



Avaya Solution & Interoperability Test Lab

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

Abstract

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

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps required to integrate the Cetus 3500IP Series and 9700IP Series SIP Telephones with Avaya IP Office Server Edition. The Cetus 3500IP Series and 9700IP Series SIP Telephones were designed for the hospitality industry. In the compliance test, Cetus SIP telephones registered with Avaya IP Office Server Edition.

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

2. General Test Approach and Test Results

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

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

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and the Cetus SIP telephones does 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 3500IP Series and 9700IP Series SIP telephones and Avaya SIP and H.323 telephones and exercising basic telephony features, such as hold, mute, transfer and conference. In addition, hospitality features, such as call forward and do

not disturb were covered. Interoperability compliance testing covered the following features and functionality:

- SIP registration of Cetus 3500IP and 9700IPSIP telephones with IP Office.
- 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 TCP and UDP.
- Proper recognition of DTMF tones.
- Basic telephony features, including inbound/outbound, hold, mute, call forward, transfer and conference.
- Use of programmable buttons on the 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 3500IP Series and 9700IP Series SIP telephones come back into service after re-connecting the Ethernet connect or rebooting the phone.

2.2. Test Results

All test cases passed with the following issue noted:

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

2.3. Support

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

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

3. Reference Configuration

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

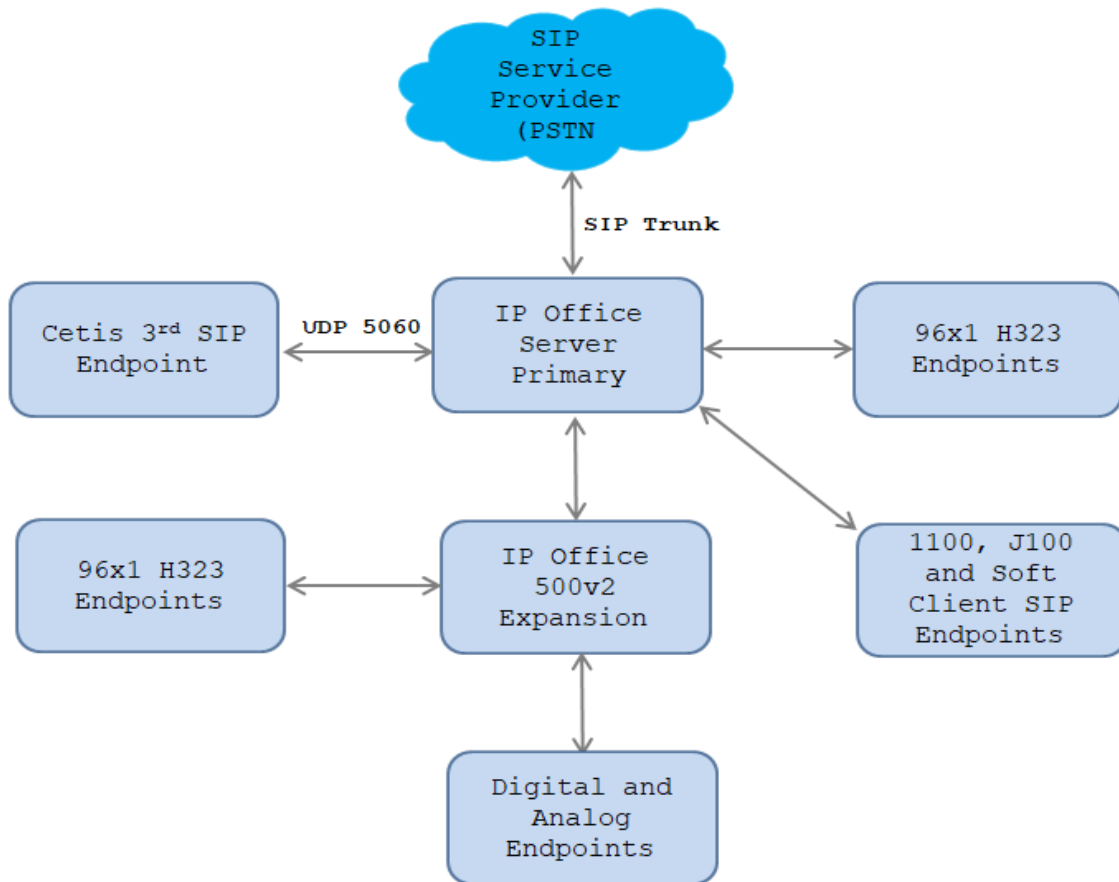


Figure 1: Test Configuration Diagram with IP Office

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

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

4. Equipment and Software Validated

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

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

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office

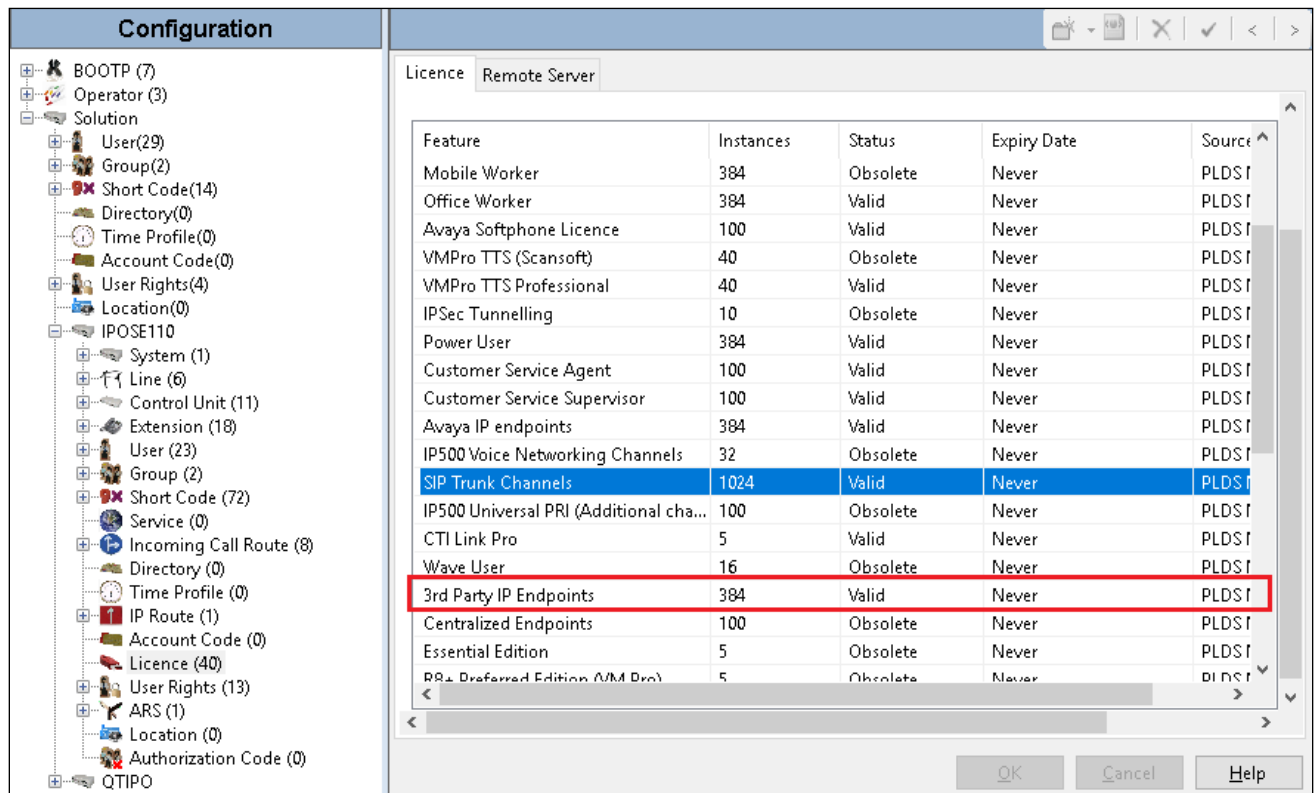
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

