



Avaya Solution & Interoperability Test Lab

Application Notes for Cetus 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 Cetus 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office. The Cetus 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 Cetus 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetus 3300IP Series and 9600IP Series SIP Telephones were designed for the hospitality industry. In the compliance test, Cetus 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 Cetus 3300IP Series and 9600IP Series SIP Telephones do not utilize TLS and secure media SRTP encryption features as requested by Cetus.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Cetus 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 Cetus 3200IP SIP Telephones with IP Office Server Edition.
- Calls between Cetus telephones and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetus 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 Cetus telephones.
- Proper system recovery after a restart of the Cetus telephones and loss of IP connectivity.

The serviceability testing focused on verifying that the Cetus 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 Cetus 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 Cetus.

2.3. Support

For technical support on the Cetus 3300IP and 9600IP Telephones, contact Cetus 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 Cetus 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetus SIP telephones registered with Avaya IP Office via SIP.

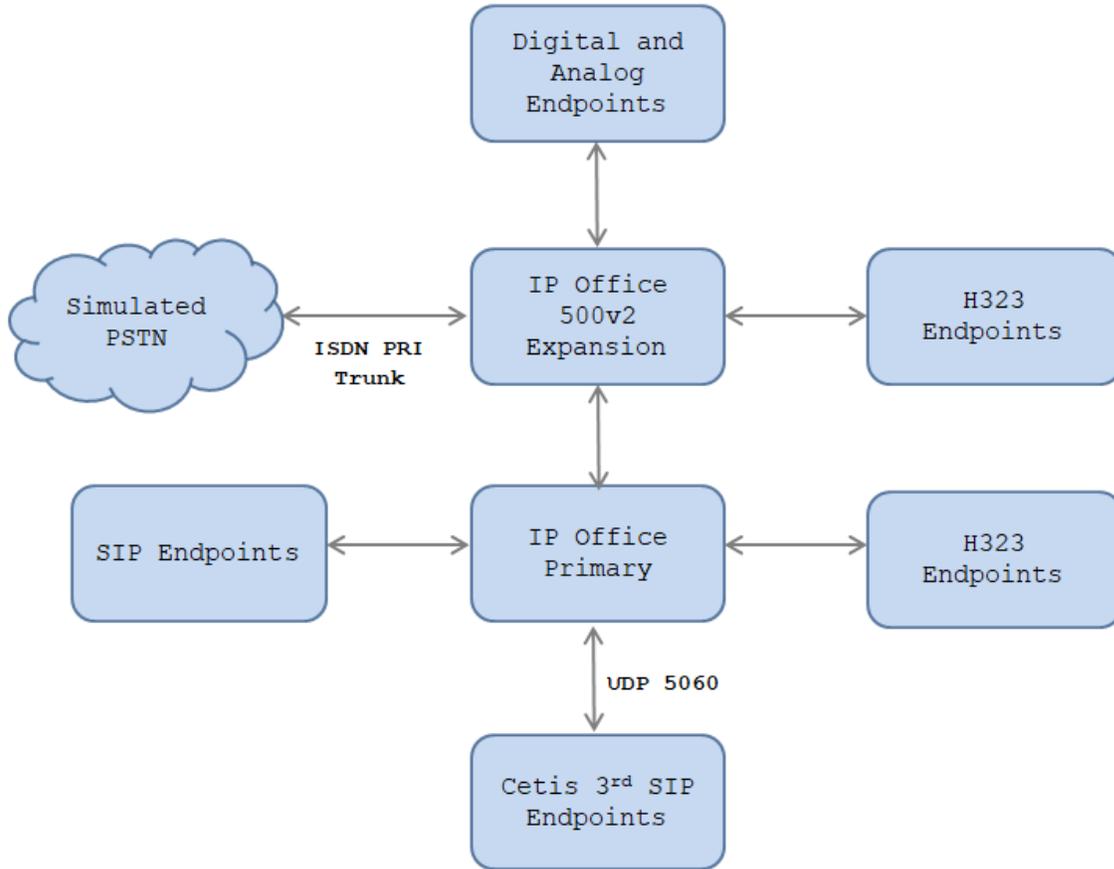


Figure 1: Test Configuration Diagram with IP Office

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
Cetus 3 rd -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 Cetis Corded 2-line models including 3300IP and 9600IP

5. Configure Avaya IP Office

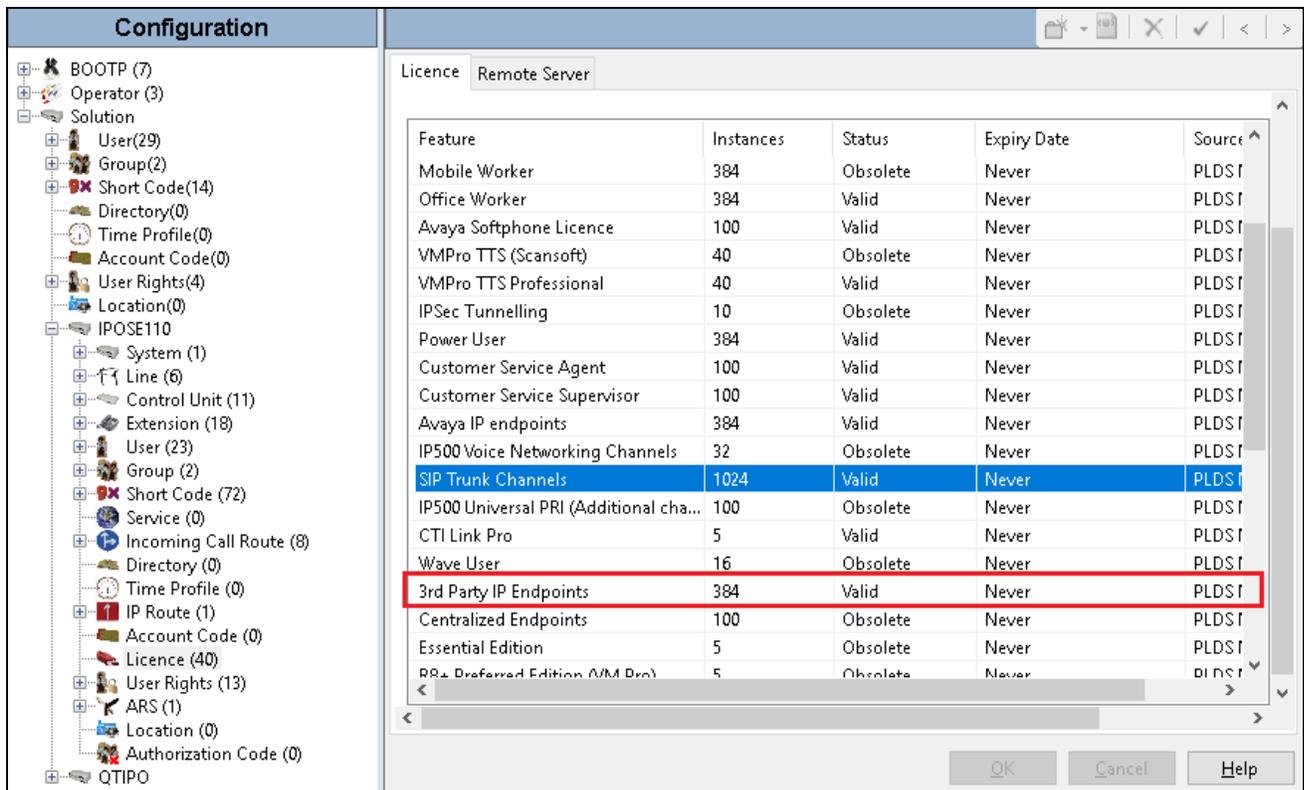
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- 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.

