NEC

# UNIVERGE® SV9300

## AUDIOCODES MEDIAPACK FXS/FXO CONFIGURATION GUIDE

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# Chapter 1 *Introduction*

## SECTION 1 OVERVIEW

The MediaPack series analog Voice-over-IP (VoIP) Session Initiation Protocol (SIP) media gateways (hereafter referred to as device) are cost-effective, cutting edge technology products. These stand-alone analog VoIP devices provide voice technology for connecting legacy telephones, fax machines and Private Branch Exchange (PBX) systems.

The device is best suited for small and medium-sized enterprises (SME), branch offices, or residential media gateway solutions. The device also provides SIP trunking capabilities for Enterprises operations.

The device provides FXO and/or FXS analog ports for direct connection to an enterprise's PBX (FXO), and / or to phones, fax machines, and modems (FXS).

The device is a compact unit that can be easily mounted on a desktop, wall, or in a 19-inch rack.

The user-friendly, Web interface provides remote configuration using any standard Web browser (such as Microsoft™ Internet Explorer™).

✎ *The default User name and Password is "**admin**" for both.*

✎ *Refer to the AudioCodes User's manual for detailed information.*

✎ *The device is shipped with the default IP address of 10.1.10.10/16.*

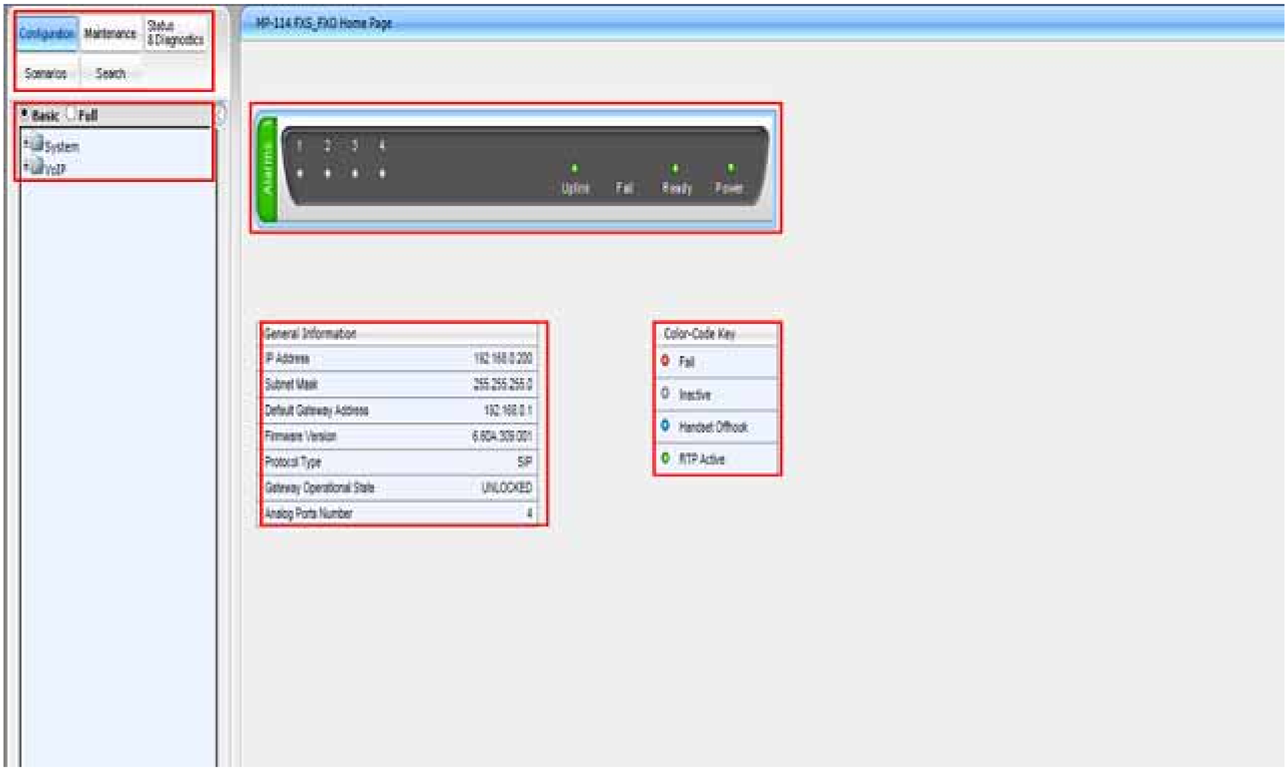
## SECTION 2 LIMITATIONS

- ❑ There are 3 supported configurations, FXS standalone, FXO standalone and FXS/FXO combination.
- ❑ This document is for reference only and actual screens may differ slightly if other than MP-114 FXS/FXO is used.
- ❑ Loop DID wink is not supported on the SV9300.
- ❑ Faxing is supported with G711 voice for FXS configuration.
- ❑ Faxing is supported with T.38 for the FXS/FXO combination configuration.
- ❑ Message Waiting is not supported in the FXS/FXO combination configuration.
- ❑ FXS/FXO combination will have limited survivability.