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

5.1. Verify Avaya IP Office License

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

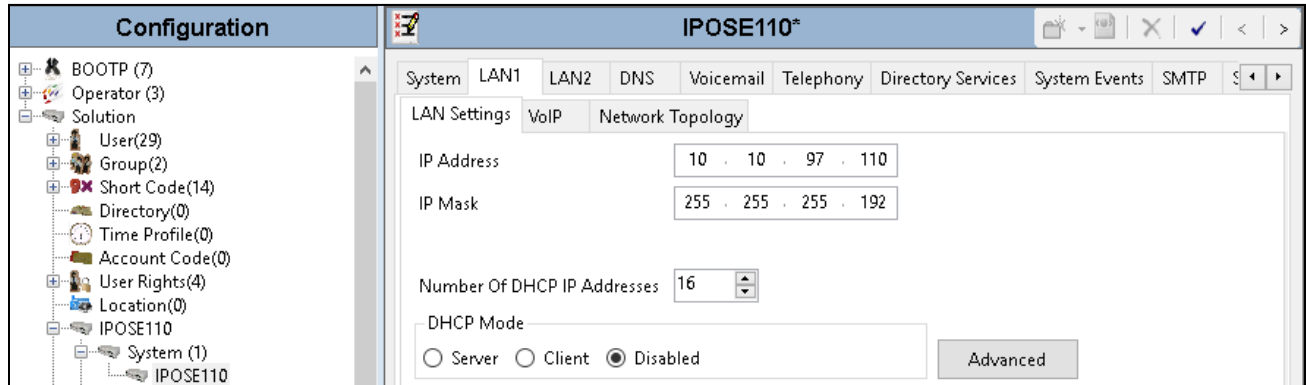
The **Avaya IP Office Manager** screen is displayed. From the configuration tree in the left pane, select **License**. Verify that the **3rd Party IP Endpoints** license is “Valid”, and that the **Instances** value is sufficient for the desired maximum number of simultaneous registrations.



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
VMPPro TTS (Scansoft)	40	Obsolete	Never	PLDS
VMPPro 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
PR. Preferred Edition (VMP Pro)	5	Obsolete	Never	PLDS

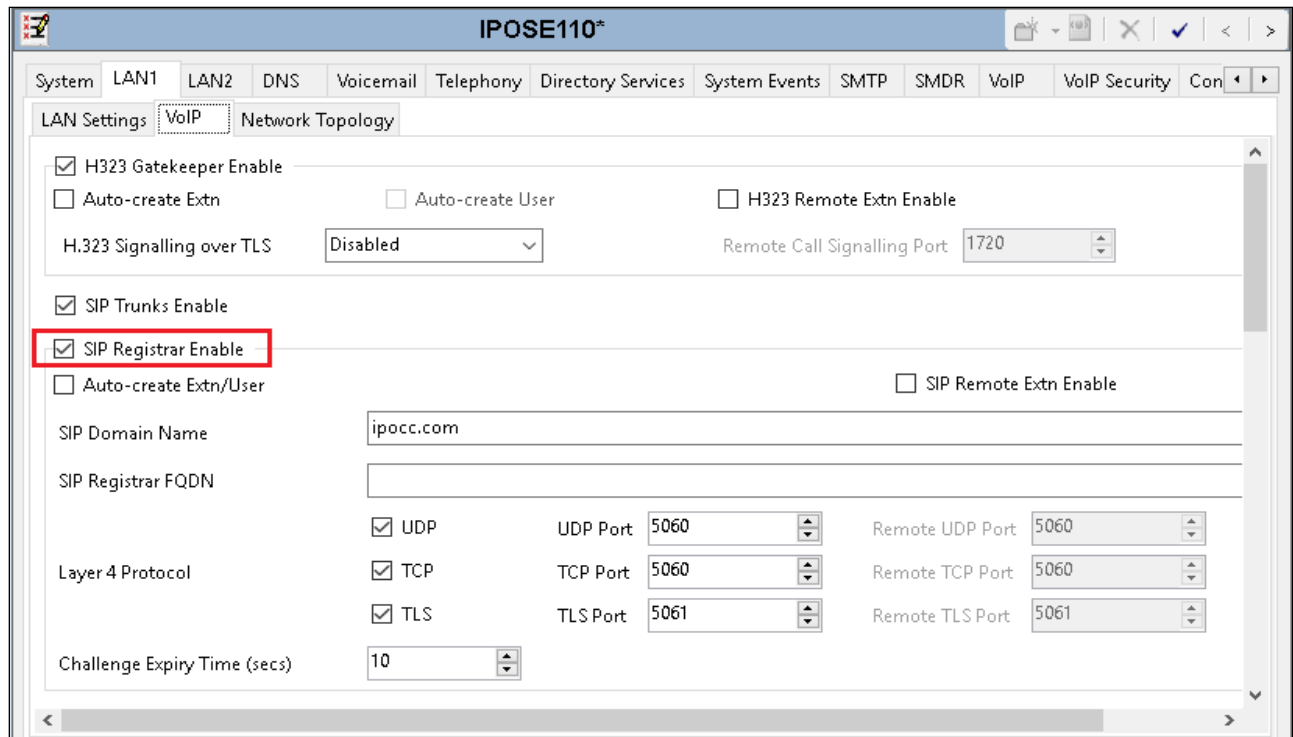
5.2. Obtain LAN IP Address

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 endpoints.



5.3. Enable SIP Registrar

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



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

The screenshot shows the IPOSE110 configuration interface for LAN1. The 'VoIP' tab is selected, and the 'RTP' section is expanded. The 'Port Number Range' is set to Minimum: 40750 and Maximum: 50750. The 'Port Number Range (NAT)' is also set to Minimum: 40750 and Maximum: 50750. Other settings include 'H.323 Gatekeeper Enable' checked, 'SIP Trunks Enable' checked, 'SIP Registrar Enable' checked, and 'SIP Domain Name' set to ipocc.com. The 'Layer 4 Protocol' section has UDP, TCP, and TLS checked, with ports 5060 and 5061 configured. The 'RTCP collector IP address for phones' is set to 0.0.0.0. The 'DiffServ Settings' section shows various DSCP and Video DSCP values.

5.4. System Telephony Settings

Navigate to the **Telephony** → **Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, **U-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN. On completion, click the **OK** button (not shown).

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 'Media Connection Preservation' set to 'Enabled'.