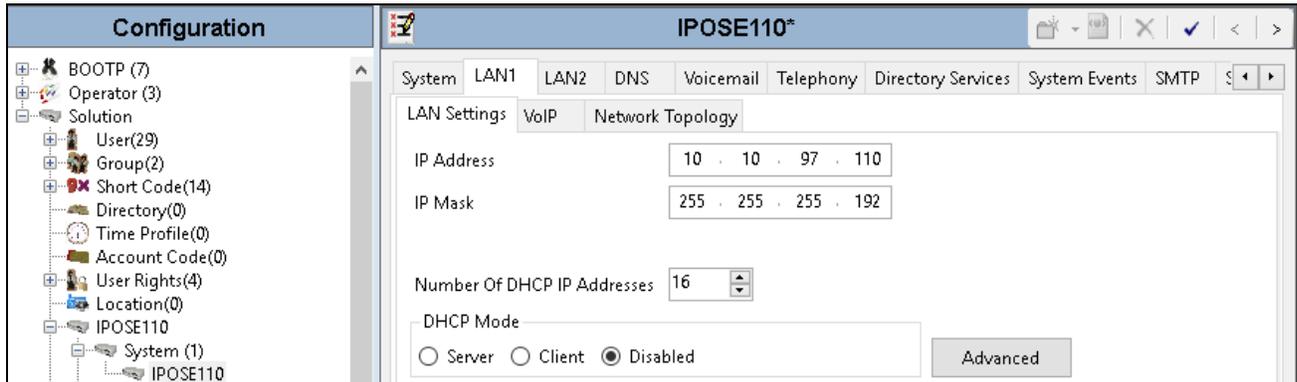
The **Avaya IP Office for Server Edition Manager** screen is displayed. From the configuration tree in the left pane, select **License**. Verify that the **3rd Party IP Endpoints** license is “Valid”, and that the **Instances** value is sufficient for the desired maximum number of simultaneous registrations.



Feature	Instances	Status	Expiry Date	Source
Mobile Worker	384	Obsolete	Never	PLDS
Office Worker	384	Valid	Never	PLDS
Avaya Softphone Licence	100	Valid	Never	PLDS
VMPro TTS (Scansoft)	40	Obsolete	Never	PLDS
VMPro TTS Professional	40	Valid	Never	PLDS
IPSec Tunnelling	10	Obsolete	Never	PLDS
Power User	384	Valid	Never	PLDS
Customer Service Agent	100	Valid	Never	PLDS
Customer Service Supervisor	100	Valid	Never	PLDS
Avaya IP endpoints	384	Valid	Never	PLDS
IP500 Voice Networking Channels	32	Obsolete	Never	PLDS
SIP Trunk Channels	1024	Valid	Never	PLDS
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS
CTI Link Pro	5	Valid	Never	PLDS
Wave User	16	Obsolete	Never	PLDS
3rd Party IP Endpoints	384	Valid	Never	PLDS
Centralized Endpoints	100	Obsolete	Never	PLDS
Essential Edition	5	Obsolete	Never	PLDS
RR+ Preferred Edition (M Pro)	5	Obsolete	Never	PLDS

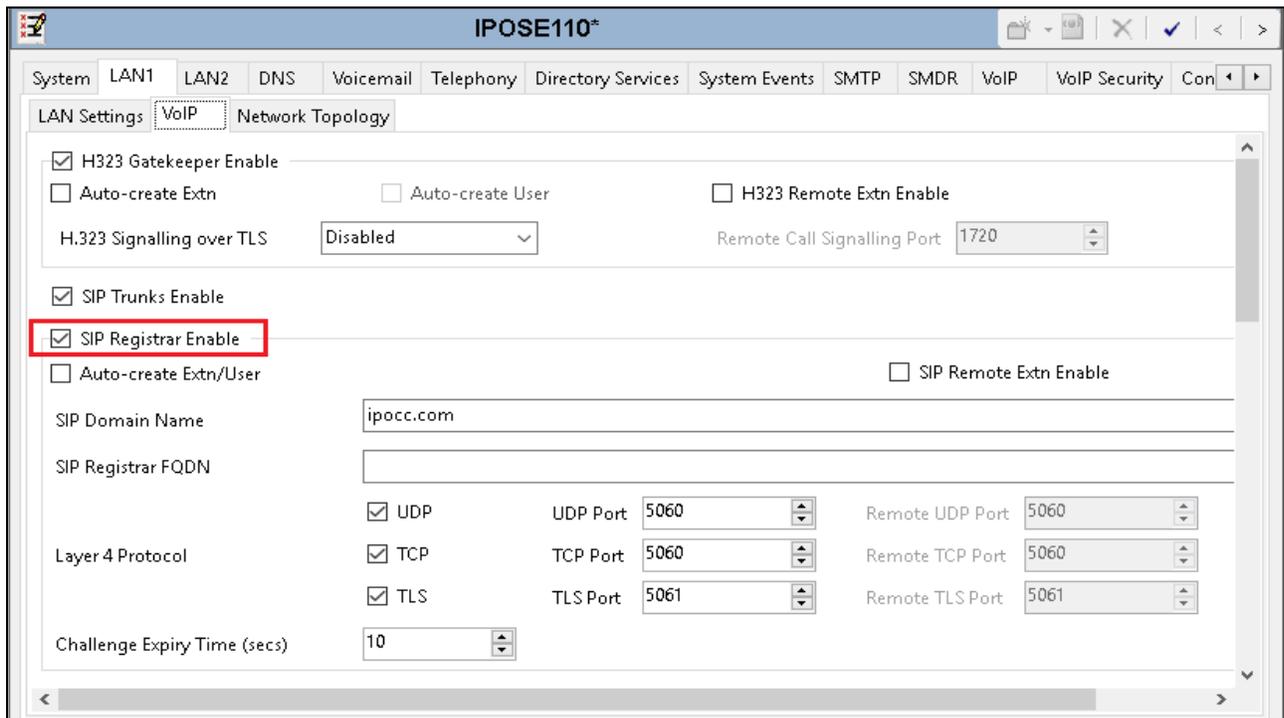
5.2. Obtain LAN IP Address

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 Cetus SIP telephone.

