



## **Avaya Solution & Interoperability Test Lab**

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# **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.

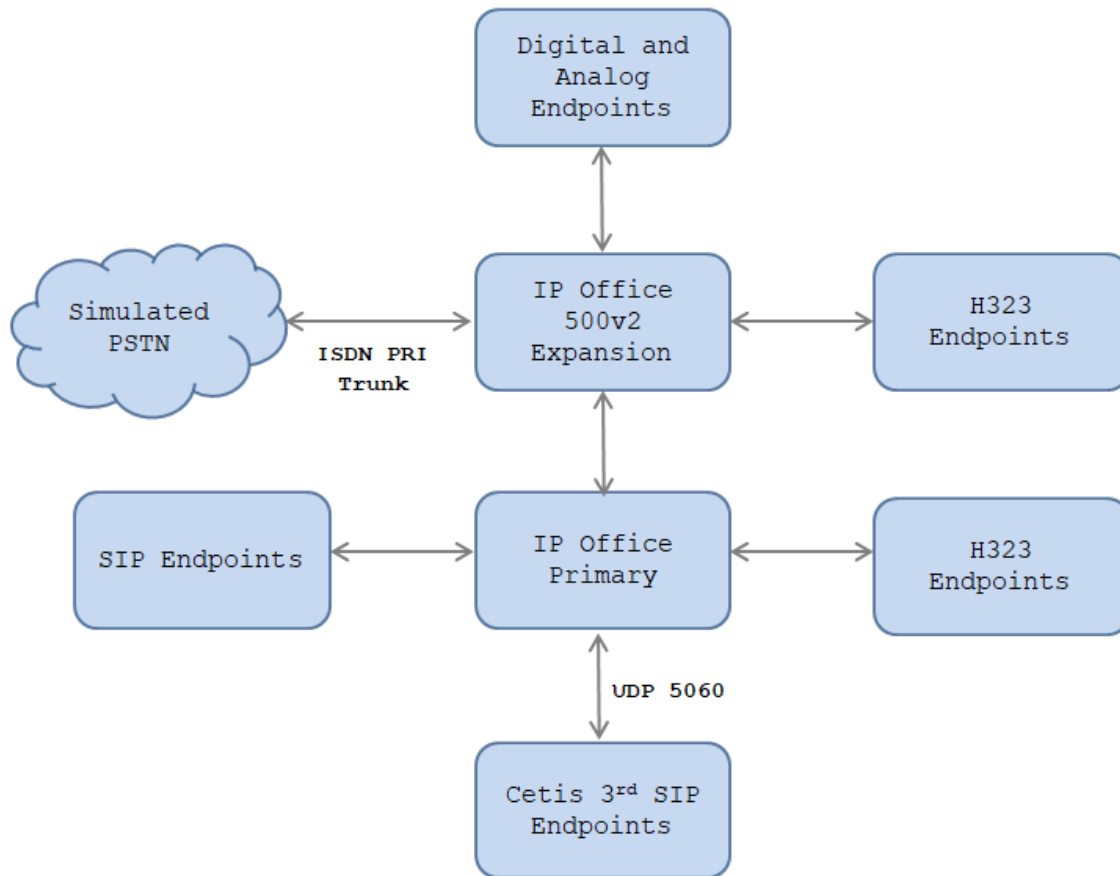
## 2.3. Support

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](mailto:customerservice@cetisgroup.com) or [sipsupport@cetisgroup.com](mailto: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 <sup>rd</sup> -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

