Avaya Solution & Interoperability Test Lab

Application Notes for Cetis 3300IP and 9600IP Series SIP Telephones with Avaya IP Office - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Cetis 3300IP and 9600IP Series SIP Telephones with Avaya IP Office. The Cetis 3300IP and 9600IP Series SIP Telephones were designed for the hospitality industry and register with Avaya IP Office. The Cetis 3302IP and 9602IP SIP Telephones, which use the same firmware, were used in this compliance test.

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.

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1. Introduction

These Application Notes describe the steps required to integrate the Cetis 3300IP and 9600IP Series SIP Telephones with Avaya IP Office. The Cetis 3300IP and 9600IP Series SIP Telephones were designed for the hospitality industry. In the compliance test, Cetis SIP telephones registered with Avaya IP Office, established calls with other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya IP Office Shortcodes. The Cetis 3302IP and 9602IP SIP Telephones, which use the same firmware, were used in this compliance test. The 3302IP is a corded SIP telephone with a display and the 9602IP is DECT SIP telephone without a display.

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**.

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.

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Cetis 3300IP and 9600IP Series SIP Telephones and Avaya SIP and H.323 telephone and exercising basic telephony features, such as hold, mute, and conference. In addition, hospitality features, such as wake-up calls and Do Not Disturb were covered. Additional telephony features, such as call forward, follow me, call park/unpark, and call pickup were also verified using Avaya IP Office Shortcodes.

The serviceability testing focused on verifying that the Cetis 3300IP and 9600IP Series SIP Telephones come back into service after re-connecting the Ethernet connect or rebooting the phone.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Cetis 3300IP and 9600IP Series SIP Telephones with Avaya IP Office.
- Calls between Cetis telephones and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Cetis telephones and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.

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- Basic telephony features, including hold, mute, redial, multiple calls, and 3-way conference.
- Extended telephony features using Avaya IP Office Shortcodes for wakeup calls, Do Not Disturb, Call Forward, Follow Me, Call park/Unpark, and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve messages.
- Use of programmable buttons on the Cetis telephones.
- Proper system recovery after a restart of the Cetis telephones and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- If the Cetis phone dials an invalid number, an audible beep (chirp) is played to the user and the call disconnects.
- The Cetis phones support 3-way, attended conferences only, if the Cetis phone drops from a conference, the other parties in the conference also drop.
- If the **End with '#'** option is enabled in the **Digital Map** of the Cetis phone (see **Section 6**), Avaya IP Office shortcodes should not end with a # since the Cetis phone would not send the #. The # should be substituted with another character, such as a *. However, if the "End with '#' option is disabled, the shortcodes may end with a #.

2.3. Support

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

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

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3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Cetis 3300IP and 9600IP Series IP Telephones with Avaya IP Office. The Cetis telephones registered with Avaya IP Office via SIP. Avaya Voicemail Pro was used to support hospitality wakeup calls.

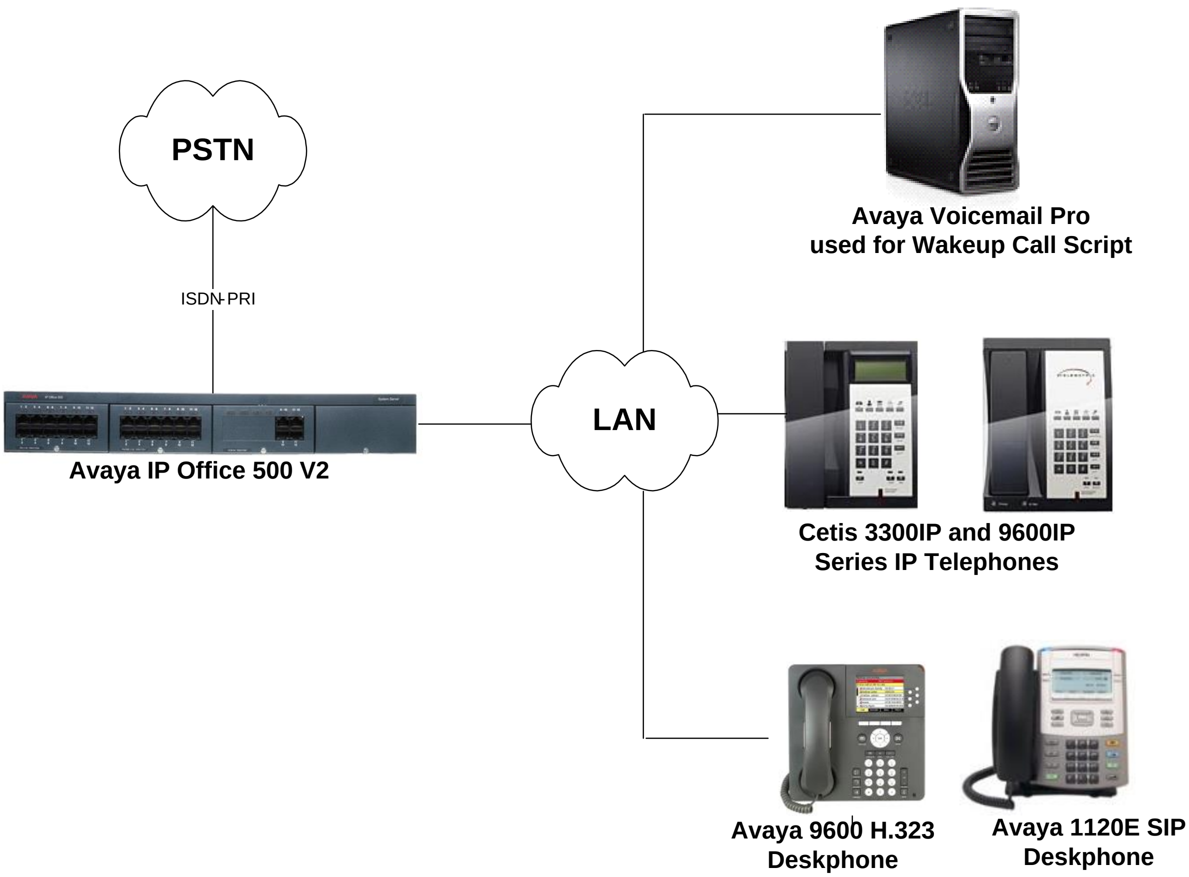

Figure 1: Cetis 3300IP and 9600IP Series IP Telephones with Avaya IP Office

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4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 500 V2	9.0.2.0 (Build 860)
Avaya Voicemail Pro	9.0 (Build 311)
Avaya 9600 Series IP Deskphone	6.3116 (H.323)
Avaya 1120E IP Deskphone	04.03.18 (SIP)
Cetis 3300IP and 9600IP Series SIP Telephones	1.8.6-1249

Notes:
Testing was performed with IP Office 500 v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks. IP Office Server Edition does not support TAPI Wave or Group Voicemail.

Common Firmware on Cetis Group IP Phones
Cetis SIP firmware follows a naming convention based on model. All Cetis IP phones share the same base firmware. Server registrations, SIP messaging, and call control are all the same. The different versions are only to accommodate variances in the number of speed dial keys and the different LCD screen sizes.