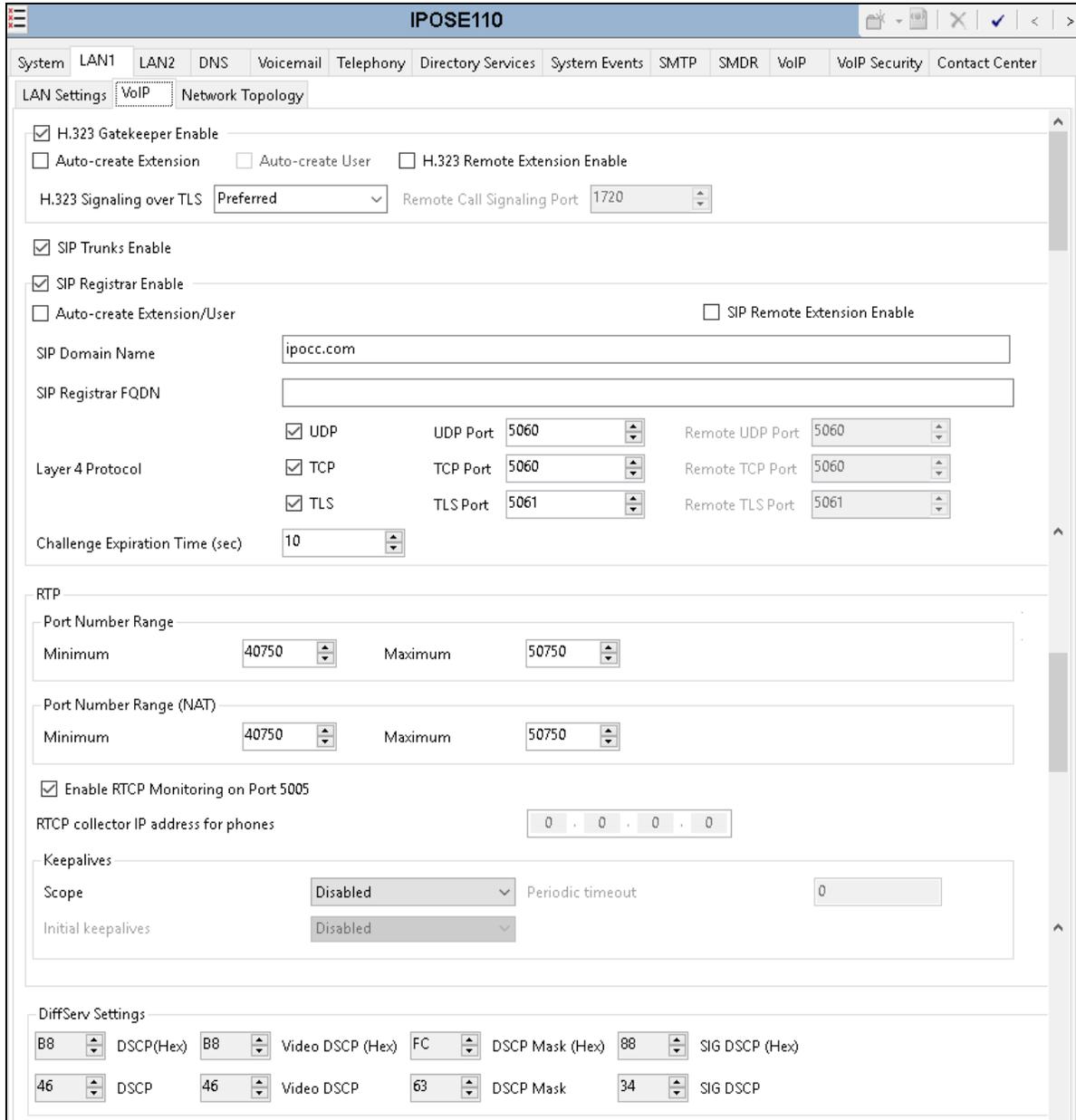


5.3. Enable SIP Registrar

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 Cetus 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**.



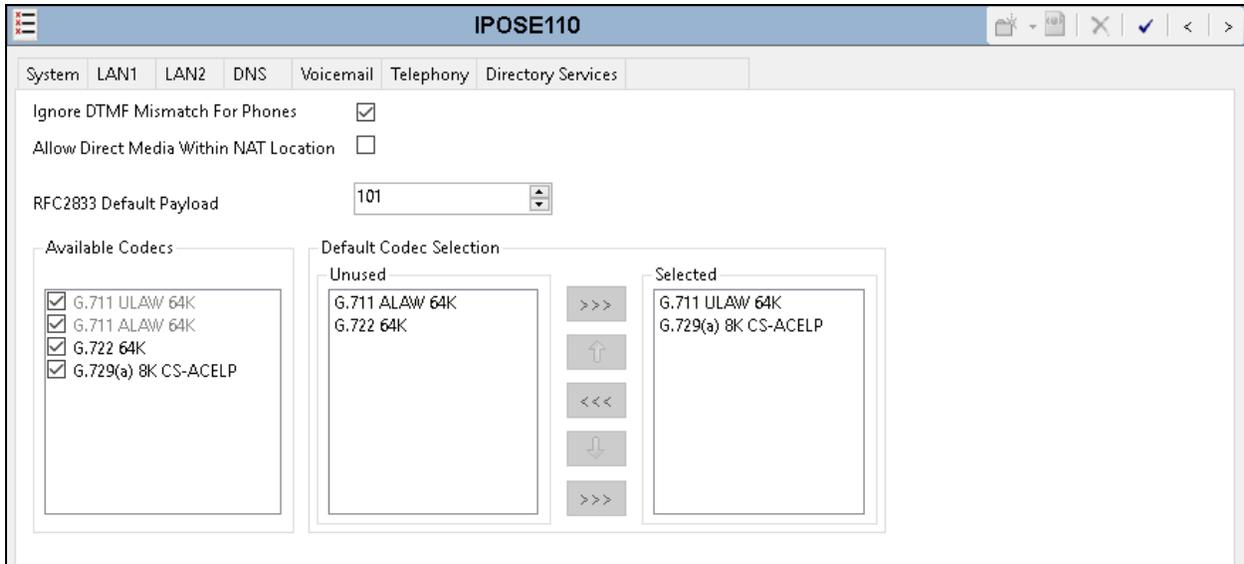
5.4. System Telephony Settings

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).