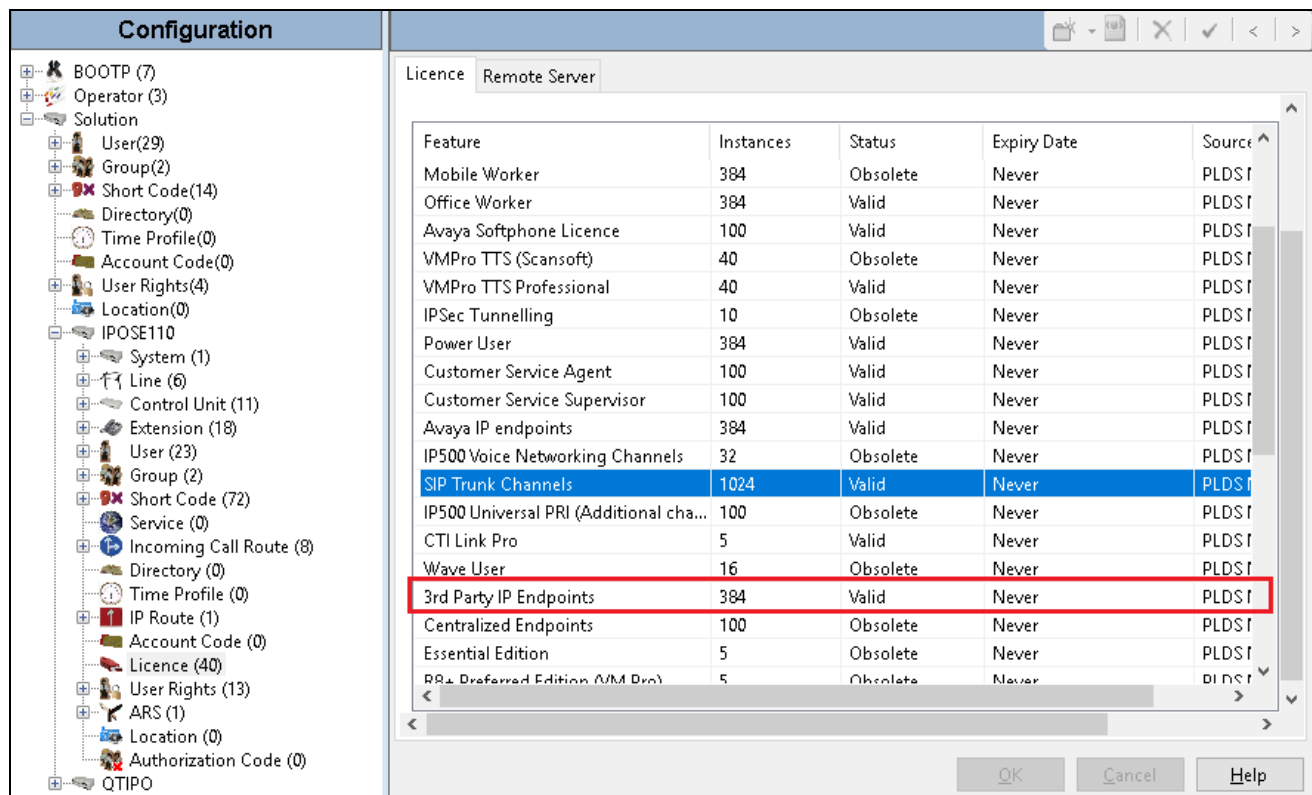
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- 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.

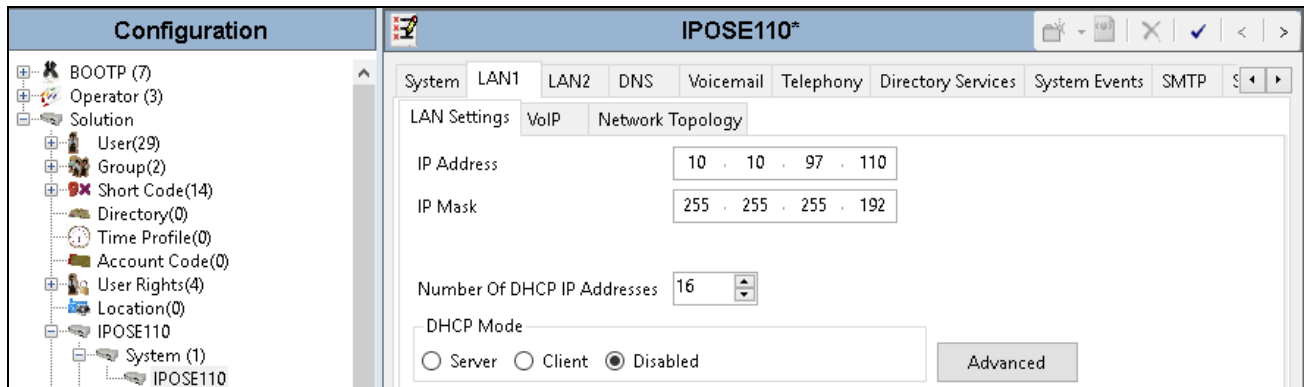
The **Avaya IP Office for Server Edition Manager** screen is displayed. From the configuration tree in the left pane, select **License**. Verify that the **3rd Party IP Endpoints** license is “Valid”, and that the **Instances** value is sufficient for the desired maximum number of simultaneous registrations.