- 3300IP / 3302IP / ND2110S / ND2210S - SC2 FW = SIP Corded
- 9600 / NDC2110S / E103P - SD1 FW = SIP DECT 1 Line
- 9602 / NDC2210S / E203P - SD2 FW = SIP DECT 2 Line
- 3300TRM - ST1 FW = SIP Trimline 1 Line
- 3302TRM - ST2 FW = SIP Trimline 2 Line
- E100P / E200P - SE2 FW = E-Series SIP Corded

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5. Configure Avaya IP Office

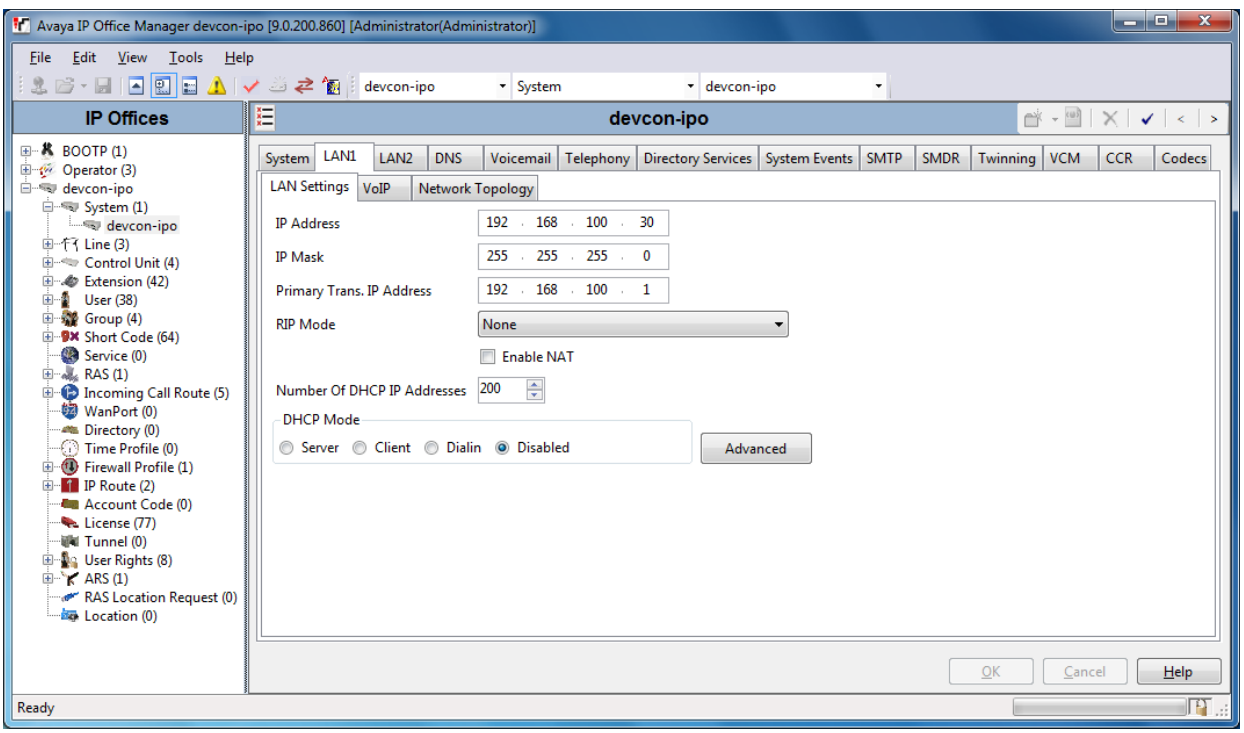
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Obtain LAN IP address
- Administer SIP registrar
- Administer SIP extension for e-IVR
- Administer SIP user for e-IVR

Note: The configuration of the Cetis 3300IP and 9600IP Series SIP Telephones are identical on Avaya IP Office. Call routing to the PSTN is outside the scope of these Application Notes.

5.1. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **System** screen for the IP Office 500 V2 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 telephones.



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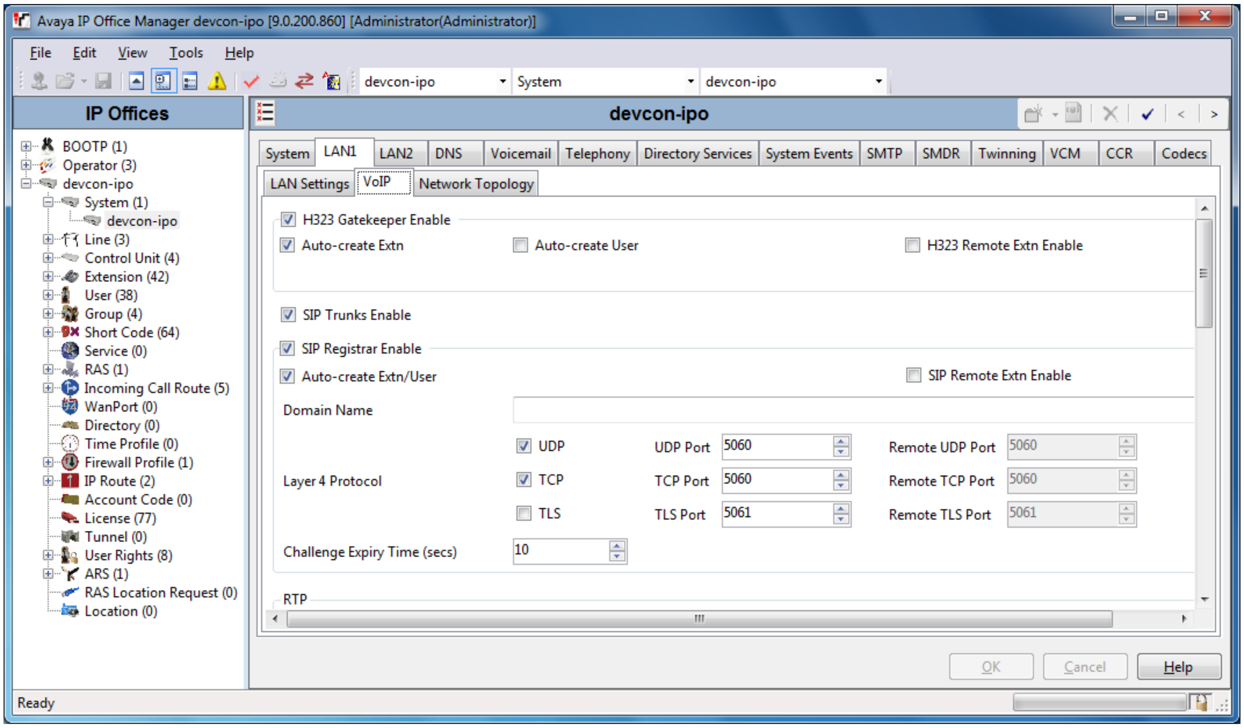
5.2. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** is checked and enter a valid **Domain Name**. In the compliance testing, the **Domain Name** field was left blank so the LAN IP address was used.