The screenshot shows the IPOSE110 configuration window with the 'Telephony' tab selected. The 'Companding Law' section is visible, showing 'U-Law' selected for both 'Switch' and 'Line'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked and highlighted with a red box. Other settings include 'Dial Delay Time (sec)' set to 4, 'Default No Answer Time (sec)' set to 15, and 'Default Currency' set to USD. The 'Login Code Complexity' section has 'Enforcement' and 'Complexity' checked. The 'RTCP Collector Configuration' section has 'Send RTCP to an RTCP Collector' unchecked, with 'Server Address' set to 0.0.0.0, 'UDP Port Number' set to 5005, and 'RTCP reporting interval (sec)' set to 5. The 'OK', 'Cancel', and 'Help' buttons are visible at the bottom right.

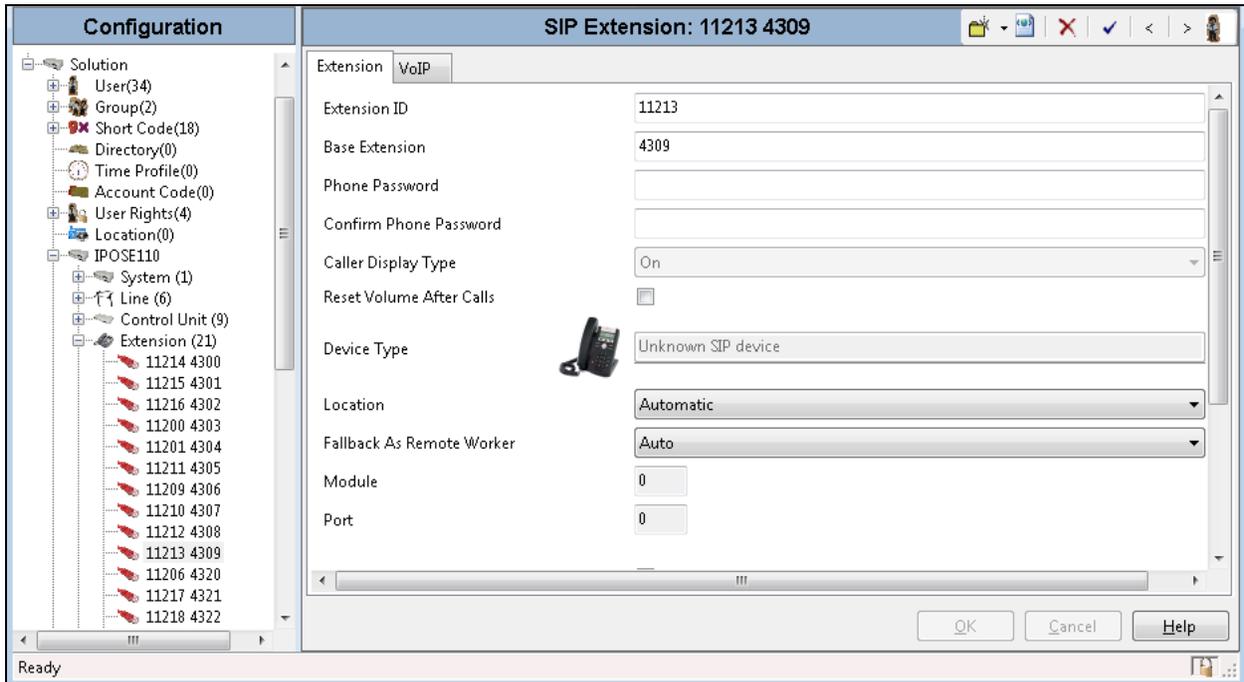
5.5. Administer Codec Settings

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 Cetus 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, Cetus SIP telephone was assigned extension 4309. This is the extension that Cetus 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, Cetus SIP telephone 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 Cetus SIP telephone that does not support the media security.

The screenshot shows the configuration window for SIP Extension 11213 4309. The window is titled "SIP Extension: 11213 4309" and has a "VoIP" tab selected. The configuration fields are as follows:

- IP Address: 0 . 0 . 0 . 0
- Codec Selection: System Default
- Unused codecs: G.711 ALAW 64K
- Selected codecs: G.711 ULAW 64K, G.729(a) 8K CS-ACELP, G.722 64K
- Reserve License: None
- Fax Transport Support: None
- DTMF Support: RFC2833/RFC4733
- 3rd Party Auto Answer: None
- Media Security: Disabled

On the right side, there are several checkboxes:

- Requires DTMF
- Local Hold Music
- Re-invite Supported
- Codec Lockdown
- Allow Direct Media Path

At the bottom, there are buttons for "OK", "Cancel", and "Help".

5.7. Administer SIP User for Cetus SIP Telephone

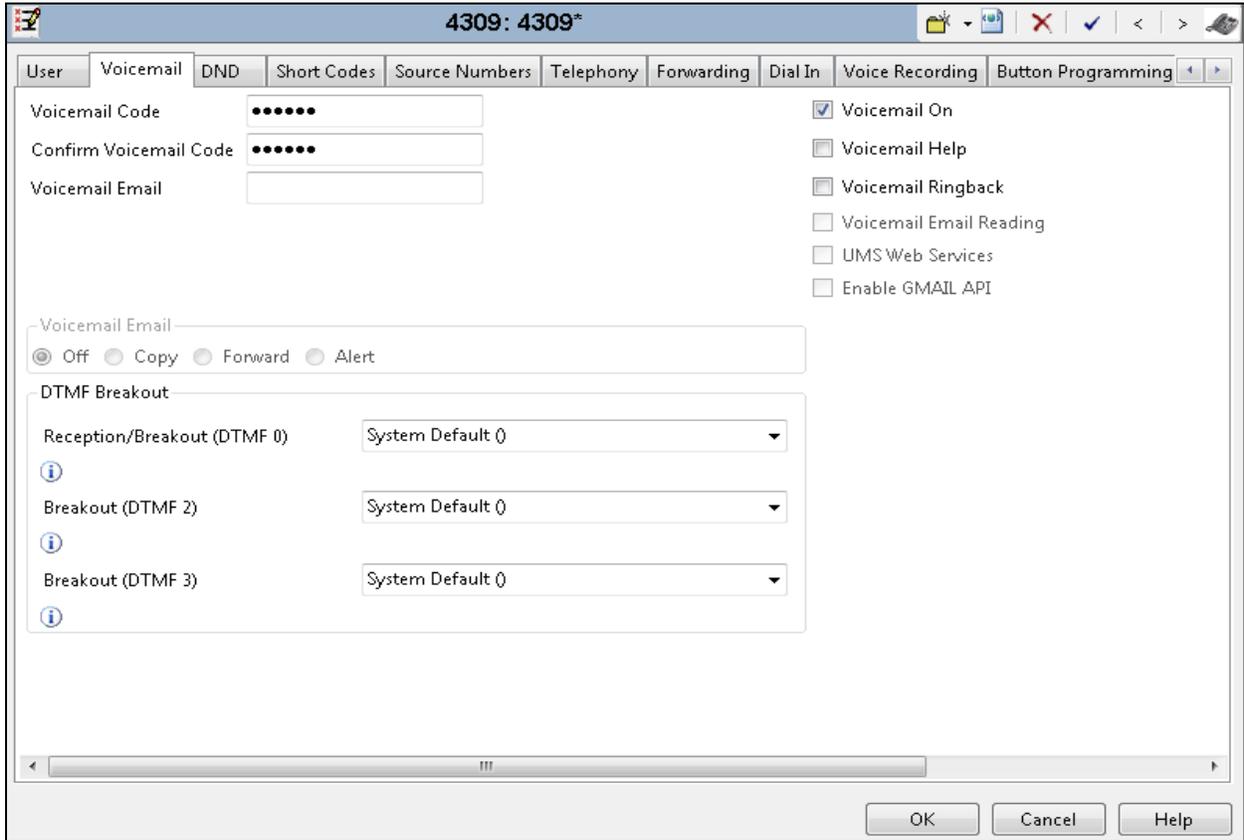
From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list (not shown). Enter desired values for the Name and Full Name fields. For the Extension field, enter the SIP extension created above.