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
RRA Preferred Edition (A/M Pro)	5	Obsolete	Never	PLDS

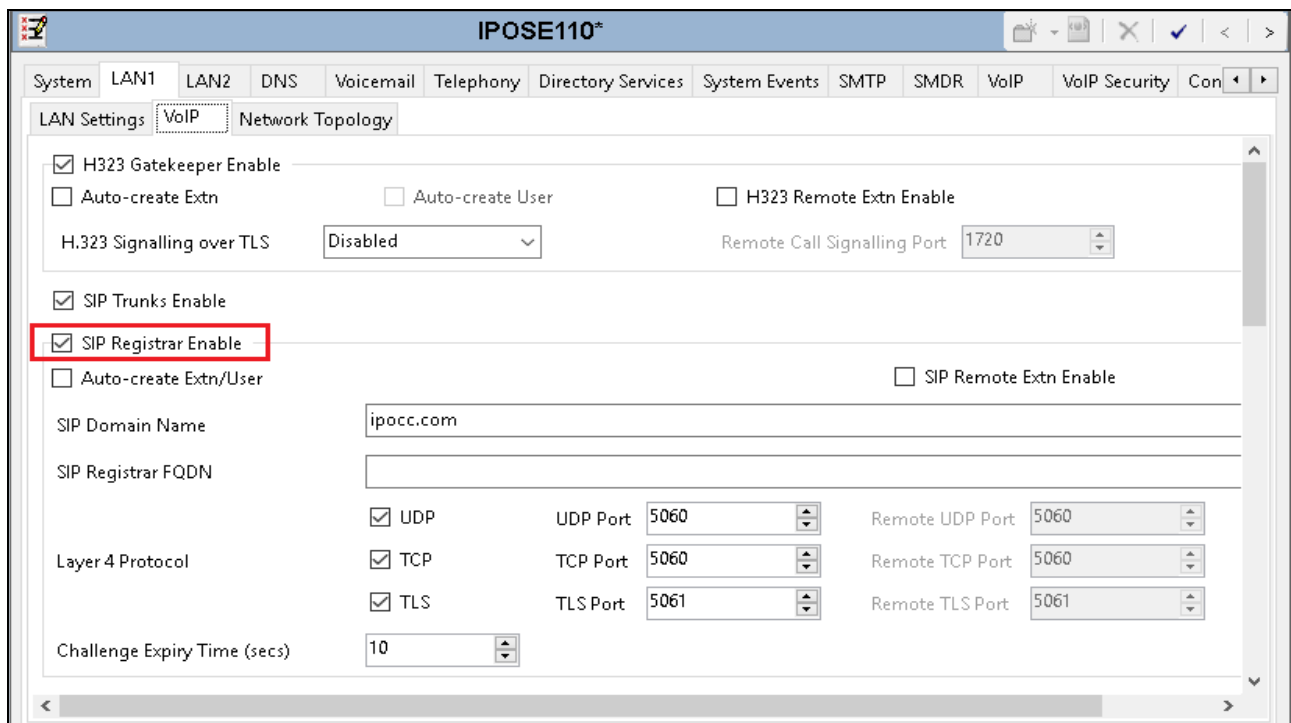
## 5.2. Obtain LAN IP Address

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 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**.

**IPOSE110**

System | **LAN1** | LAN2 | DNS | Voicemail | Telephony | Directory Services | System Events | SMTP | SMDR | VoIP | VoIP Security | Contact Center

LAN Settings | **VoIP** | Network Topology

☒ H.323 Gatekeeper Enable  
☐ Auto-create Extension ☐ Auto-create User ☐ H.323 Remote Extension Enable  
H.323 Signaling over TLS: Preferred Remote Call Signaling Port: 1720

☒ SIP Trunks Enable  
☒ SIP Registrar Enable  
☐ Auto-create Extension/User ☐ SIP Remote Extension Enable  
SIP Domain Name: ipocc.com  
SIP Registrar FQDN:   
Layer 4 Protocol: ☒ UDP UDP Port: 5060 Remote UDP Port: 5060  
☒ TCP TCP Port: 5060 Remote TCP Port: 5060  
☒ TLS TLS Port: 5061 Remote TLS Port: 5061  
Challenge Expiration Time (sec): 10

**RTP**  
Port Number Range  
Minimum: 40750 Maximum: 50750  
Port Number Range (NAT)  
Minimum: 40750 Maximum: 50750  
☒ Enable RTCP Monitoring on Port 5005  
RTCP collector IP address for phones: 0 . 0 . 0 . 0  
Keepalives  
Scope: Disabled Periodic timeout: 0  
Initial keepalives: Disabled

**DiffServ Settings**  
B8 DSCP(Hex) B8 Video DSCP (Hex) FC DSCP Mask (Hex) 88 SIG DSCP (Hex)  
46 DSCP 46 Video DSCP 63 DSCP Mask 34 SIG DSCP



## 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 expanded, 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 'Ring Delay (sec)' set to 5. The 'Login Code Complexity' section has 'Enforcement' and 'Complexity' checked. The 'RTCP Collector Configuration' section has 'Send RTCP to an RTCP Collector' unchecked.