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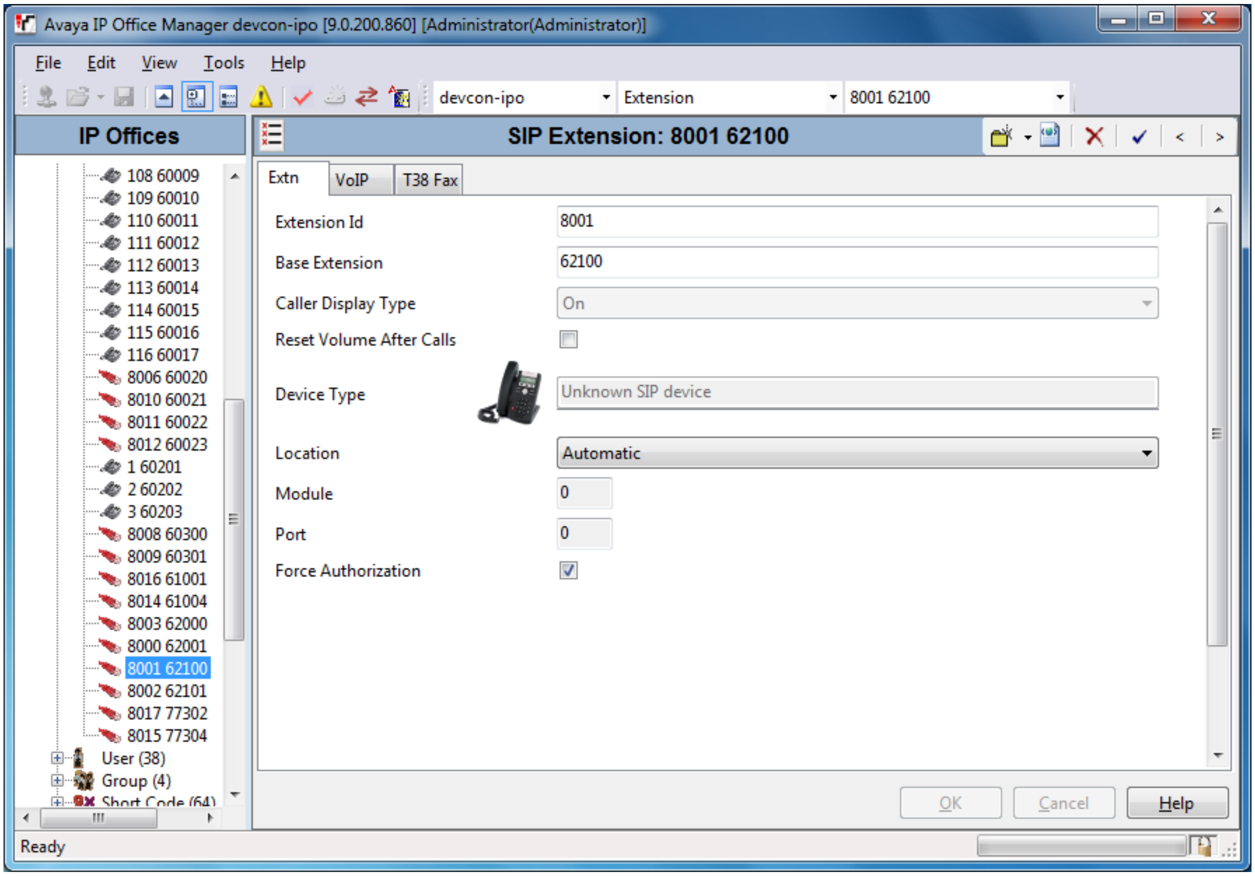
5.3. Administer SIP Extension for Cetis 3302IP and 9602IP SIP Telephones

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. Enter the desired extension for the **Base Extension** field as shown below. In this example, the Cetis 3302IP was assigned extension *62100*. This is the extension that the Cetis SIP telephone will use to register with IP Office.



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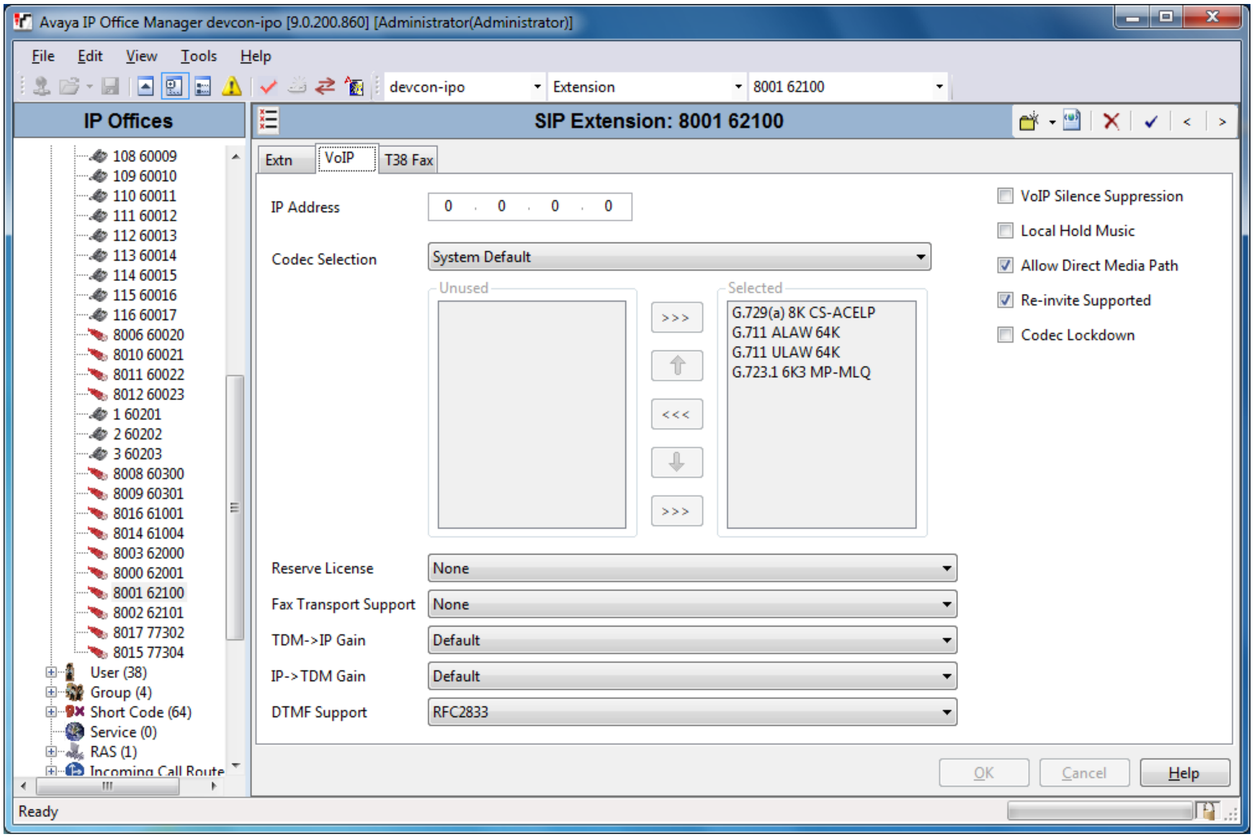
Select the **VoIP** tab and retain the default values in the all fields. During the compliance test, Cetis SIP telephones were tested using G.711 and G.729 codecs. Enable **Allow Direct Media Path** so that audio/RTP flows directly between two SIP endpoints without using media resources in Avaya IP Office.



Repeat these steps for each extension required. During the compliance test, extensions 62100 and 62101 were used for the 3302IP and 9600IP, respectively.