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

Companding Law

Switch	Line
<input checked="" type="radio"/> U-Law	<input checked="" type="radio"/> U-Law Line
<input type="radio"/> A-Law	<input type="radio"/> A-Law Line

DSS Status

Auto Hold

Dial By Name

Show Account Code

Inhibit Off-Switch Forward/Transfer

Restrict Network Interconnect

Include location specific information

Drop External Only Impromptu Conference

Visually Differentiate External Call

High Quality Conferencing

Directory Overrides Barring

Advertise Callee State To Internal Callers

Internal Ring on Transfer

Login Code Complexity

Enforcement

Minimum length: 4

Complexity

RTCP Collector Configuration

Send RTCP to an RTCP Collector

Server Address: 0 . 0 . 0 . 0

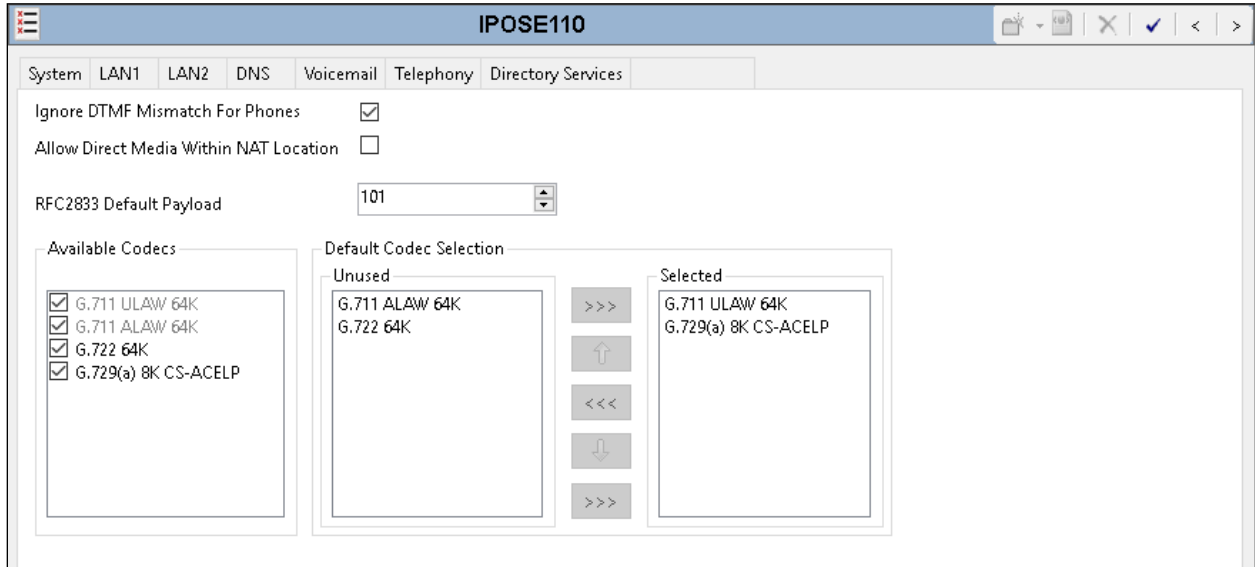
UDP Port Number: 5005

RTCP reporting interval (sec): 5

Buttons: OK, Cancel, Help

5.5. Administer Codec Settings

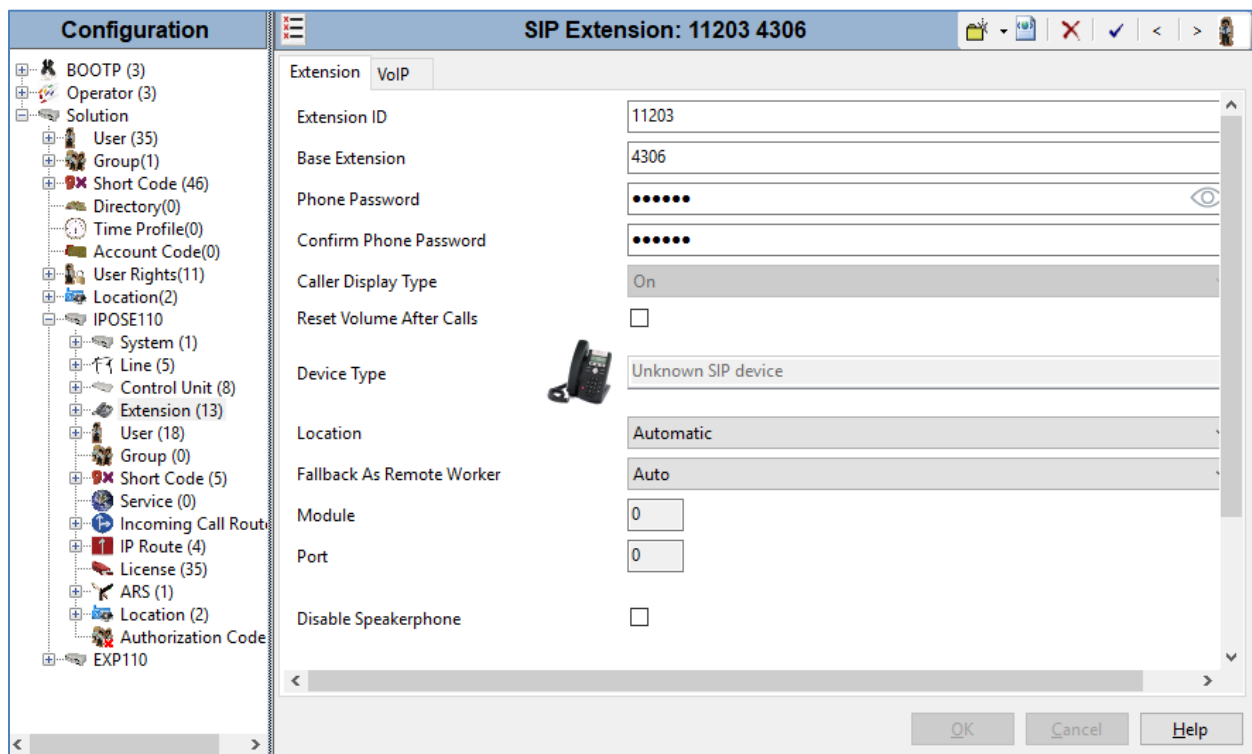
Navigate to the **VoIP** tab on the Details Pane. Check the **Available Codecs** boxes as required for the IP endpoints. Note that **G.711 ULAW 64K** and **G.711 ALAW 64K** are greyed out and always available. Once available codecs are selected, they can be used or unused by using the horizontal arrows 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).