The screenshot shows the 'User' configuration window for extension 4309. The window has a title bar '4309: 4309*' and a menu bar with options: User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The main area contains the following fields and options:

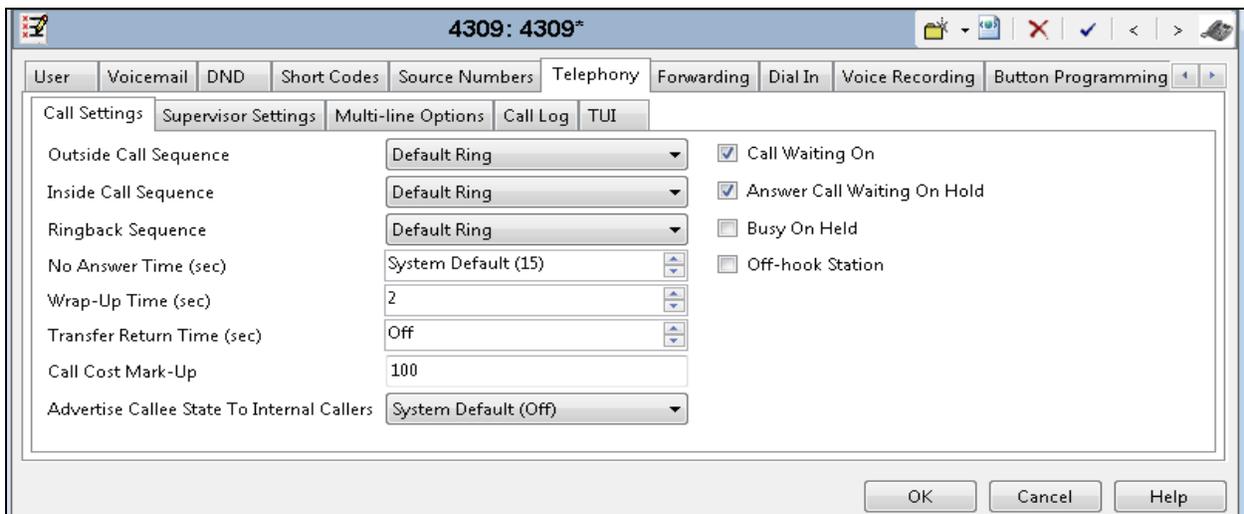
- Name: 4309
- Password: [Redacted]
- Confirm Password: [Redacted]
- Unique Identity: [Redacted]
- Conference PIN: [Redacted]
- Confirm Audio Conference PIN: [Redacted]
- Account Status: Enabled (dropdown)
- Full Name: SIP 3RD 4309
- Extension: 4309
- Email Address: [Redacted]
- Locale: [Redacted]
- Priority: 5 (dropdown)
- System Phone Rights: None (dropdown)
- Profile: Basic User (dropdown)
 - Receptionist
 - Enable Softphone
 - Enable one-X Portal Services
 - Enable one-X TeleCommuter
 - Enable Remote Worker
 - Enable Desktop/Tablet VoIP client
 - Enable Mobile VoIP Client
 - Send Mobility Email
 - Web Collaboration
- Exclude From Directory
- Device Type: [Phone icon] Unknown SIP device
- User Rights:
 - User Rights view: User data (dropdown)
 - Working hours time profile: <None> (dropdown)
 - Working hours User Rights: [Redacted] (dropdown)
 - Out of hours User Rights: [Redacted] (dropdown)

Buttons at the bottom: OK, Cancel, Help.

Select the **Voicemail** tab and select **Voicemail On** to enable voicemail for Cetus 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 Cetus 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 Cetus SIP telephone to register with IP Office Server Edition.

4309: 4309*

User Voicemail DND Short Codes Source Numbers **Telephony** Forwarding Dial In Voice Recording Button Programming

Call Settings **Supervisor Settings** Multi-line Options Call Log TUI

Login Code: [Masked] Force Login

Confirm Login Code: [Masked]

Login Idle Period (sec): [Empty]

Monitor Group: <None>

Coverage Group: <None>

Status on No-Answer: Logged On (No change)

IPOCC Agent Type: <None>

Privacy Override Group: <None>

Reset Longest Idle Time: [Empty]