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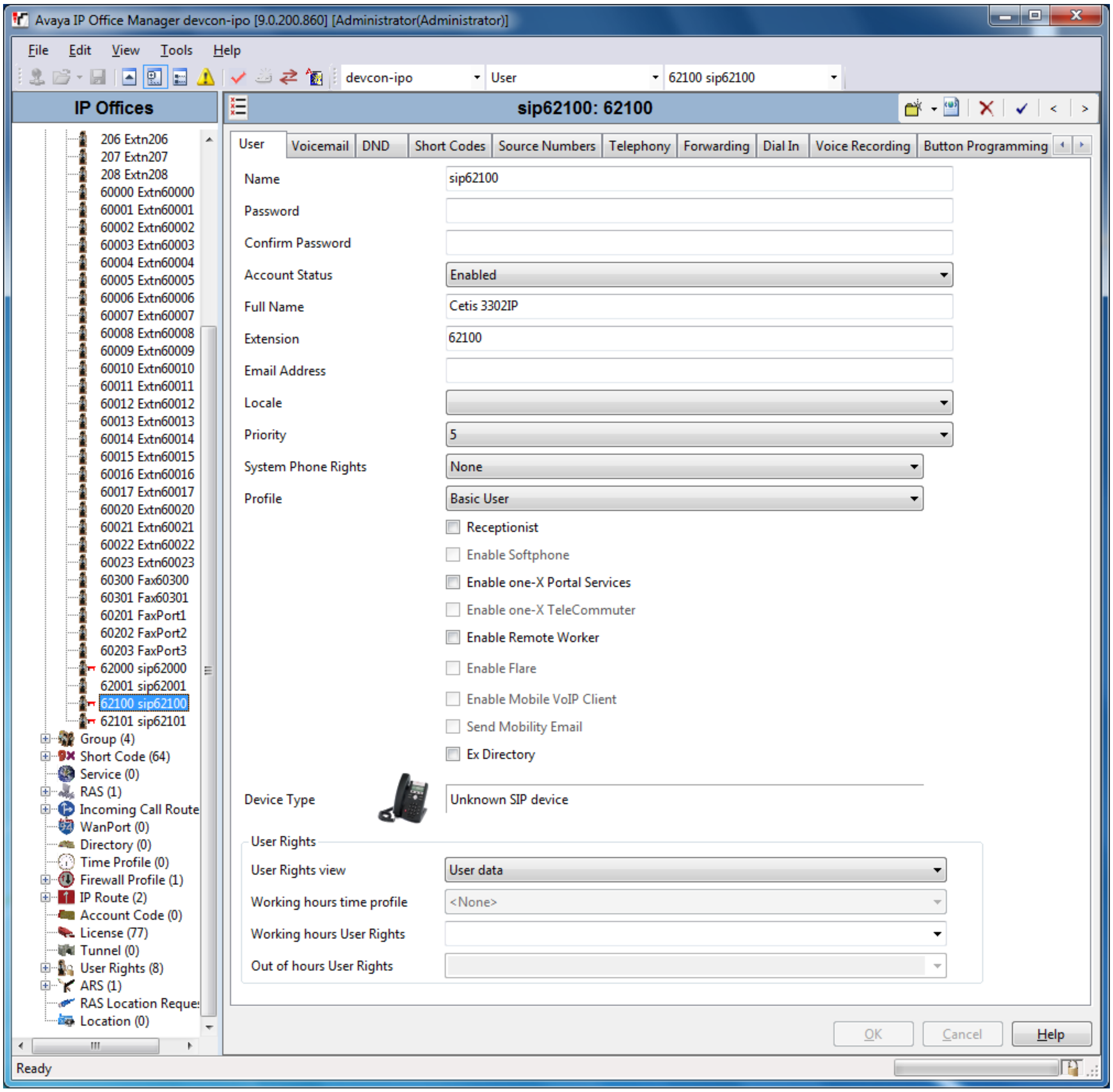
5.4. Administer SIP User for Cetis 3300IP and 9600IP Series SIP Telephones

From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list. Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension created above.



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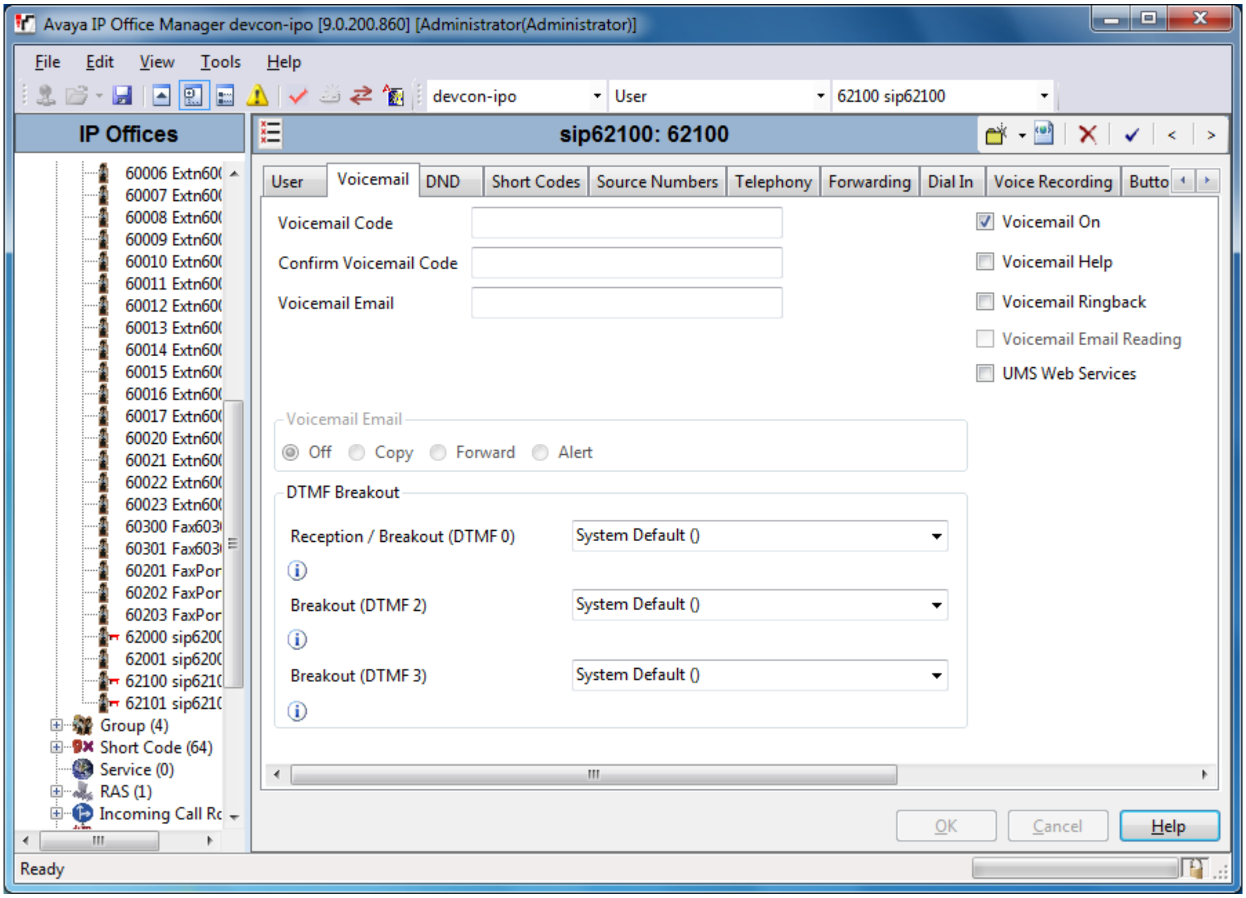
Select the **Voicemail** tab and select **Voicemail On** to enable voicemail for the Cetis phone.