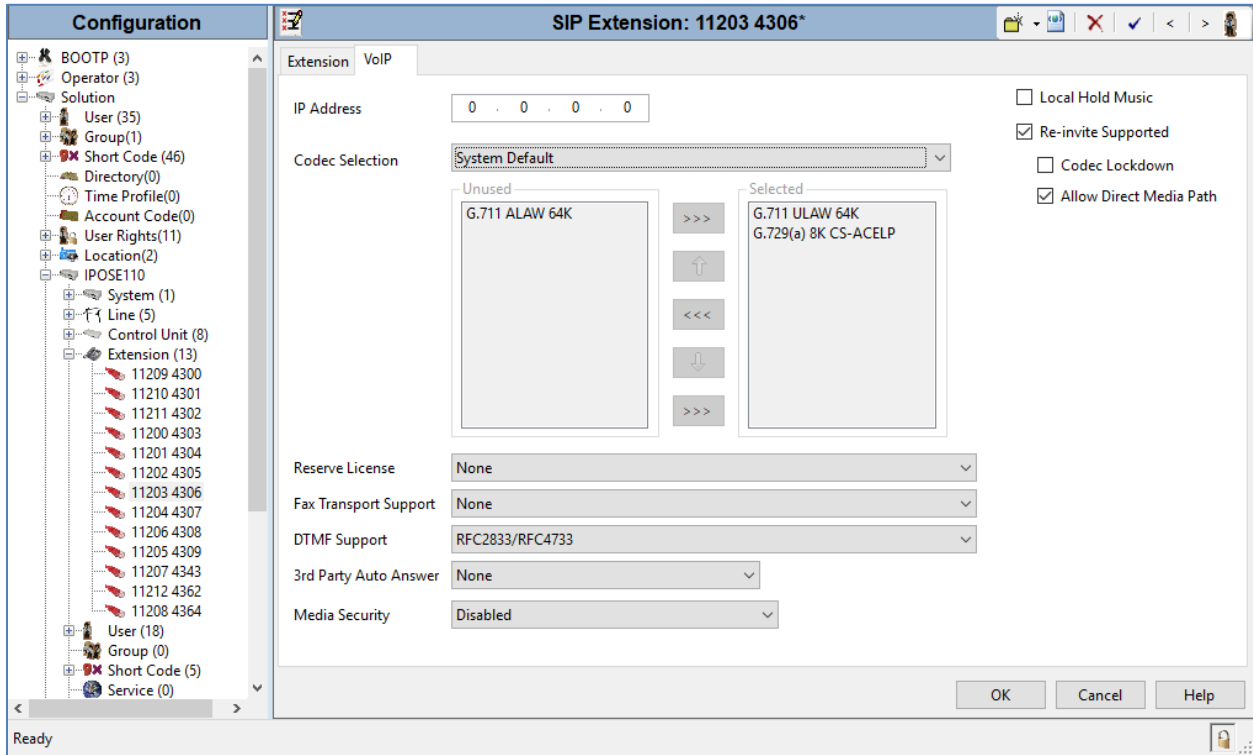
5.6. Administer Extension for Cetus SIP Endpoint

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

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



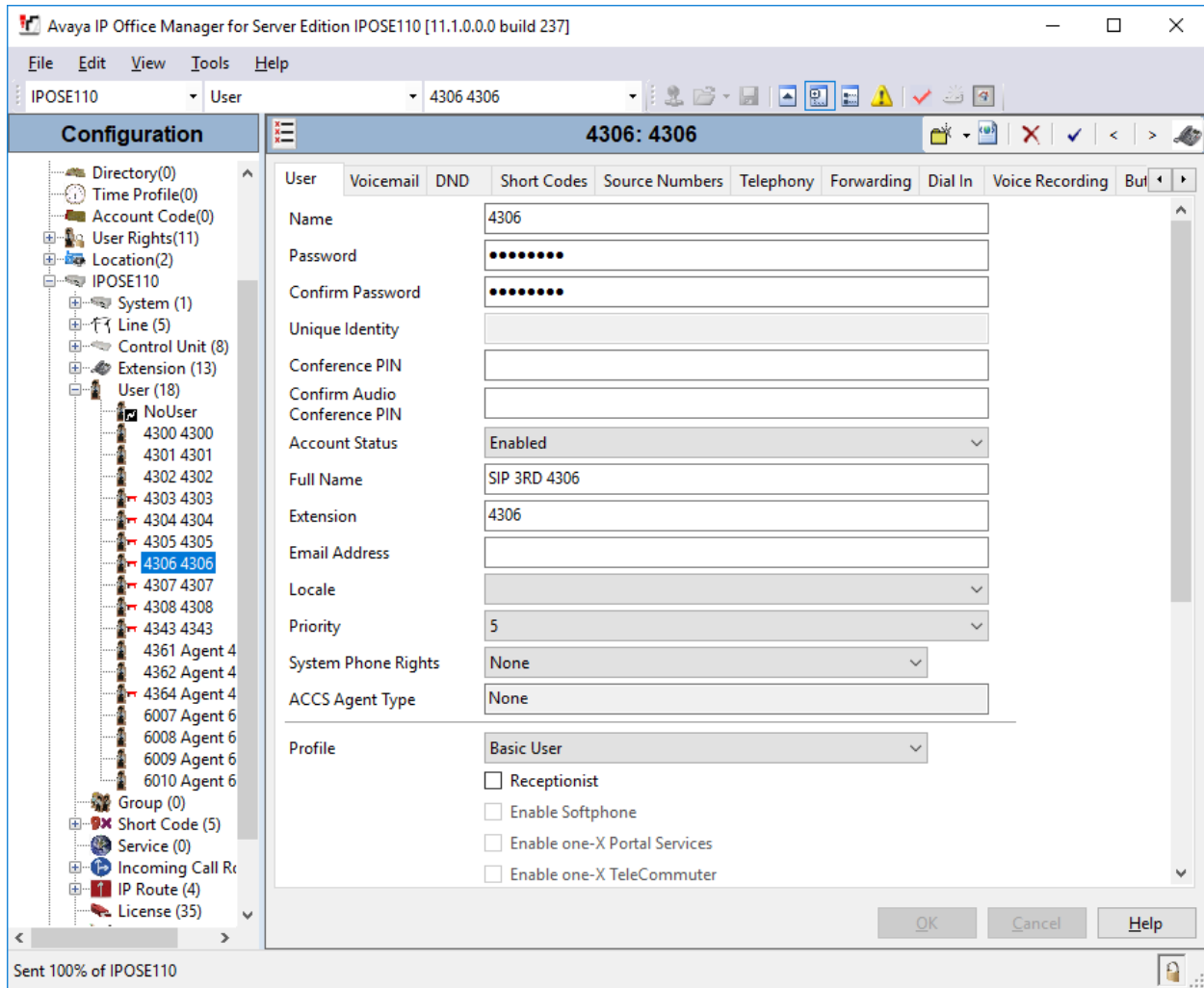
Select the **VoIP** tab and retain the default values in the all fields. During the compliance test, Cetus SIP endpoint was tested with G.711 and G.729 codecs. Enable **Allow Direct Media Path** so that audio/RTP flows directly between two SIP endpoints without using media resources in Avaya IP Office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