Setting	Value
Dial Delay Time (sec)	4
Dial Delay Count	0
Default No Answer Time (sec)	15
Hold Timeout (sec)	0
Park Timeout (sec)	300
Ring Delay (sec)	5
Call Priority Promotion Time (sec)	Disabled
Default Currency	USD
Default Name Priority	Favor Trunk
Media Connection Preservation	Enabled
Phone Failback	Automatic
Login Code Complexity	Enforcement: <input checked="" type="checkbox"/> Minimum length: 4 Complexity: <input checked="" type="checkbox"/>
RTCP Collector Configuration	Send RTCP to an RTCP Collector: <input type="checkbox"/> Server Address: 0.0.0.0 UDP Port Number: 5005 RTCP reporting interval (sec): 5
Companding Law	Switch: <input checked="" type="radio"/> U-Law <input type="radio"/> A-Law Line: <input checked="" type="radio"/> U-Law Line <input type="radio"/> A-Law Line
Other Settings	<input type="checkbox"/> DSS Status <input type="checkbox"/> Auto Hold <input checked="" type="checkbox"/> Dial By Name <input checked="" type="checkbox"/> Show Account Code <input type="checkbox"/> Inhibit Off-Switch Forward/Transfer <input type="checkbox"/> Restrict Network Interconnect <input type="checkbox"/> Include location specific information <input checked="" type="checkbox"/> Drop External Only Impromptu Conference <input type="checkbox"/> Visually Differentiate External Call <input checked="" type="checkbox"/> High Quality Conferencing <input checked="" type="checkbox"/> Directory Overrides Barring <input type="checkbox"/> Advertise Callee State To Internal Callers <input type="checkbox"/> Internal Ring on Transfer

## 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).

The screenshot displays the IPOSE110 configuration window with the 'Voicemail' tab selected. The 'Ignore DTMF Mismatch For Phones' checkbox is checked, and 'Allow Direct Media Within NAT Location' is unchecked. The 'RFC2833 Default Payload' is set to 101. The 'Available Codecs' section lists four codecs with checkboxes: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, and G.729(a) 8K CS-ACELP. The 'Default Codec Selection' section contains two lists: 'Unused' (G.711 ALAW 64K, G.722 64K) and 'Selected' (G.711 ULAW 64K, G.729(a) 8K CS-ACELP). Navigation buttons (horizontal and vertical arrows) are positioned between the lists.

IPOSE110						
System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services
<p>Ignore DTMF Mismatch For Phones <input checked="" type="checkbox"/></p> <p>Allow Direct Media Within NAT Location <input type="checkbox"/></p> <p>RFC2833 Default Payload: 101</p> <div><div><p>Available Codecs</p><ul style="list-style-type: none"><li><input checked="" type="checkbox"/> G.711 ULAW 64K</li><li><input checked="" type="checkbox"/> G.711 ALAW 64K</li><li><input checked="" type="checkbox"/> G.722 64K</li><li><input checked="" type="checkbox"/> G.729(a) 8K CS-ACELP</li></ul></div><div><p>Default Codec Selection</p><div><p>Unused</p><ul style="list-style-type: none"><li>G.711 ALAW 64K</li><li>G.722 64K</li></ul></div><div><p>Selected</p><ul style="list-style-type: none"><li>G.711 ULAW 64K</li><li>G.729(a) 8K CS-ACELP</li></ul></div></div></div>						

## 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.

The screenshot shows the 'SIP Extension: 11213 4309' configuration window. The left pane displays a configuration tree with 'Extension (21)' selected, showing a list of extensions from 11214 4300 to 11218 4322. The right pane contains the following fields:

Field	Value
Extension ID	11213
Base Extension	4309
Phone Password	
Confirm Phone Password	
Caller Display Type	On
Reset Volume After Calls	<input type="checkbox"/>
Device Type	Unknown SIP device
Location	Automatic
Fallback As Remote Worker	Auto
Module	0
Port	0

At the bottom of the right pane are buttons for 'OK', 'Cancel', and 'Help'. The status bar at the bottom left indicates 'Ready'.

