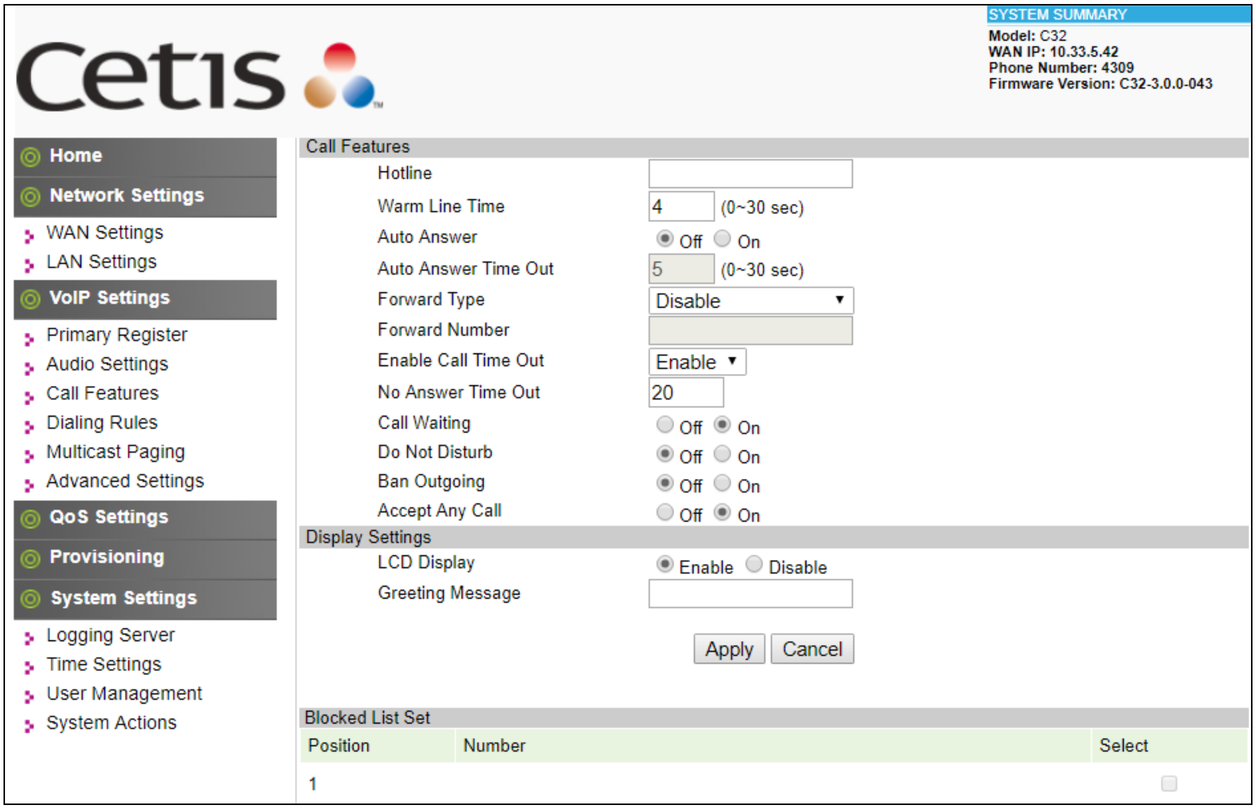


Select **Call Features** under the **VoIP Settings** section. In this page, a customer can program the memory buttons. For Cetis SIP telephone comes with 10 memory buttons. Enter the voicemail short code of IP Office messaging in the **MWI Number** box this setting allows user to access to the voicemail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



Under the **Call Features** section in the right pane, three features (Auto Answer, Do Not Disturb and Call Forward) were tested. The configuration below shows these features at their default values.