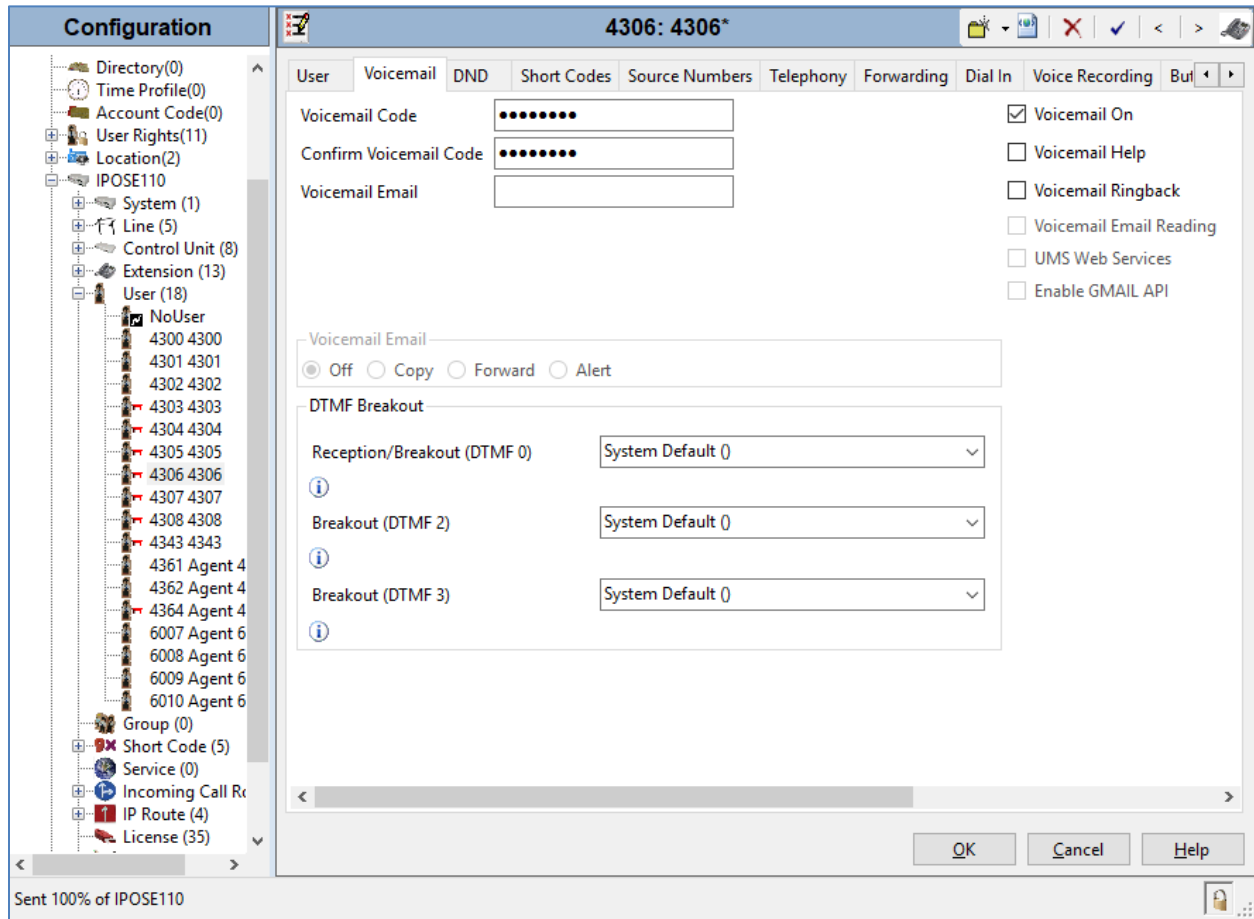
Repeat the same procedure to create another extension 4307 for second Cetus SIP phone.

5.7. Administer SIP User for Cetus SIP Endpoint

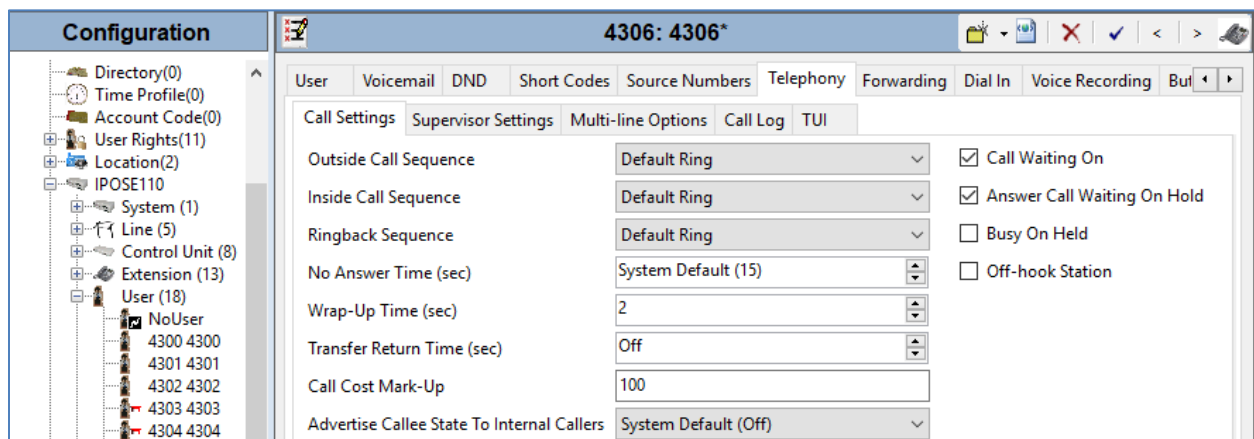
From the configuration tree in the left pane, do a right-click on **User** and select **New** from the pop-up window (now shown). Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension number as created above.



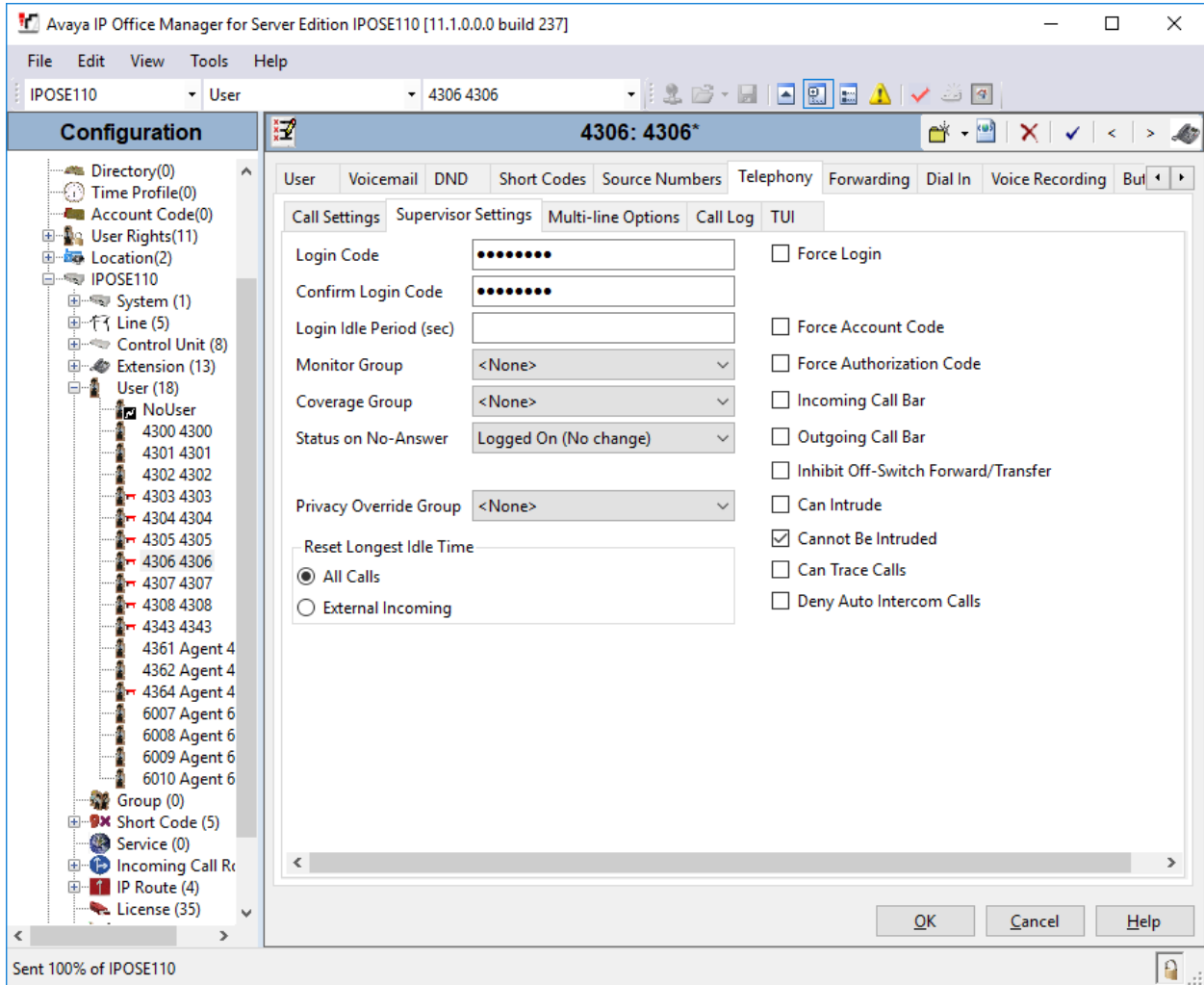
Select the **Voicemail** tab and select **Voicemail On** to enable voicemail and enter a passcode in the **Voicemail Code** and **Confirm Voicemail Code** field.