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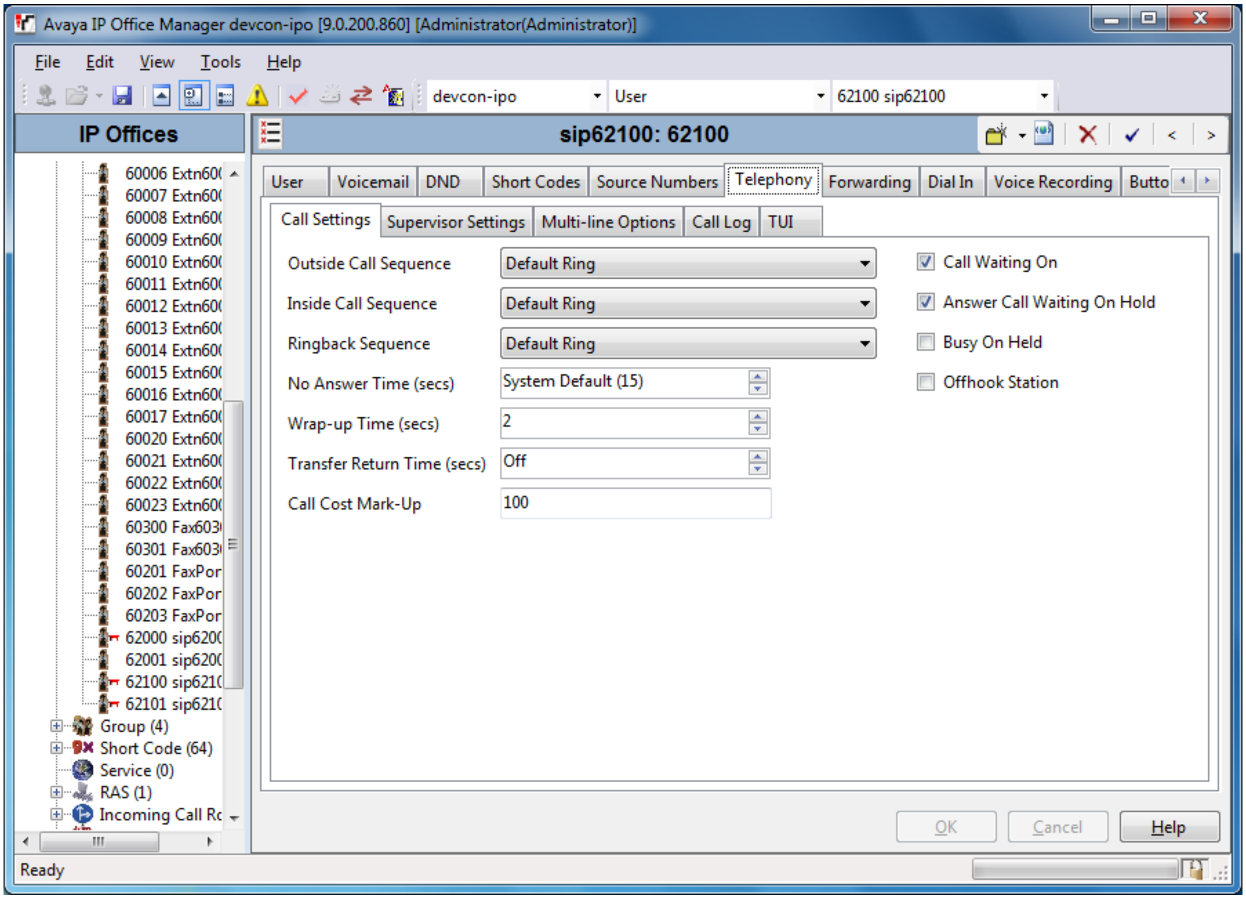
Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note the settings below for the user.

Note: Call Waiting is required to allow a secondary incoming call to the Cetis phone; otherwise, the call second incoming call will be denied.



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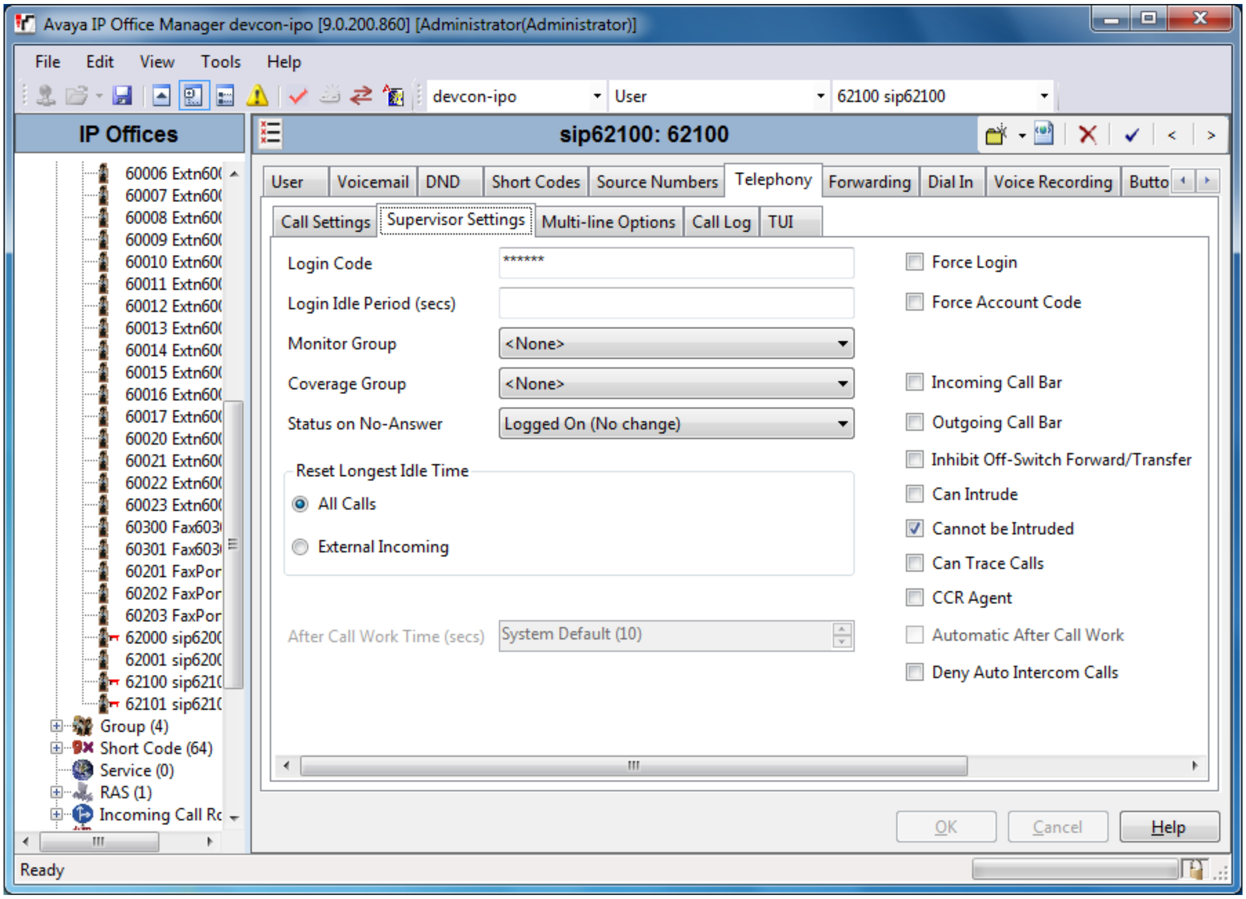
Select the **Supervisor Settings** tab and enter a desired **Login Code**. The Login Code is the password that will be used by the Cetis SIP phone to register with IP Office.



Repeat these steps for each user required. During the compliance test, users 62100 and 62101 were used for the 3302IP and 9602IP, respectively.

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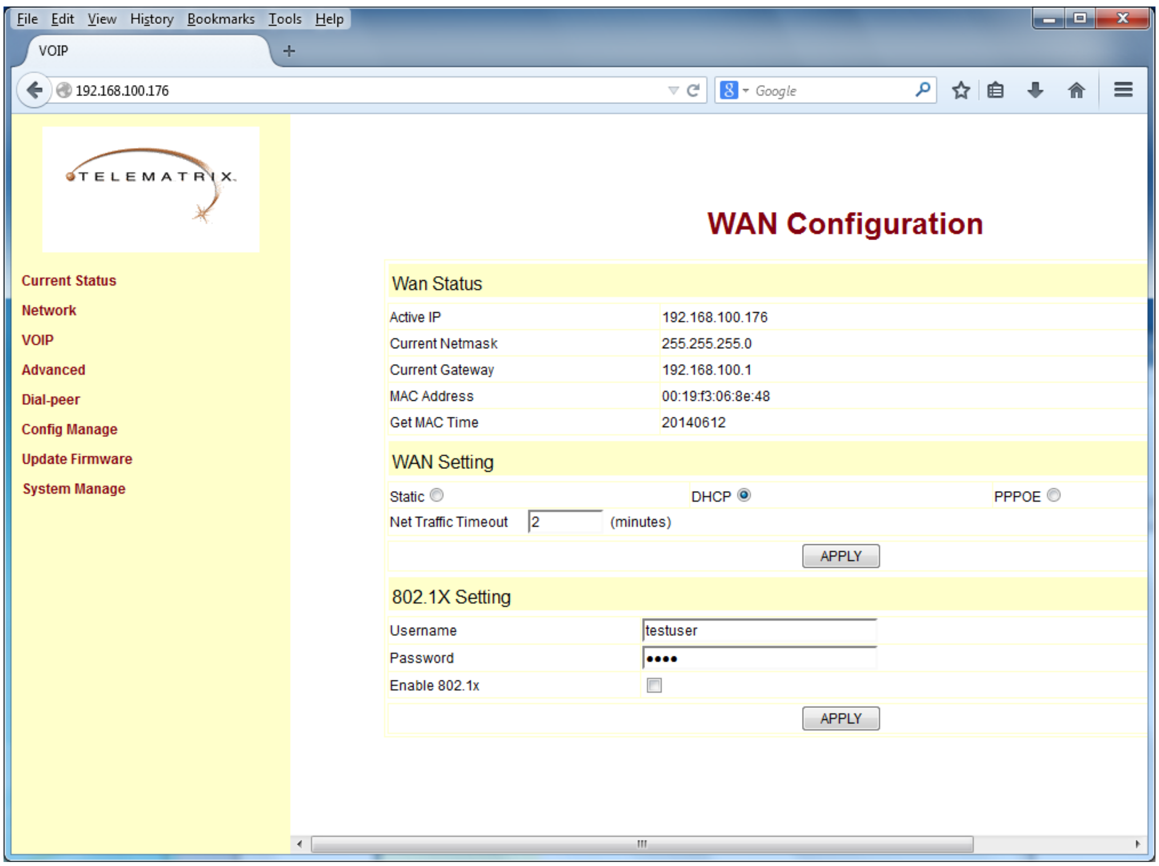
6. Configure Cetis 3302IP and 9602IP SIP Telephones