Select the **VoIP** tab and retain the default values in the all fields. During the compliance test, Cetis 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 Cetis SIP telephone that does not support the media security.

The screenshot shows the 'SIP Extension: 11213 4309' configuration window. The 'VoIP' tab is selected. The 'IP Address' field is set to '0 . 0 . 0 . 0'. The 'Codec Selection' dropdown is set to 'System Default'. Below this, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ALAW 64K'. The 'Selected' list contains 'G.711 ULAW 64K', 'G.729(a) 8K CS-ACELP', and 'G.722 64K'. Between these lists are buttons for moving items: '>>>', '<<<', '<<<', '>>>', and '<<<'. To the right of these lists are checkboxes for 'Requires DTMF', 'Local Hold Music', 'Re-invite Supported' (checked), 'Codec Lockdown', and 'Allow Direct Media Path' (checked). Below these are dropdown menus for 'Reserve License' (None), 'Fax Transport Support' (None), 'DTMF Support' (RFC2833/RFC4733), '3rd Party Auto Answer' (None), and 'Media Security' (Disabled). At the bottom are 'OK', 'Cancel', and 'Help' buttons.

Field	Value
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
Requires DTMF	<input type="checkbox"/>
Local Hold Music	<input type="checkbox"/>
Re-invite Supported	<input checked="" type="checkbox"/>
Codec Lockdown	<input type="checkbox"/>
Allow Direct Media Path	<input checked="" type="checkbox"/>
Reserve License	None
Fax Transport Support	None
DTMF Support	RFC2833/RFC4733
3rd Party Auto Answer	None
Media Security	Disabled

## 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.

4309: 4309\*

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

Name: 4309

Password: ••••••••

Confirm Password: ••••••••

Unique Identity:

Conference PIN:

Confirm Audio Conference PIN:

Account Status: Enabled

Full Name: SIP 3RD 4309

Extension: 4309

Email Address:

Locale:

Priority: 5

System Phone Rights: None

Profile: Basic User

☐ 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: Unknown SIP device

User Rights

User Rights view: User data

Working hours time profile: <None>

Working hours User Rights:

Out of hours User Rights:

OK Cancel Help

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

The screenshot shows the 'Voicemail' configuration window for user '4309: 4309\*'. The window has several tabs: User, Voicemail (selected), DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The 'Voicemail' tab contains the following settings:

- Voicemail Code: [Text field with 6 dots]
- Confirm Voicemail Code: [Text field with 6 dots]
- Voicemail Email: [Text field]
- Voicemail On: ☒ (checked)
- Voicemail Help: ☐
- Voicemail Ringback: ☐
- Voicemail Email Reading: ☐
- UMS Web Services: ☐
- Enable GMAIL API: ☐
- Voicemail Email: [Text field]
- Off: ☒ (selected), Copy: ☐, Forward: ☐, Alert: ☐
- DTMF Breakout:
  - Reception/Breakout (DTMF 0): [System Default ()]
  - Breakout (DTMF 2): [System Default ()]
  - Breakout (DTMF 3): [System Default ()]

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

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.

The screenshot shows the 'Call Settings' sub-tab within the 'Telephony' configuration window for user '4309: 4309\*'. The window has several tabs: User, Voicemail, DND, Short Codes, Source Numbers, Telephony (selected), Forwarding, Dial In, Voice Recording, and Button Programming. The 'Call Settings' sub-tab contains the following settings:

- Outside Call Sequence: [Default Ring]
- Inside Call Sequence: [Default Ring]
- Ringback Sequence: [Default Ring]
- No Answer Time (sec): [System Default (15)]
- Wrap-Up Time (sec): [2]
- Transfer Return Time (sec): [Off]
- Call Cost Mark-Up: [100]
- Advertise Callee State To Internal Callers: [System Default (Off)]
- Call Waiting On: ☒ (checked)
- Answer Call Waiting On Hold: ☒ (checked)
- Busy On Hold: ☐
- Off-hook Station: ☐

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

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.