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

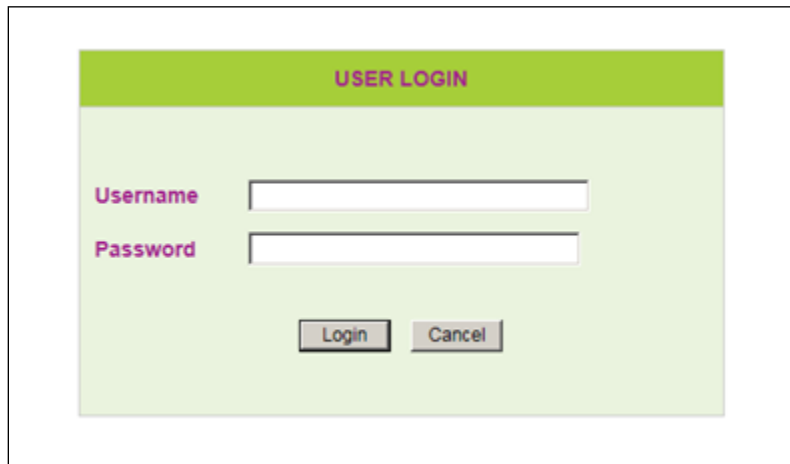


Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by Cetis SIP endpoint to register with IP Office Server Edition.



6. Configure 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 dynamic 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.



The image shows a web interface titled "USER LOGIN". It features two input fields: "Username" and "Password". Below the input fields are two buttons: "Login" and "Cancel". The interface has a light green background with a darker green header bar.

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: CC2, WAN IP: 172.16.199.6, Phone Number1: 4307, Phone Number2, and Firmware Version: CC2-3.0.0-053. 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' is highlighted. The main content area shows the 'Home' page with a 'Summary of Network Parameters' section. The 'WAN' status is 'Connected'. Network details include: Network Mode: DHCP, Current Gateway: 172.16.199.1, MAC Address: 00:19:F3:10:32:E1, Current IP Address: 172.16.199.6, and Current Netmask: 255.255.255.0. Below this is a 'Summary of VoIP Settings' section. The 'First Register' is 'Registered' with details: User Name: 4307, Register Server: 10.33.1.110, Register Server Port: 5060, SIP Backup Register Status: Not configured, SIP Backup Server, and SIP Backup Type: None. The 'Domain Realm' is 'ipoccc.com' and 'Outbound Proxy' is empty. The 'Second Register' is 'Not configured'. The 'Other' section shows '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: C32-3.0.0-050.bin is the firmware for Cetis Corded 2-line models including 3500IP and 9700IP.

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

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

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


The screenshot displays the Cetis WAN Settings interface. The left sidebar contains navigation menus for Home, Network Settings, VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area is titled 'WAN Settings' and shows the 'WAN Interface: Connected' status. The configuration is organized into several sections: Basic Settings (Network Mode: DHCP, Link Mode: AUTO, Primary DNS: 8.8.8.8, Secondary DNS: 8.8.4.4), Static IP Settings (Static IP Address: 192.168.1.100, Subnet Mask: 255.255.255.0, Default Gateway: 192.168.1.1), PPPoE Settings (User Account and Password fields), 802.1X Settings (802.1X: Disable, User Name and Password fields, Type: multicast), and LLDP Settings (LLDP: Enable, Packet Interval: 120). The interface includes 'Apply' and 'Cancel' buttons at the bottom.

Section	Parameter	Value
Basic Settings	Network Mode	<input checked="" type="radio"/> DHCP <input type="radio"/> Fixed <input type="radio"/> PPPoE
	Link Mode	AUTO
	Primary DNS	8.8.8.8
	Secondary DNS	8.8.4.4
Static IP Settings (Required if Network Mode is set to Static IP)	Static IP Address	192.168.1.100
	Subnet Mask	255.255.255.0
	Default Gateway	192.168.1.1
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

6.2. VoIP Settings

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

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

Cetis 

SYSTEM SUMMARY
Model: CC2
WAN IP: 172.16.199.6
Phone Number1: 4307
Phone Number2:
Firmware Version: CC2-3.0.0-053

Home • VoIP Settings • Primary Register

Primary Register

First Server: Registered *Backup Server: Not configured*

First Register Server

Use Service	Enable ▾
Display Name	<input type="text"/>
User Name	4307
Authorization User Name	4307
Password	*****
Register Server Port	5060
Register Server Address	10.33.1.110
Domain Realm	ipocc.com
Outbound proxy	10.33.1.110
Register Expire	300
SIP Backup Type	None ▾
SIP Backup Server	<input type="text"/>
MWI Subscribe	Enable ▾
Subscribe Expire	300

Second Server: Not configured *Backup Server: Not configured*

Second Register Server

Use Service	Disable ▾
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Protocol Control

Local SIP Port	5060
Local RTP Port	20000
Keep Alive Packet	<input type="radio"/> Off <input checked="" type="radio"/> On
Keep Alives Period	60

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

- **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: CC2
WAN IP: 172.16.199.6
Phone Number1: 4307
Phone Number2:
Firmware Version: CC2-3.0.0-053