Access the Cetis 3302IP and 9602IP web interface using the URL "http://ip-address" in an Internet browser window, where "ip-address" is the IP address of the Cetis telephone. By default, DHCP is enabled on the Cetis telephones. For this compliance test, an IP address was assigned to the Cetis telephones using a DHCP server. To determine the IP address assigned to the Cetis telephone, enter **47# on the telephone to hear the IP address. Log in using the appropriate credentials. To view the network configuration, navigate to **Network → WAN Config** on the 3302IP and **Network → LAN Config** on the 9602IP. Alternatively, selecting **Current Status** in the left pane will also display the network configuration.



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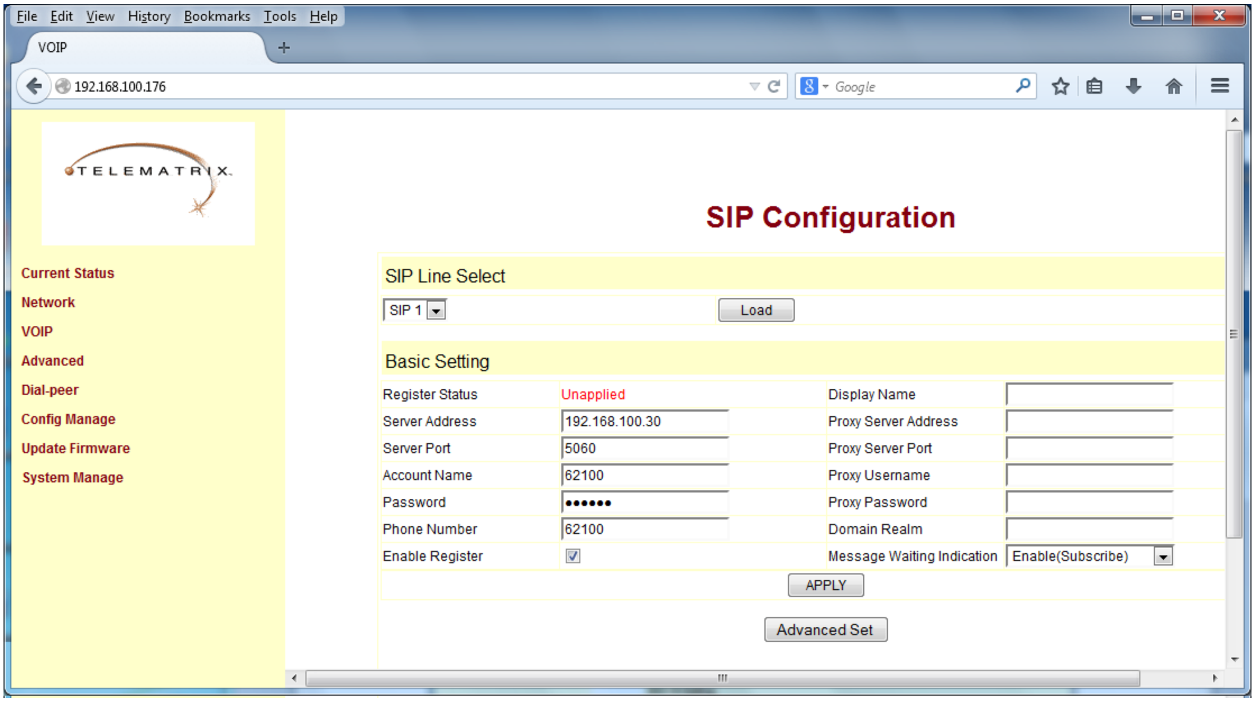
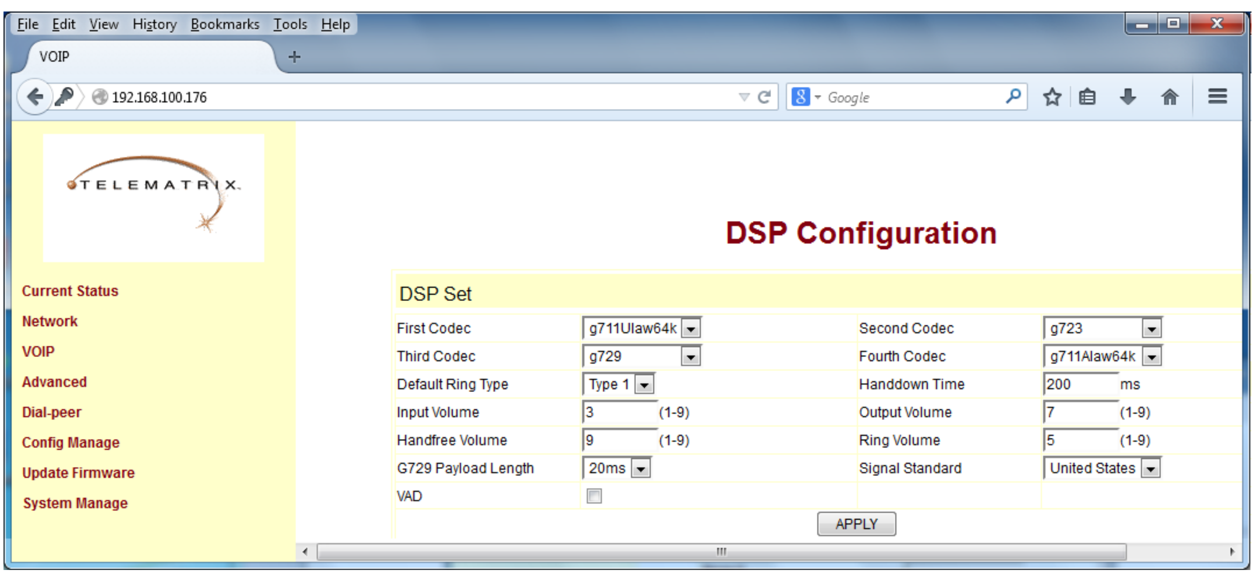
Select **VoIP** in the left pane to display the **SIP Configuration** screen as shown below. Set the **Server Address** to the Avaya IP Office LAN1 IP address noted in **Section 5.1** and specify 5060 as the **Server Port**. For the **Account Name** and **Phone Number** fields, specify the SIP extension (e.g., *62100*) assigned to the Cetis phone, and set the **Password** field to the password configured in **Section 5.4**. Select the **Enable Register** checkbox and enable the **Message Waiting Indication** option for voicemail notification. Click **Apply** button.