All Calls External Incoming

Force Account Code

Force Authorization Code

Incoming Call Bar

Outgoing Call Bar

Inhibit Off-Switch Forward/Transfer

Can Intrude

Cannot Be Intruded

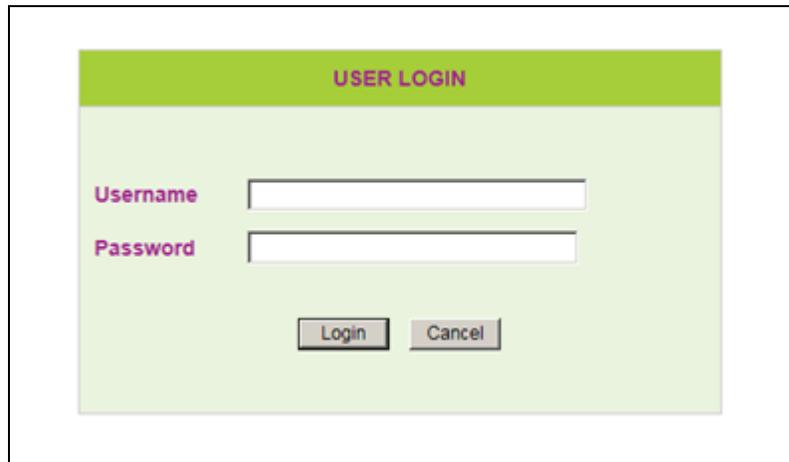
Can Trace Calls

Deny Auto Intercom Calls

OK Cancel Help

6. Configure Cetus SIP Telephones

Access the Cetus SIP Telephones web interface using the URL "<http://ip-address>" in an Internet browser window, where "ip-address" is the IP address of the Cetus telephone. By default, DHCP is enabled on the Cetus telephones. For this compliance test, a static IP address was assigned to the Cetus telephone. To determine the IP address assigned to the Cetus telephone, enter **47# on the telephone to hear the IP address. Default **Username/Password** are admin/admin.



The image shows a web interface for user login. It features a green header bar with the text "USER LOGIN" in white. Below the header, there are two input fields: "Username" and "Password", both with white text labels and empty text boxes. At the bottom of the form, there are two buttons: "Login" and "Cancel", both with black text on a light gray background.

6.1. Network Settings

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

The screenshot displays the Cetis SIP phone's web interface. The top left features the Cetis logo. The top right corner contains a 'SYSTEM SUMMARY' box with the following information: Model: C32, WAN IP: 10.33.5.42, Phone Number: 4309, and Firmware Version: C32-3.0.0-040. A left-hand navigation menu lists various settings categories: Home, Network Settings (selected), VoIP Settings, QoS Settings, and Provisioning. Under Network Settings, 'WAN Settings' is highlighted. The main content area shows the 'Summary of Network Parameters' for the WAN, which is 'Connected'. It lists Network Mode: DHCP, Current Gateway: 10.33.5.1, MAC Address: 00:19:F3:0F:54:B9, Current IP Address: 10.33.5.42, and Current Netmask: 255.255.255.0. Below this is the 'Summary of VoIP Settings' for the 'Primary Register: Registered'. It shows User Name: 4309, Register Server: 10.10.97.110, Register Server Port: 5060, SIP Backup Register Status: Not configured, SIP Backup Server, and SIP Backup Type: None. The 'Other' section indicates NAT Traversal(STUN): Disabled and QoS: Disabled.

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 Cetis Corded 2-line models including 3300IP and 9600IP.

Common Firmware on Cetus, Inc. SIP Phones

Cetus' current SIP firmware follows a naming convention based on and mated to the phone model name. The newest Cetus 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 = Cetus Corded | CD = Cetus DECT/Cordless | CT/CM = Cetus Trimline | C3 = Cetus 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 | 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.

The screenshot displays the Cetis WAN Settings interface. The left sidebar contains navigation options: Home, Network Settings (selected), VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area shows the WAN Settings page with the following configuration:

SYSTEM SUMMARY
Model: C32
WAN IP: 10.33.5.42
Phone Number: 4309
Firmware Version: C32-3.0.0-043

Home • Network Settings • WAN Settings

WAN Settings
WAN Interface: Connected

Basic Settings

Network Mode	<input checked="" type="radio"/> DHCP <input type="radio"/> Fixed <input type="radio"/> PPPoE
Link Mode	AUTO
Primary DNS	10.10.98.60
Secondary DNS	

Static IP Settings (Required if Network Mode is set to Static IP)

Static IP Address	10.33.5.200
Subnet Mask	255.255.255.0
Default Gateway	10.33.5.1

PPPoE Settings (Required if Network Mode is set to PPPoE)

User Account	
Password	

802.1X Settings

802.1X	Disable
User Name	
Password	
Type	multicast

LLDP Settings

LLDP	Enable
Packet Interval	120

Apply Cancel

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.

The screenshot displays the Cetis web interface for configuring VoIP settings. The top right corner shows a 'SYSTEM SUMMARY' box with the following information: Model: C32, WAN IP: 10.33.5.42, Phone Number: 4309, and Firmware Version: C32-3.0.0-043. The main navigation menu on the left includes Home, Network Settings (WAN, LAN), VoIP Settings (Primary Register, Audio, Call Features, Dialing Rules, Multicast Paging, Advanced Settings), QoS Settings, Provisioning, and System Settings. The current page is 'Primary Register' under 'VoIP Settings'. It shows the status 'Main Server: Registered' and 'Backup Server: Not configured'. The 'Register Server' section contains the following configuration fields:

Field	Value
Use Service	Enable
Display Name	
User Name	4309
Authorization User Name	4309
Password
Register Server Port	5060
Register Server Address	10.10.97.110
Domain Realm	ipocc.com
Outbound proxy	
Register Expire	300
SIP Backup Type	None
SIP Backup Server	

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

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

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

Cetis 

SYSTEM SUMMARY
Model: C32
WAN IP: 10.33.5.42
Phone Number: 4309
Firmware Version: C32-3.0.0-043

Home
Network Settings
WAN Settings
LAN Settings
VoIP Settings
Primary Register
Audio Settings
Call Features
Dialing Rules
Multicast Paging
Advanced Settings
QoS Settings
Provisioning
System Settings
Logging Server
Time Settings
User Management
System Actions

SIP Backup type: none
SIP Backup Server:

Protocol Control

MWI Subscribe: Enable
Subscribe Expire: 300
Local SIP Port: 5060
Local RTP Port: 20000
Keep Alive Packet: Off On
Keep Alives Period: 60
DTMF: RFC2833 Inband SIP Info
DTMF SIP INFO Mode: Send *#
DNS Type: NAPTR/SRV
Jitter Buffer Max: 150
Anonymous Call Rejection: Off On
Session Switch: Disable
Session Time (Min=90s): 1800
PRACK: Disable
Support Update Method: Enable
Rport: Enable
SIP Transport: UDP
SIP URI: sip
SRTP: Disable

Apply Cancel

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.