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

Subscribe Expire: 300

Second Server: Not configured *Backup Server: Not configured*

Second Register Server

Use Service: Disable

Protocol Control

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

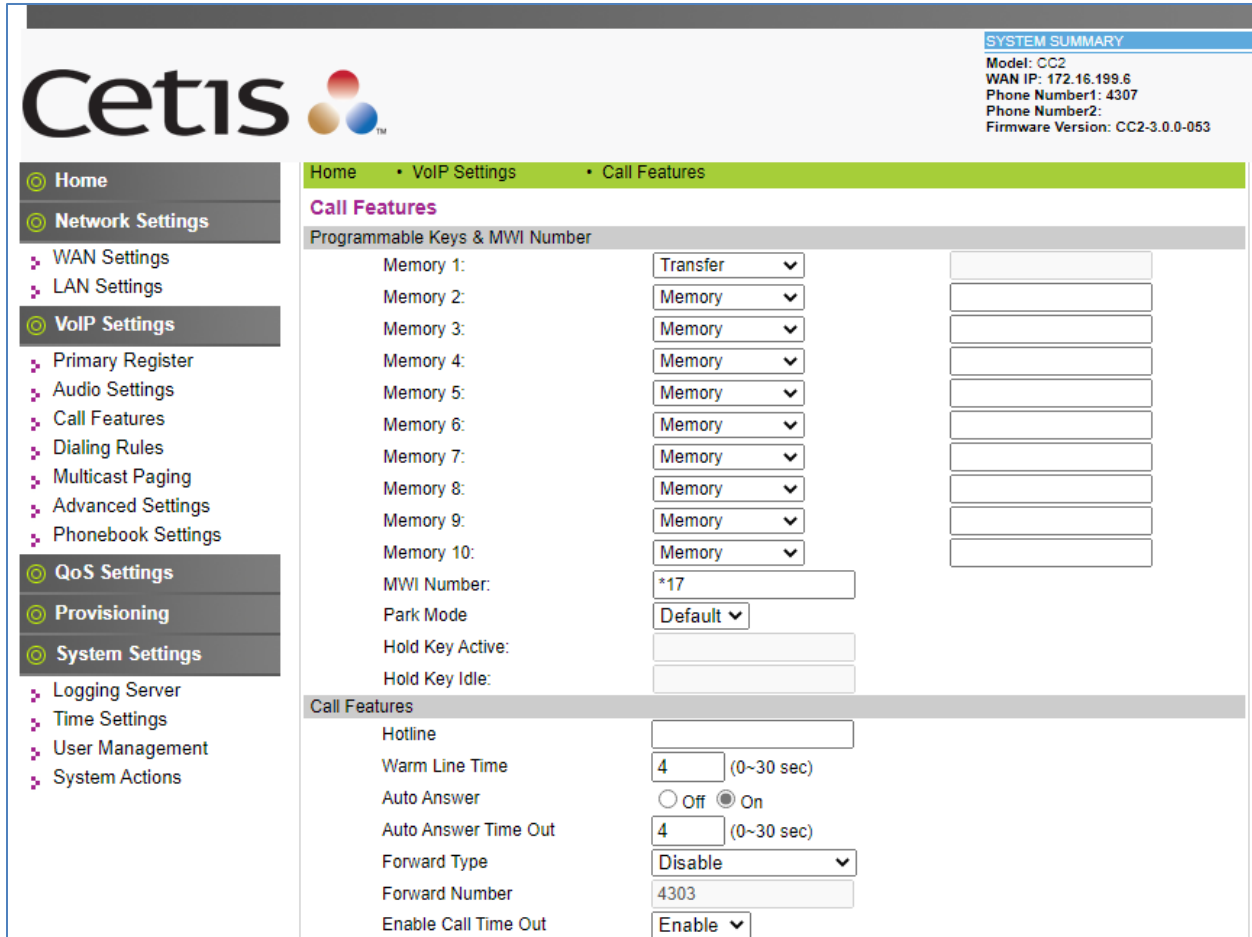
Select **Audio Settings** under the **VoIP Settings** section. In this page, a customer can prioritize codec settings. The picture below shows the list of codecs used for the compliance test.

The screenshot displays the Cetis web interface for VoIP settings. The left sidebar contains a navigation menu with categories like Home, Network Settings, VoIP Settings, QoS Settings, Provisioning, and System Settings. The main content area is titled 'Audio Settings' and is divided into several sections:

- Sound and Volume Control:** Includes settings for Handset (5), Speaker (7), Ringer Tone (1), Signal Standard (United States), Ringer (On), and Ringer Type (ringer 1).
- Codecs Settings:** Lists Codec Priority 1 through 6 with dropdown menus (G.711u, G.729, G.711a, G.723.1, iLBC, G.722) and Packet Data Size (20 ms). It also includes checkboxes for iLBC 15.2K and G.723.1 5.3K.
- Voice VAD/CNG:** Includes checkboxes for Voice VAD and CNG, both currently set to Off.
- Codec ID Settings:** Includes DTMF Payload(RFC2833) set to 101.

Buttons for 'Apply' and 'Cancel' are located at the bottom of the settings area. A 'SYSTEM SUMMARY' box in the top right corner provides device information: Model: CC2, WAN IP: 172.16.199.6, Phone Number1: 4307, Phone Number2, and Firmware Version: CC2-3.0.0-053.

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



Cetis

SYSTEM SUMMARY
 Model: CC2
 WAN IP: 172.16.199.6
 Phone Number1: 4307
 Phone Number2:
 Firmware Version: CC2-3.0.0-053

Home • VoIP Settings • Call Features

Call Features

Programmable Keys & MWI Number

Memory 1:	Transfer	
Memory 2:	Memory	
Memory 3:	Memory	
Memory 4:	Memory	
Memory 5:	Memory	
Memory 6:	Memory	
Memory 7:	Memory	
Memory 8:	Memory	
Memory 9:	Memory	
Memory 10:	Memory	
MWI Number:	*17	
Park Mode	Default	
Hold Key Active:		
Hold Key Idle:		

Call Features

Hotline	
Warm Line Time	4 (0~30 sec)
Auto Answer	<input type="radio"/> Off <input checked="" type="radio"/> On
Auto Answer Time Out	4 (0~30 sec)
Forward Type	Disable
Forward Number	4303
Enable Call Time Out	Enable

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

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

The screenshot displays the Cetis web interface. On the left is a navigation menu with categories: Home, Network Settings (WAN, LAN), VoIP Settings (Primary Register, Audio, Call Features, Dialing Rules, Multicast Paging, Advanced Settings, Phonebook), QoS Settings, Provisioning, and System Settings (Logging Server, Time Settings, User Management, System Actions). The main content area is titled 'Call Features' and includes sections for 'Call Features', 'Display Settings', and 'Blocked List Set'. The 'Call Features' section contains various settings with input fields and radio buttons. The 'Display Settings' section has 'LCD Contrast' and 'Greeting Message'. The 'Blocked List Set' is a table with columns for Position, Number, and Select.