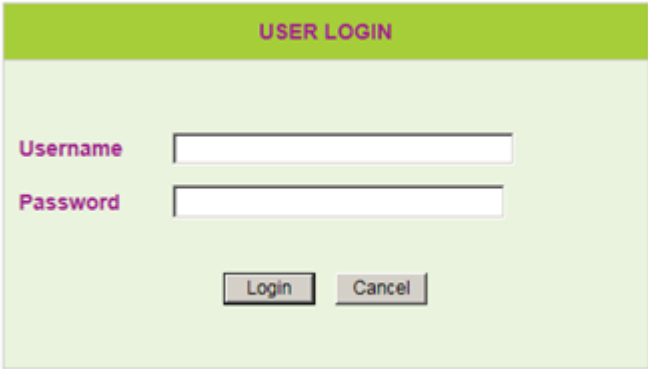
The screenshot shows a configuration window titled "4309: 4309\*" with a standard Windows-style title bar. The window contains a series of tabs: "User", "Voicemail", "DND", "Short Codes", "Source Numbers", "Telephony", "Forwarding", "Dial In", "Voice Recording", and "Button Programming". The "Supervisor Settings" sub-tab is selected under the "Telephony" tab. The sub-tab contains the following settings:

Field	Value	Field	Value
Login Code	••••••	Force Login	<input type="checkbox"/>
Confirm Login Code	••••••	Force Account Code	<input type="checkbox"/>
Login Idle Period (sec)		Force Authorization Code	<input type="checkbox"/>
Monitor Group	<None>	Incoming Call Bar	<input type="checkbox"/>
Coverage Group	<None>	Outgoing Call Bar	<input type="checkbox"/>
Status on No-Answer	Logged On (No change)	Inhibit Off-Switch Forward/Transfer	<input type="checkbox"/>
IPOCC Agent Type	<None>	Can Intrude	<input type="checkbox"/>
Privacy Override Group	<None>	Cannot Be Intruded	<input checked="" type="checkbox"/>
Reset Longest Idle Time		Can Trace Calls	<input type="checkbox"/>
All Calls	<input checked="" type="radio"/>	Deny Auto Intercom Calls	<input type="checkbox"/>
External Incoming	<input type="radio"/>		

At the bottom of the window are three buttons: "OK", "Cancel", and "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 screenshot displays the 'USER LOGIN' web interface. It features a light green background with a darker green header bar containing 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 header features the Cetis logo and 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, Provisioning, and System Settings. Under 'Network Settings', 'WAN Settings' and 'LAN Settings' are listed. The main content area shows the 'Summary of Network Parameters' for the 'WAN : Connected' status. 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' status. It lists 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 shows NAT Traversal(STUN): Disabled and QoS: Disabled.

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

- 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

### Summary of Network Parameters

**WAN : Connected**

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

### Summary of VoIP Settings

**Primary Register: Registered**

User Name: 4309	Domain Realm: ipocc.com
Register Server: 10.10.97.110	Outbound Proxy:
Register Server Port: 5060	
SIP Backup Register Status: Not configured	
SIP Backup Server:	
SIP Backup Type: None	

**Other**

NAT Traversal(STUN): Disabled	QoS: Disabled
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**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 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 | CD = Cetis DECT/Cordless | CT/CM = Cetis Trimline | 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 | 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 C32 web interface for WAN Settings. The left sidebar contains navigation links: Home, Network Settings (selected), VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area is titled 'WAN Settings' and shows the 'WAN Interface: Connected' status. The 'Basic Settings' section includes 'Network Mode' set to DHCP (selected), 'Link Mode' set to AUTO, 'Primary DNS' set to 10.10.98.60, and 'Secondary DNS' set to 10.10.98.60. The 'Static IP Settings' section is required if Network Mode is set to Static IP, showing 'Static IP Address' as 10.33.5.200, 'Subnet Mask' as 255.255.255.0, and 'Default Gateway' as 10.33.5.1. The 'PPPoE Settings' section is required if Network Mode is set to PPPoE, showing 'User Account' and 'Password' fields. The '802.1X Settings' section shows '802.1X' set to Disable, 'User Name' and 'Password' fields, and 'Type' set to multicast. The 'LLDP Settings' section shows 'LLDP' set to Enable and 'Packet Interval' set to 120. The 'Apply' and 'Cancel' buttons are at the bottom right.

**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

**WAN Settings**  
*WAN Interface: Connected*

**Basic Settings**

Network Mode ☒ DHCP ☐ Fixed ☐ PPPoE  
Link Mode   
Primary DNS   
Secondary DNS

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

Static IP Address   
Subnet Mask   
Default Gateway

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

User Account   
Password

**802.1X Settings**

802.1X   
User Name   
Password   
Type

**LLDP Settings**

LLDP   
Packet Interval

## 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 shows the Cetis web interface for configuring VoIP settings. The left sidebar contains a navigation menu with categories: Home, Network Settings, VoIP Settings, QoS Settings, Provisioning, and System Settings. Under VoIP Settings, the 'Primary Register' option is selected. The main content area is titled 'Primary Register' and shows the 'Register Server' configuration. The 'Main Server' is 'Registered' and the 'Backup Server' is 'Not configured'. The configuration fields are as follows:

Register Server	
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 top right corner, a 'SYSTEM SUMMARY' box displays the following information:

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

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.