Cetis 

SYSTEM SUMMARY
 Model: C32
 WAN IP: 10.33.5.42
 Phone Number: 4309
 Firmware Version: C32-3.0.0-043

Home • VoIP Settings • Audio Settings

Audio Settings

Sound and Volume Control

Handset	<input type="text" value="7"/>	(1~7)
Speaker	<input type="text" value="5"/>	(1~7)
Ringer Tone	<input type="text" value="5"/>	(1~7)
Signal Standard	United States ▾	
Ringer	<input type="radio"/> Off <input checked="" type="radio"/> On	
Ringer Type	ringer 1 ▾	

Codecs Settings

Codec Priority 1	G.711u ▾	
Codec Priority 2	G.723.1 ▾	
Codec Priority 3	G.729 ▾	
Codec Priority 4	G.711a ▾	
Codec Priority 5	iLBC ▾	
Codec Priority 6	G.722 ▾	
Packet Data Size	20 ms ▾	
iLBC 15.2K	<input checked="" type="radio"/> Off <input type="radio"/> On	
G.723.1 5.3K	<input checked="" type="radio"/> Off <input type="radio"/> On	

Voice VAD/CNG

Voice VAD	<input checked="" type="radio"/> Off <input type="radio"/> On	
CNG	<input checked="" type="radio"/> Off <input type="radio"/> On	

Codec ID Settings

DTMF Payload(RFC2833)	<input type="text" value="101"/>	(95~127)
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Apply Cancel

Select **Call Features** under the **VoIP Settings** section. In this page, a customer can program the memory buttons. For Cetus 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.

Cetus 

SYSTEM SUMMARY
 Model: C32
 WAN IP: 10.33.5.42
 Phone Number: 4309
 Firmware Version: C32-3.0.0-043

Home • VoIP Settings • Call Features

Call Features

Programmable Keys & MWI Number

Memory 1:	Memory	<input type="text"/>
Memory 2:	Memory	<input type="text"/>
Memory 3:	Memory	<input type="text"/>
Memory 4:	Memory	<input type="text"/>
Memory 5:	Memory	<input type="text"/>
Memory 6:	Memory	<input type="text"/>
Memory 7:	Memory	<input type="text"/>
Memory 8:	Memory	<input type="text"/>
Memory 9:	Memory	<input type="text"/>
Memory 10:	Memory	<input type="text"/>
MWI Number:	*17	
Park Mode	Default	
Hold Key Active:	<input type="text"/>	
Hold Key Idle:	<input type="text"/>	

Call Features

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**.

The screenshot shows the Cetis web interface. The top right corner displays system information: Model: C32, WAN IP: 10.33.5.42, Phone Number: 4309, and Firmware Version: C32-3.0.0-043. The left sidebar contains navigation menus for Home, Network Settings, VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area is divided into sections: Call Features, Display Settings, and Blocked List Set.

Call Features

- Hotline: [Empty text box]
- Warm Line Time: 4 (0~30 sec)
- Auto Answer: Off On
- Auto Answer Time Out: 5 (0~30 sec)
- Forward Type: Disable
- Forward Number: [Empty text box]
- Enable Call Time Out: Enable
- No Answer Time Out: 20
- Call Waiting: Off On
- Do Not Disturb: Off On
- Ban Outgoing: Off On
- Accept Any Call: Off On

Display Settings

- LCD Display: Enable Disable
- Greeting Message: [Empty text box]

Buttons: Apply, Cancel

Blocked List Set

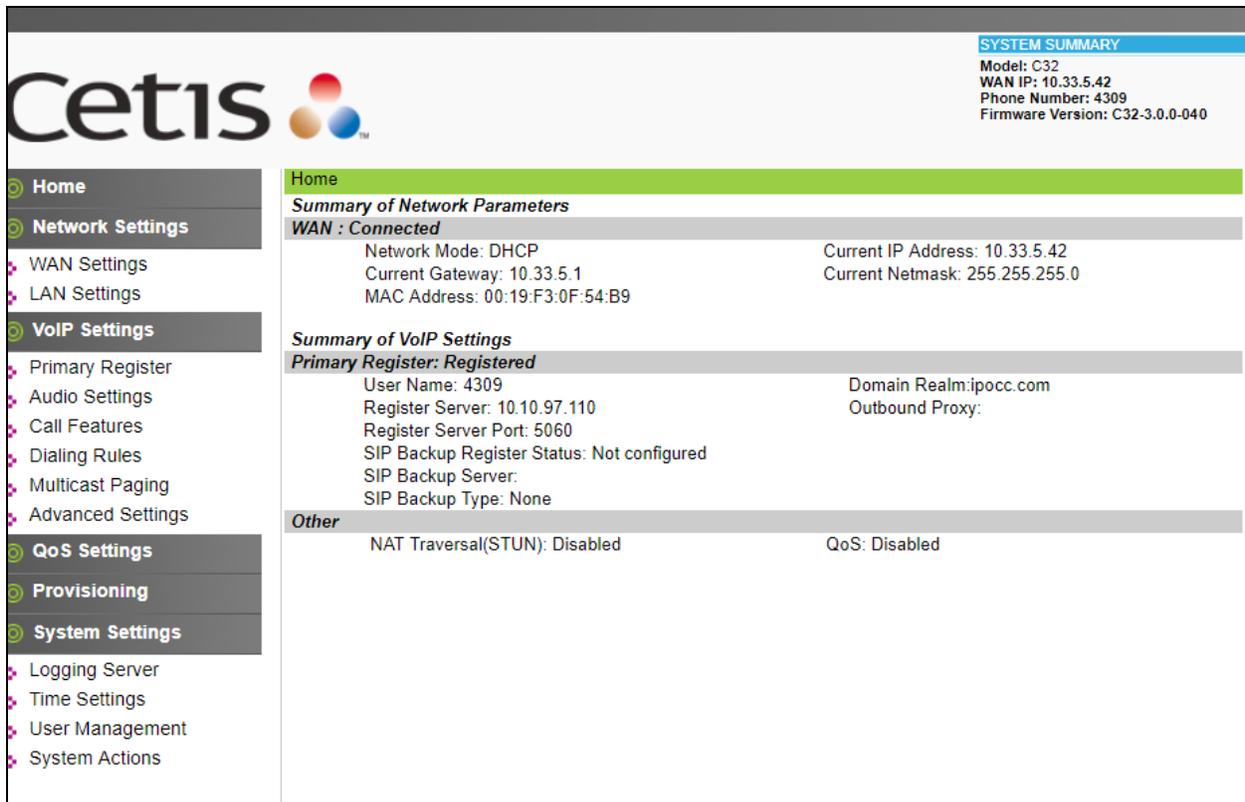
Position	Number	Select
1		<input type="checkbox"/>

7. Verification Steps

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

7.1. Verify Cetus SIP Telephones

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



The screenshot displays the Cetus SIP phone web interface. The top right corner shows the **SYSTEM SUMMARY** with the following details: Model: C32, WAN IP: 10.33.5.42, Phone Number: 4309, and Firmware Version: C32-3.0.0-040. The left navigation pane is expanded to **VoIP Settings**, with **Primary Register** selected. The main content area shows the **Summary of Network Parameters** and **Summary of VoIP Settings**. The **Primary Register** is **Registered**. The **Other** section shows NAT Traversal(STUN) and QoS are both Disabled.