Navigate to **Advanced → DSP** to view the codec set configured on the Cetis telephone. This is displayed for informational purposes only and no change is required.



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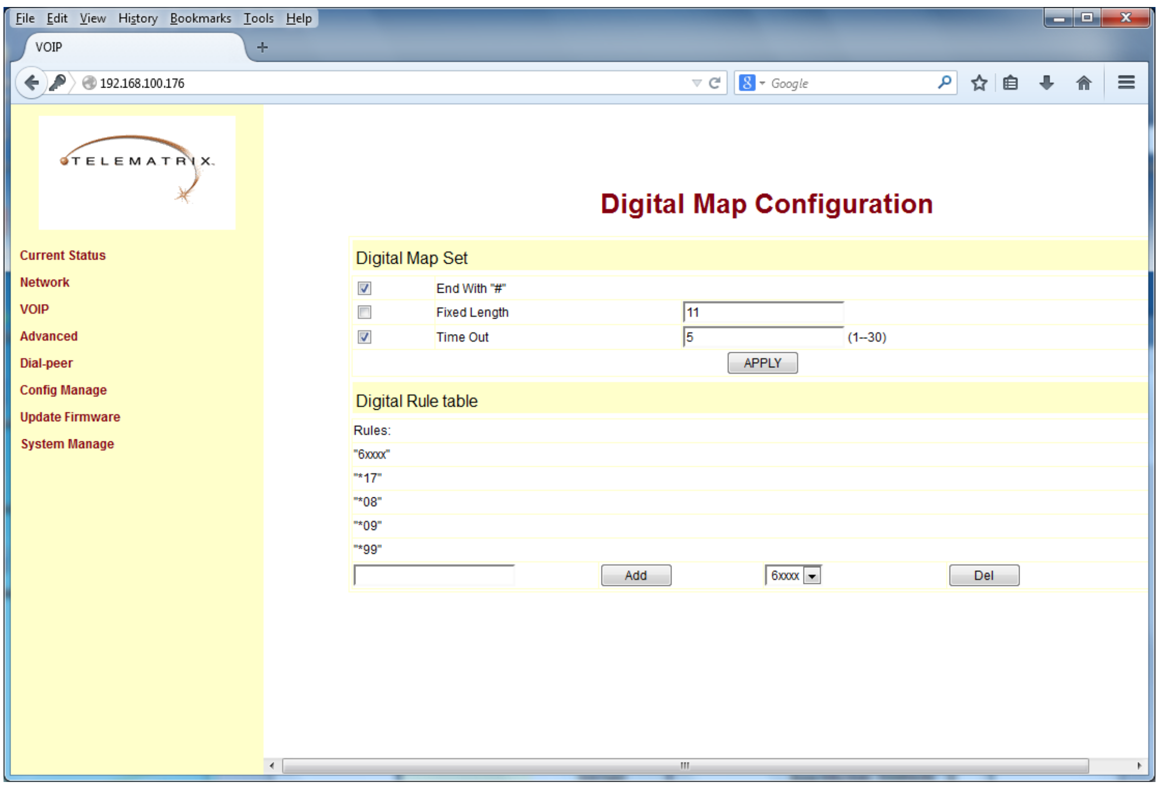
Navigate to Advanced → Digital Map to configure the dial plan supported by Avaya IP Office. In this configuration, local extensions were 5-digits in length beginning with '6'. Therefore, an entry for *6xxxx* was added to the **Digital Map**. By doing so, the Cetis telephone would dial as soon as the dial pattern is recognized instead of waiting for the inter-digit timeout interval to expire or dialing a '#' at the end of the digits to indicate end of dialing. The timer is configured on this page in the **Time Out** field and it is configured at 5 secs. The End With '#' field checkbox is selected to allow the '#' to signal end of digits. In addition, some shortcodes were entered in the **Digit Map** so that they would be recognized as soon as a user dialed them. Click **Apply**.

Note: If the **End With '#'** checkbox is selected, Avaya IP Office shortcodes cannot end with a '#' otherwise the Cetis telephone will never send the '#' and the feature will not be activated or deactivated. If this option is enabled, substitute the '#' at the end of a short code with another symbol, such as '*'. Alternatively, if a shortcode ends with a '#', disable this option. Note that if this option is disabled, the inter-digit timer will have to expire if a user dials a number that is not recognized in the **Digit Map**.



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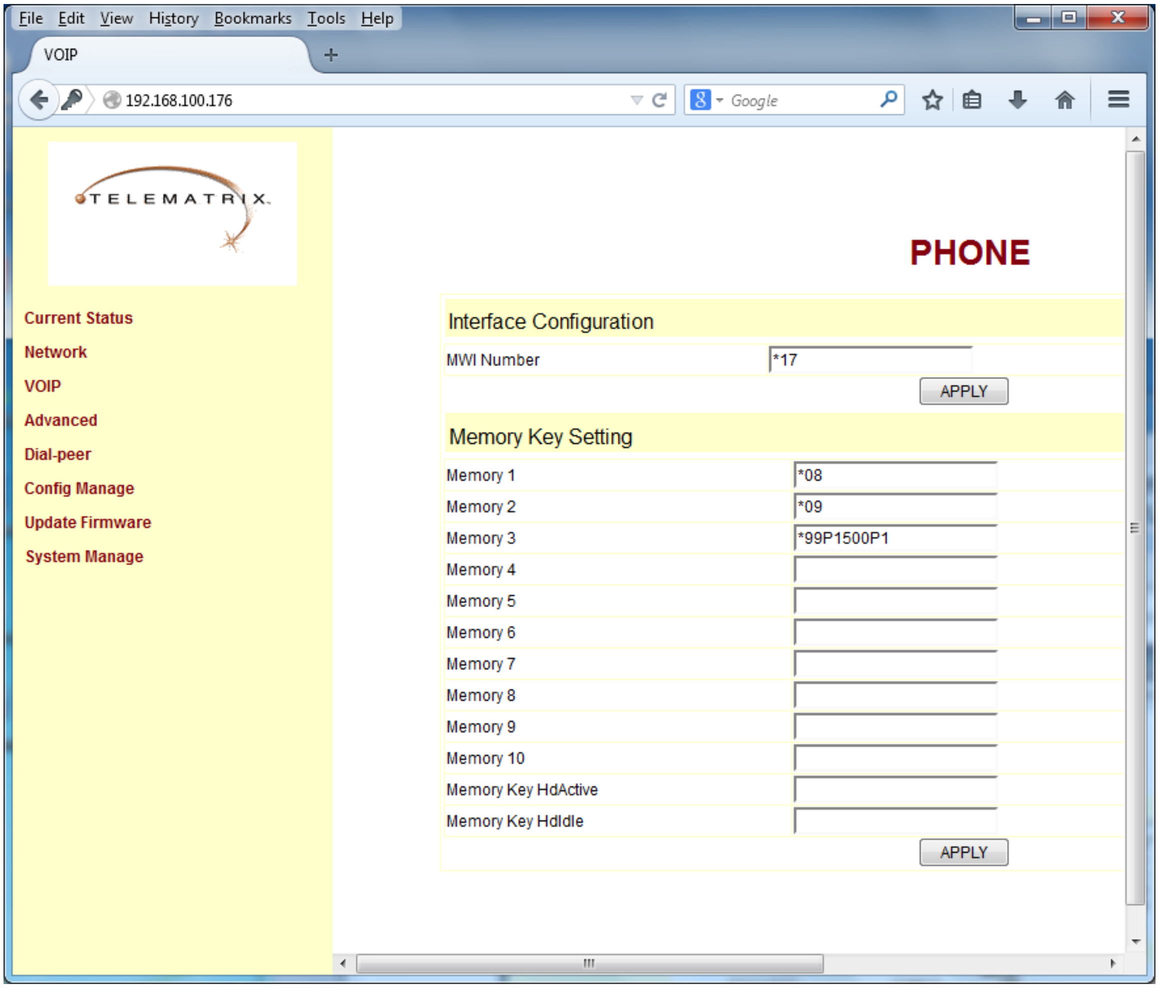
To program the voicemail button and other buttons on the Cetis telephone, navigate to **Advanced → Phone**. In the **MWI Number** field, configure the shortcode associated with the Voicemail Collect feature. Under the **Memory Key Setting** section, configure the other buttons as desired. The first three buttons were also configured with other shortcodes as shown below. Click **Apply**.