The screenshot displays the Cetis web interface. On the left is a navigation menu with categories: Home, Network Settings, VoIP Settings, QoS Settings, Provisioning, and System Settings. The 'Protocol Control' section is highlighted. The main area shows various settings for SIP Backup, Protocol Control, and other features. The 'Protocol Control' section is expanded, showing settings for MWI Subscribe, DTMF, SIP Transport, etc. The 'Apply' button is visible at the bottom right of the settings area.


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

Category	Setting	Value
SIP Backup	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	<input type="radio"/> Off <input checked="" type="radio"/> On
	Keep Alives Period	60
	DTMF	<input checked="" type="radio"/> RFC2833 <input type="radio"/> Inband <input type="radio"/> SIP Info
	DTMF SIP INFO Mode	Send */#
	DNS Type	NAPTR/SRV
	Jitter Buffer Max	150
	Anonymous Call Rejection	<input checked="" type="radio"/> Off <input type="radio"/> 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.



**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

Home • VoIP Settings • Audio Settings

### Audio Settings

Sound and Volume Control

Handset	7	(1~7)
Speaker	5	(1~7)
Ringer Tone	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)	101	(95~127)
-----------------------	-----	----------

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.



**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

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"/>	

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**.



**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

**Call Features**

Hotline	<input type="text" value=""/>
Warm Line Time	<input type="text" value="4"/> (0~30 sec)
Auto Answer	<input checked="" type="radio"/> Off <input type="radio"/> On
Auto Answer Time Out	<input type="text" value="5"/> (0~30 sec)
Forward Type	Disable ▾
Forward Number	<input type="text" value=""/>
Enable Call Time Out	Enable ▾
No Answer Time Out	<input type="text" value="20"/>
Call Waiting	<input type="radio"/> Off <input checked="" type="radio"/> On
Do Not Disturb	<input checked="" type="radio"/> Off <input type="radio"/> On
Ban Outgoing	<input checked="" type="radio"/> Off <input type="radio"/> On
Accept Any Call	<input type="radio"/> Off <input checked="" type="radio"/> On

**Display Settings**

LCD Display	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Greeting Message	<input type="text" value=""/>

**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 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*.

The screenshot displays the Cetis SIP Phone Web Interface. The left sidebar contains a navigation menu with the following items: Home, Network Settings (WAN Settings, LAN Settings), VoIP Settings (Primary Register, Audio Settings, Call Features, Dialing Rules, Multicast Paging, Advanced Settings), QoS Settings, Provisioning, and System Settings (Logging Server, Time Settings, User Management, System Actions). The main content area shows the 'VoIP Summary' page. At the top right, a 'SYSTEM SUMMARY' box lists: Model: C32, WAN IP: 10.33.5.42, Phone Number: 4309, and Firmware Version: C32-3.0.0-040. The 'Summary of Network Parameters' section shows 'WAN : Connected' with details: 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. The 'Summary of VoIP Settings' section shows 'Primary Register: Registered' with details: 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 shows 'NAT Traversal(STUN): Disabled' and 'QoS: Disabled'.