SYSTEM SUMMARY	
Model:	C32
WAN IP:	10.33.5.42
Phone Number:	4309
Firmware Version:	C32-3.0.0-040

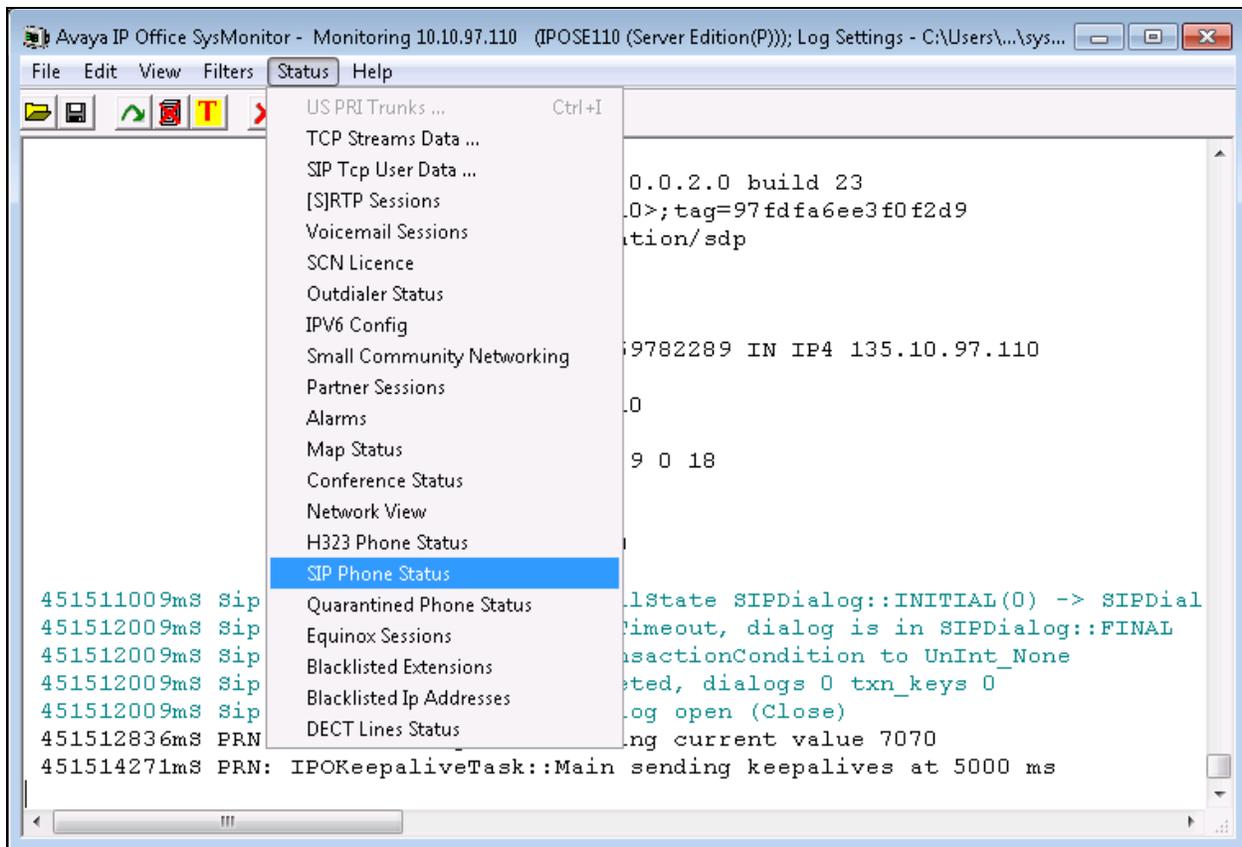
Summary of Network Parameters	
WAN : Connected	
Network Mode:	DHCP
Current Gateway:	10.33.5.1
Current IP Address:	10.33.5.42
Current Netmask:	255.255.255.0
MAC Address:	00:19:F3:0F:54:B9

Summary of VoIP Settings	
Primary Register: Registered	
User Name:	4309
Register Server:	10.10.97.110
Domain Realm:	ipocc.com
Register Server Port:	5060
Outbound Proxy:	
SIP Backup Register Status:	Not configured
SIP Backup Server:	
SIP Backup Type:	None

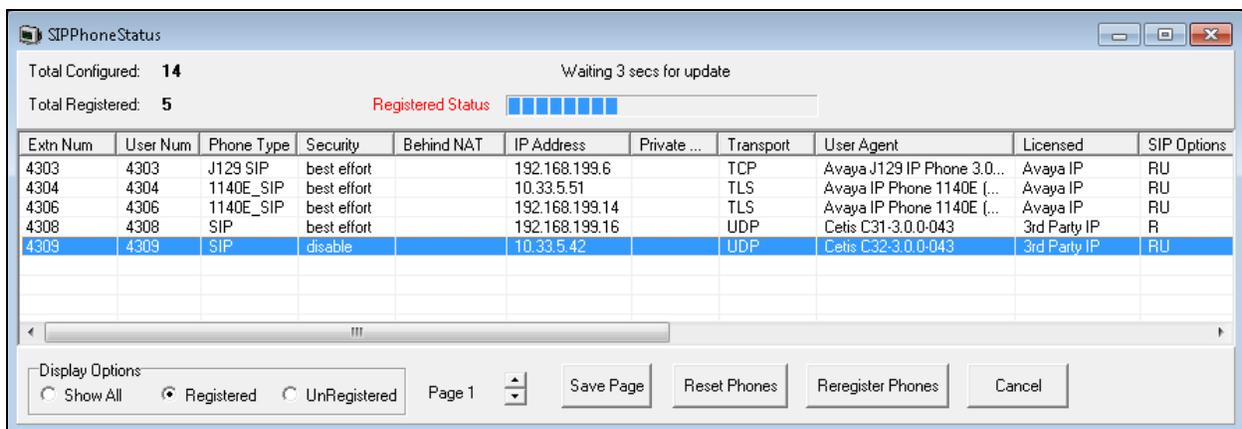
Other	
NAT Traversal(STUN):	Disabled
QoS:	Disabled

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 Cetus C32.3.0.40 in the list.



8. Conclusion

These Application Notes have described the administration steps required to integrate the Cetus 3300IP Series and 9600IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetus 3300IP Series and 9600IP Series SIP Telephones registered successfully with Avaya IP Office via SIP. Incoming and outgoing calls were placed to/from the Cetus 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/>

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