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

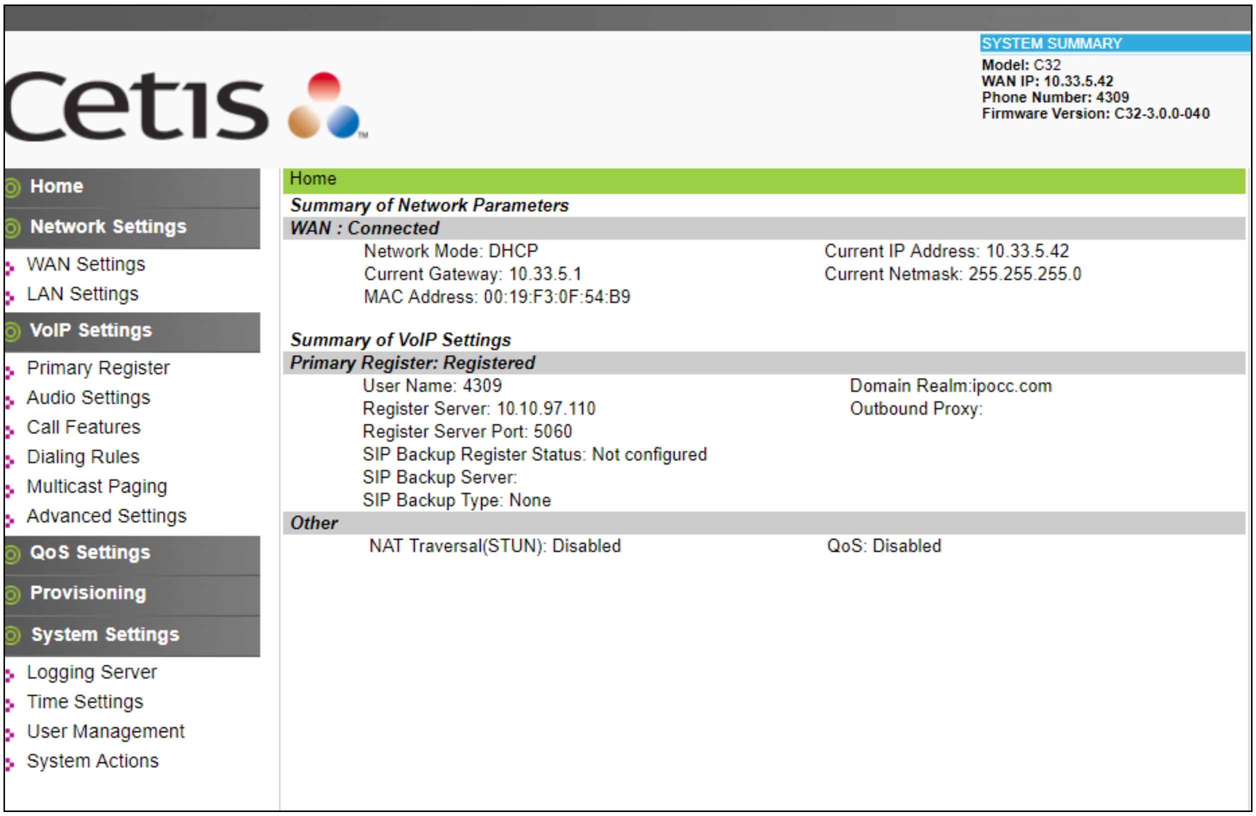


7. Verification Steps

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

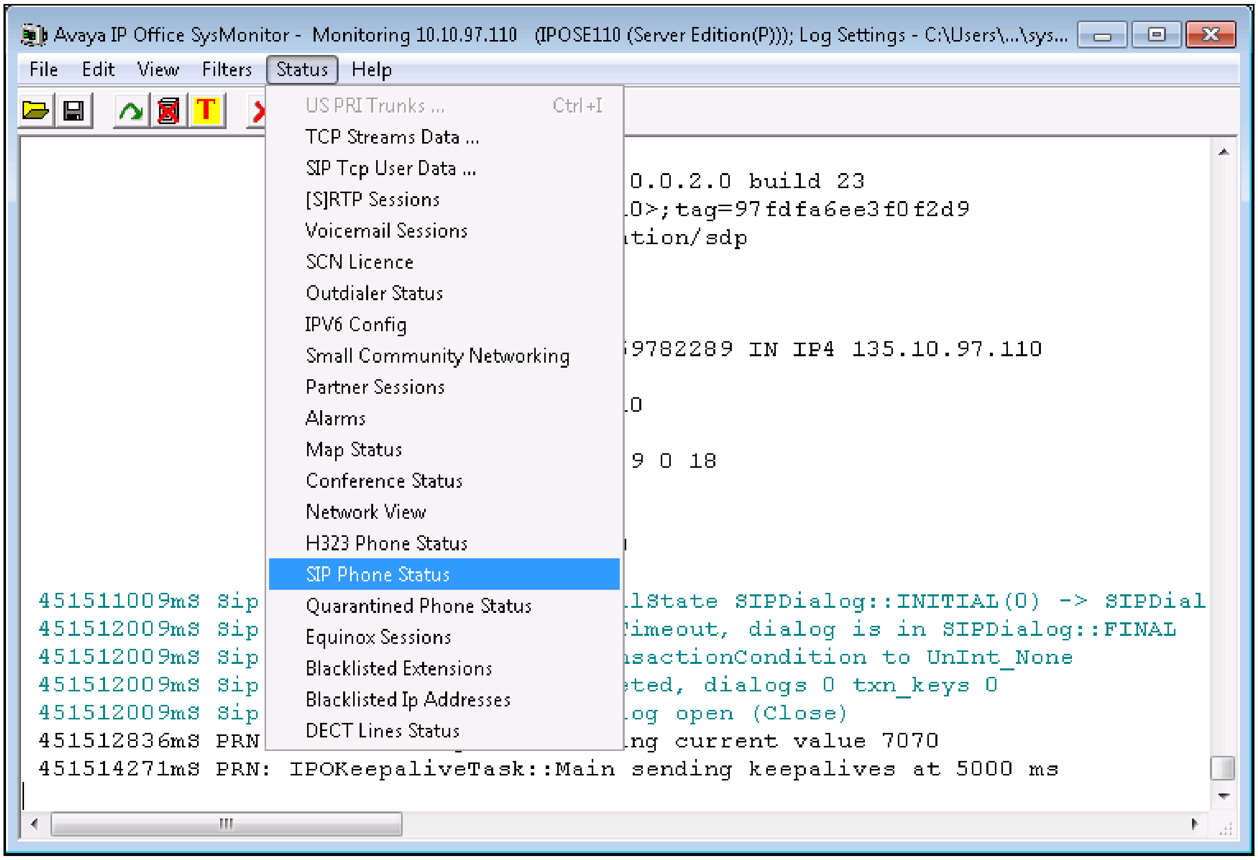
7.1. Verify Cetis SIP Telephones

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

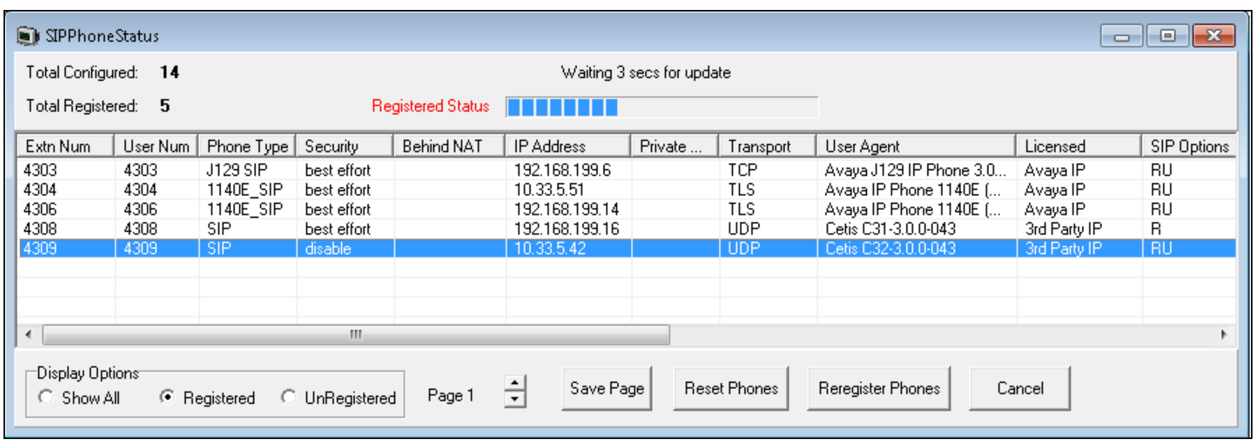


7.2. Verify Avaya IP Office

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 C32.3.0.0.40 in the list.



8. Conclusion

These Application Notes have described the administration steps required to integrate the Cetis 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetis 3300IP Series and 9600IP Series 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] *Avaya IP Office Platform Solution Description*, Release 11.0, May 2018.
[2] *Avaya IP Office Platform Feature Description*, Release 11.0, May 2018.
[3] *IP Office Platform 11.0 Deploying Avaya IP Office Essential Edition*, Document Number 15-601042, Issue 33g, 20 May 2018.
[4] *Administering Avaya IP Office Platform with Manager*, Release 11.0, May 2018.
[5] *IP Office Platform 10.1 Using Avaya IP Office Platform System Status*, Document 15-601758, Issue 13a, 05 April, 2018.
[6] *IP Office Platform 11.0 Using IP Office System Monitor*, Document 15-601019, Issue 09b, 10 may, 2018.

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.