SYSTEM SUMMARY
 Model: CC2
 WAN IP: 172.16.199.6
 Phone Number1: 4307
 Phone Number2:
 Firmware Version: CC2-3.0.0-053

Call Features

Hotline:

Warm Line Time: (0~30 sec)

Auto Answer: Off On

Auto Answer Time Out: (0~30 sec)

Forward Type:

Forward Number:

Enable Call Time Out:

No Answer Time Out:

Call Waiting: Off On

Do Not Disturb: Off On

Ban Outgoing: Off On

Accept Any Call: Off On

Display Settings

LCD Contrast: (1~8)

Greeting Message:

Blocked List Set

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

7. Verification Steps

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

7.1. Verify Cetus SIP Telephones

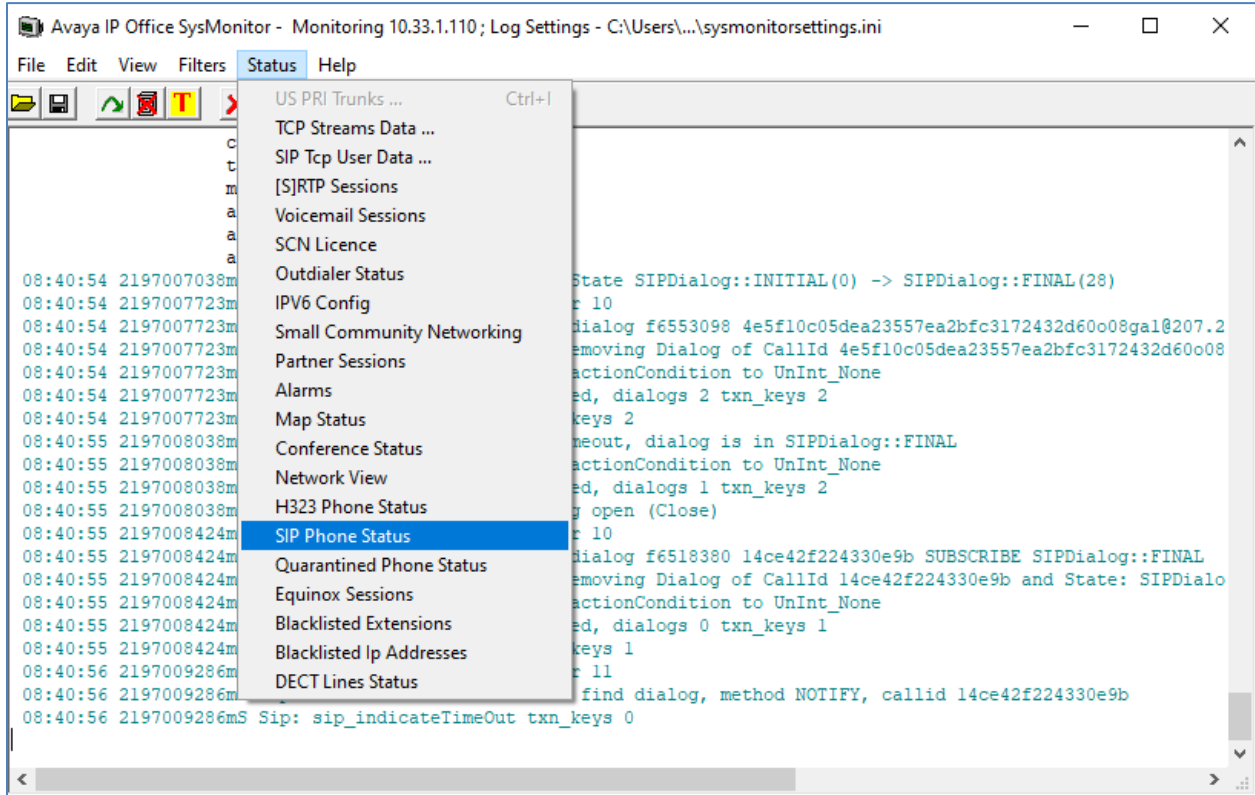
Select **VOIP Settings** in the left pane to display the **VoIP Summary** page. Verify that the **Primary Register** is set to *Registered*.

The screenshot shows the Cetus web interface. The top left features the Cetus logo. The top right corner has a 'SYSTEM SUMMARY' box with the following details: Model: CC2, WAN IP: 172.16.199.6, Phone Number1: 4307, Phone Number2: (blank), and Firmware Version: CC2-3.0.0-053. A left-hand navigation menu includes Home, Network Settings (with sub-items WAN Settings and LAN Settings), VoIP Settings (selected), QoS Settings, Provisioning, and System Settings (with sub-items Logging Server, Time Settings, User Management, and System Actions). The main content area is titled 'VoIP Summary' and shows the following configuration:

Home • VoIP Settings	
VoIP Summary	
First Register: Registered	
User Name: 4307	Domain Realm: ipocc.com
Register Server: 10.33.1.110	Outbound Proxy:
Register Server Port: 5060	
SIP Backup Register Status: Not configured	
SIP Backup Server:	
SIP Backup Type: None	
Second Register: Not configured	
User Name:	Domain Realm:
Register Server:	Outbound Proxy:
Register Server Port: 5060	
SIP Backup Register Status: Not configured	
SIP Backup Server:	
SIP Backup Type: None	
Other	
NAT Traversal(STUN): Disabled	STUN Sever Address:

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 C32.3.0.0.53 in the list.

The screenshot shows the SIPPhoneStatus application window. It displays a table of registered SIP users. The table has columns for Extn Num, User Num, Phone Type, Security, Behi..., IP Address, Privat..., Transport, User Agent, Licensed, SIP..., SIP..., SIP Subs..., and Status. The table shows 6 registered users, including one for Cetus C32.3.0.0.53. The Display Options section at the bottom shows the "Registered" radio button selected.

Extn Num	User Num	Phone Type	Security	Behi...	IP Address	Privat...	Transport	User Agent	Licensed	SIP...	SIP...	SIP Subs...	Status
4303	4303	1140E_SIP	best effort		192.168.193.3		UDP	Avaya IP Phone 1140E (SIP1140e.04...	Avaya IP	RU	TH	message...	SIP: Registered
4305	4305	J129 SIP	best effort		192.168.193.4		TLS	Avaya J129 IP Phone 4.0.6.0.7 a478...	Avaya IP	RU		message...	SIP: Registered
4306	4306	SIP	disable		172.16.193.7		UDP	Cetus C32.3.0.0.53	3rd Party IP	RU		message...	SIP: Registered
4307	4307	SIP	disable		172.16.193.6		UDP	Cetus C32.3.0.0.053	3rd Party IP	RU		message...	SIP: Registered
4343	4343	AWAYA_ACCS	best effort		10.33.1.57		TLS	Avaya Nebraska Contact Center 7.0...	Avaya IP	R	0	message...	SIP: Registered
\$1.4304	4304	EQNX_DESKTOP	best effort		192.168.193.2...		TLS	Avaya Communicator/3.0 (3.7.4.22.1...	Avaya Softph...	RU		avaya-ccs...	SIP: Registered

8. Conclusion

These Application Notes have described the administration steps required to integrate the Cetus 3500IP Series and 9700IP Series SIP telephones SIP with Avaya IP Office Server Edition. The Cetus 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] *Deploying IP Office Server Edition*, Release 11.1, Issue 14, April 2020.
- [2] *IP Office Platform 11.0, Deploying Avaya IP Office Servers as Virtual Machines*, 15-601011 Issue 07d, June 9, 2020.
- [3] *IP Office Platform 11.0, Deploying Avaya IP Office Essential Edition (IP500 V2)*, 15-601042, Issue 35f, January 2020.
- [4] *Administering Avaya IP Office Platform with Manager*, Release 11.1 Issue 2, May 2020.
- [5] *Administering Avaya IP Office™ Platform with Web Manager*, Release 11.1 Issue 2, May 2020.
- [6] *Planning for and Administering Avaya IX™ Workplace Client for Android, iOS, Mac and Windows*, Issue 1, Release 3.9, June 2020.
- [7] *Using Avaya IX™ Workplace Client for IP Office*, Release 11.1 Issue 9, June 2020.

Additional Avaya IP Office documentation can be found at:

<http://marketingtools.avaya.com/knowledgebase/>

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