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

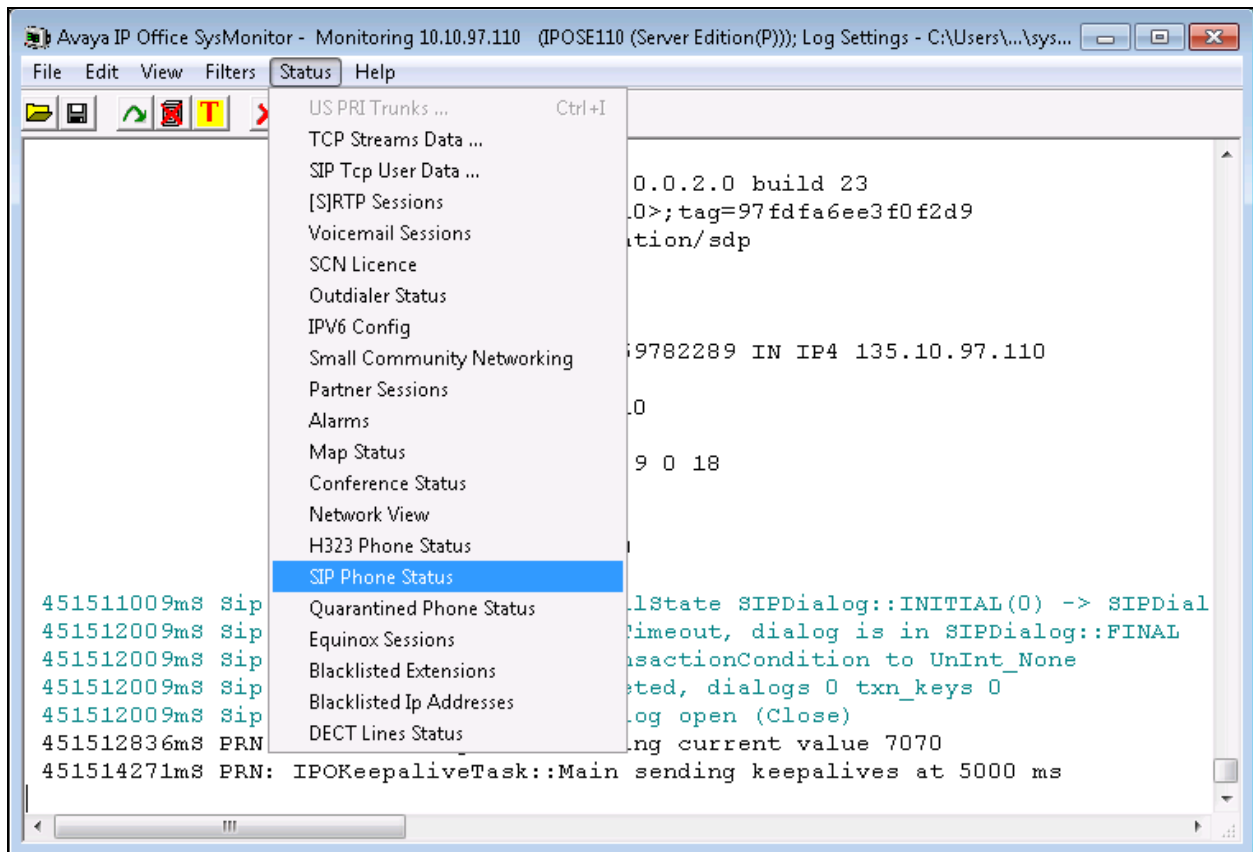
Summary of Network Parameters	
WAN : Connected	
Network Mode:	DHCP
Current Gateway:	10.33.5.1
MAC Address:	00:19:F3:0F:54:B9
Current IP Address:	10.33.5.42
Current Netmask:	255.255.255.0

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

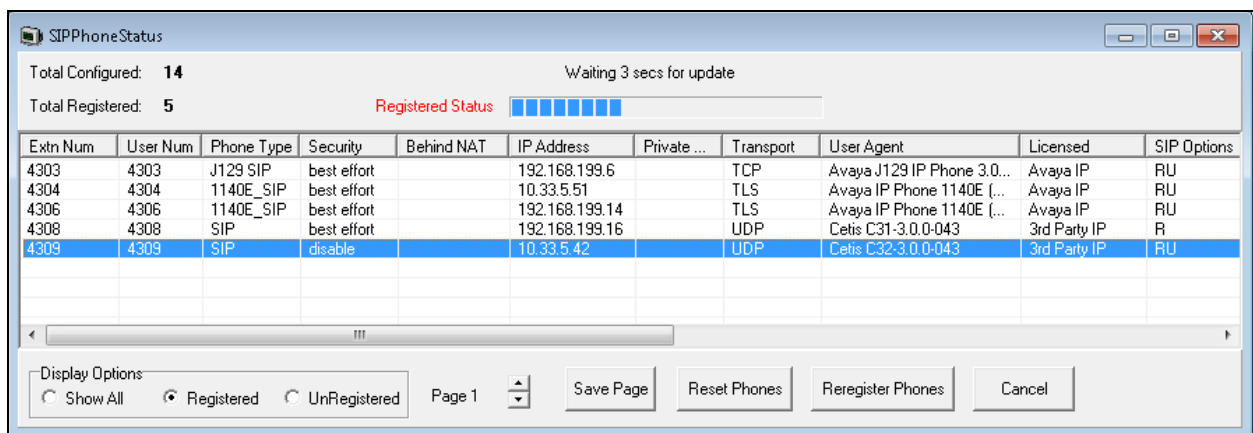
Other	
NAT Traversal(STUN):	Disabled
QoS:	Disabled

### 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 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/>

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