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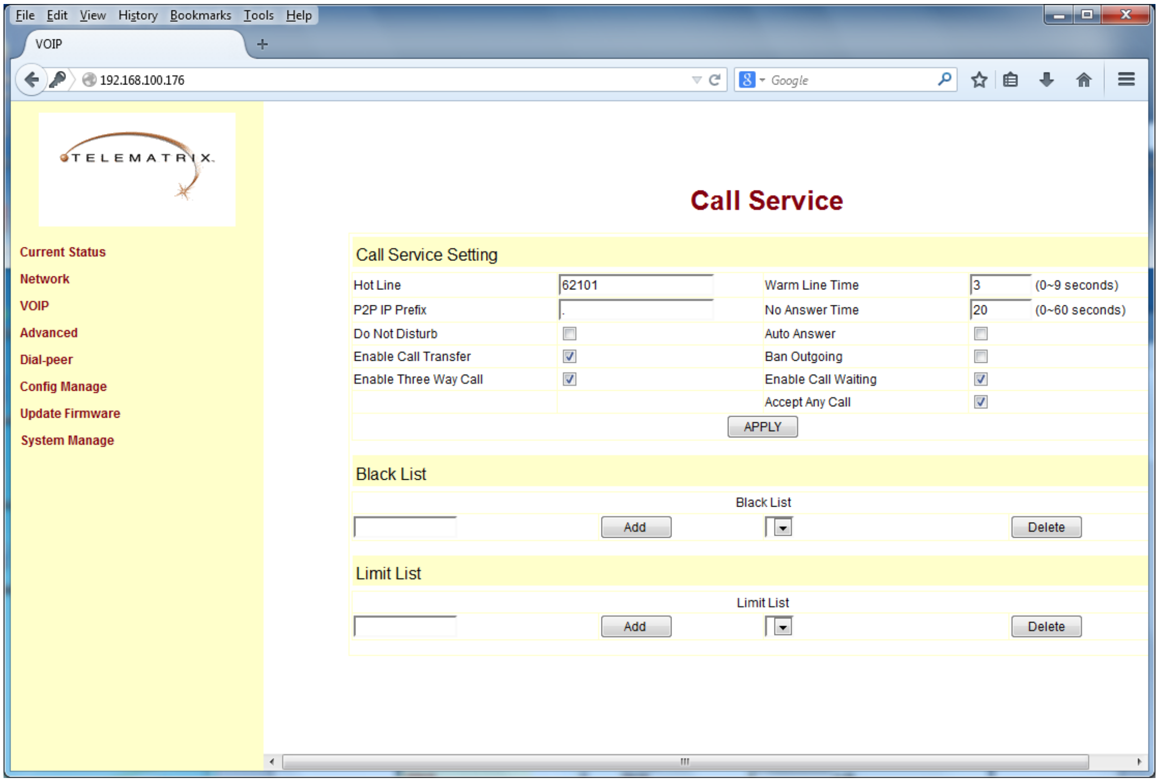
To enable the **Hot Line** feature, navigate to **Advanced → Call Service** and enter an extension in the **Hot Line** field. When **Hot Line** is configured, Cetis will dial this number as soon as a user goes off hook. The **Warm Line Time** field may be configured to allow Cetis to wait a pre-determined amount of time (e.g., 3 secs) before dialing the **Hot Line** number. This would give the user time to dial a number. Click **Apply**.



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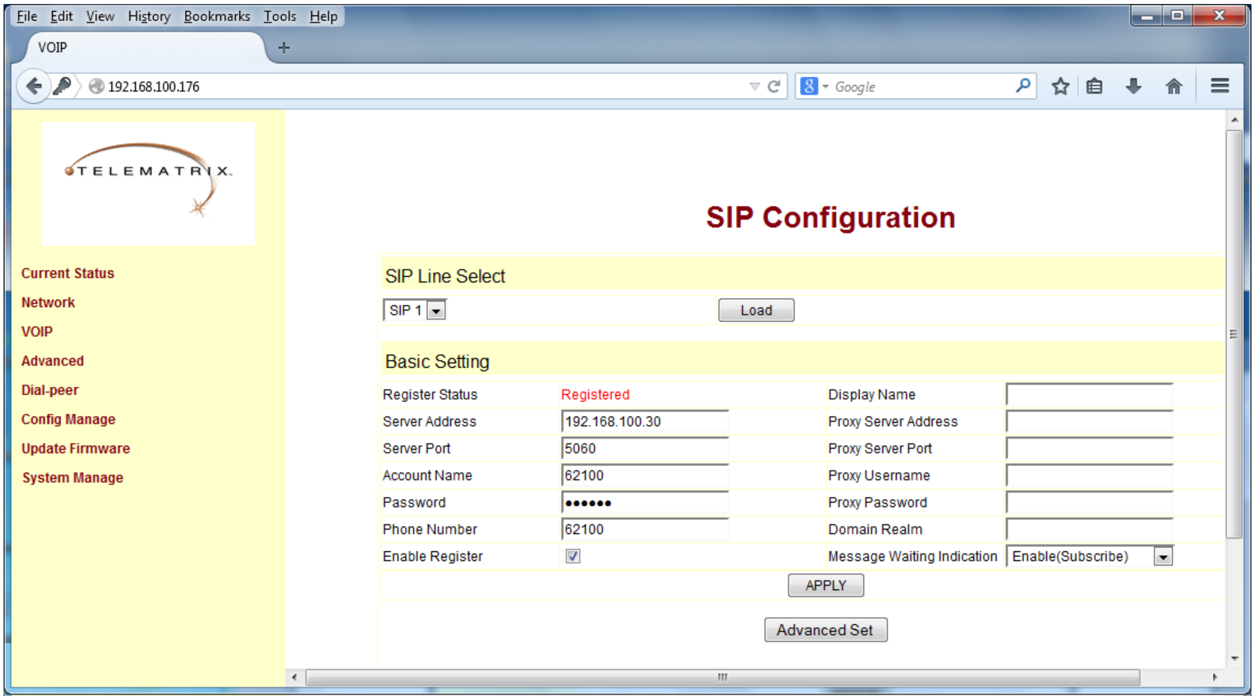
7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya IP Office and the Cetis 3300IP and 9600IP SIP Telephones.

1. Select **VOIP** in the left pane to display the SIP Configuration. Verify that the **Register Status** is set to *Registered*.



2. Verify basic telephony features by establishing calls between a Cetis telephone and another phone.

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8. Conclusion

These Application Notes have described the administration steps required to integrate the Cetis 3300IP and 9600IP SIP Telephones with Avaya IP Office. The Cetis SIP telephones registered successfully with Avaya IP Office via SIP. Incoming and outgoing calls were placed to/from the Cetis SIP telephones and basic telephony and hospitality features were exercised. All test cases passed with observations noted in **Section 2.2**.

9. References

This section references the Avaya documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *Avaya IP Office Manager*, Release 9.0, Issue 9.01, September 2013, Document Number 15-601011.
[2] *Cetis 3302IP VoIP Phone User's Manual*.
[3] *Cetis 9602IP VoIP Phone User's Manual*.

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