## SECTION 3 AUDIO CODES OVERVIEW

The Audio Codes main screen is shown in the figure below.

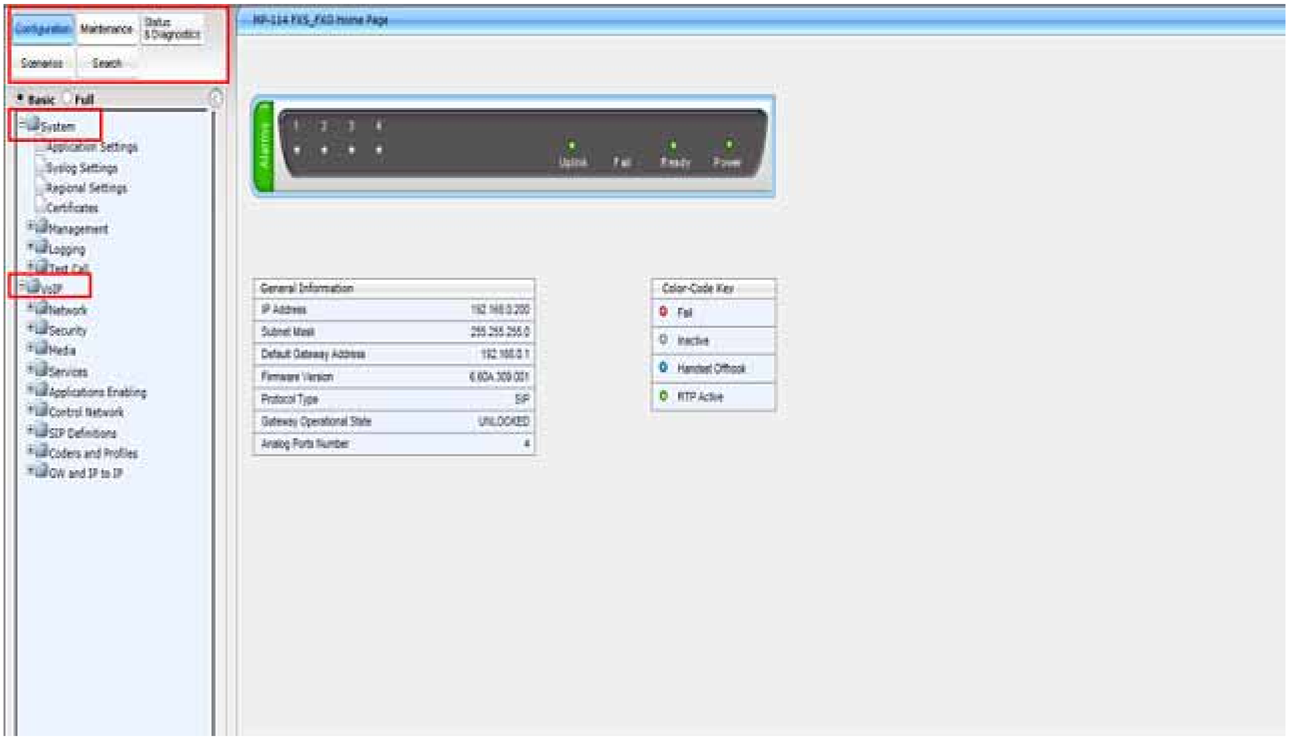
**Figure 1-1 Audio Codes Main Screen**




The Audio Codes main screen is composed of the following:

1. Several tabs for configuration purposes.
2. Shows the current LED indications.
3. Shows the current IP address, Subnet Mask, Default Gateway, Firmware version, Protocol type, Operational state and port numbers.
4. Has a color code chart for the LED's.
5. There are several tabs in the left window for configuration, maintenance, diagnostics, scenarios as well as a search window.

**Figure 1-2 Audio Codes Configuration Tab**




The following are features of the Configuration tab.

1. The configuration tab will show Basic or Full mode. Within each screen there may be additional settings that will not be available in basic mode. You will need to select the **Full** mode to see additional settings.

2. The configuration mode has a **System** tab and a **VOIP** tab.

3. The System tab has **Application** settings such NTP, Day light saving, Stun, NFS and DHCP. These are advanced settings and not part of this set up.

4. Syslog settings are for trouble shooting.

5. Regional settings are current time zone.

6. Certificates Management is an advanced setting and not part of this setup.

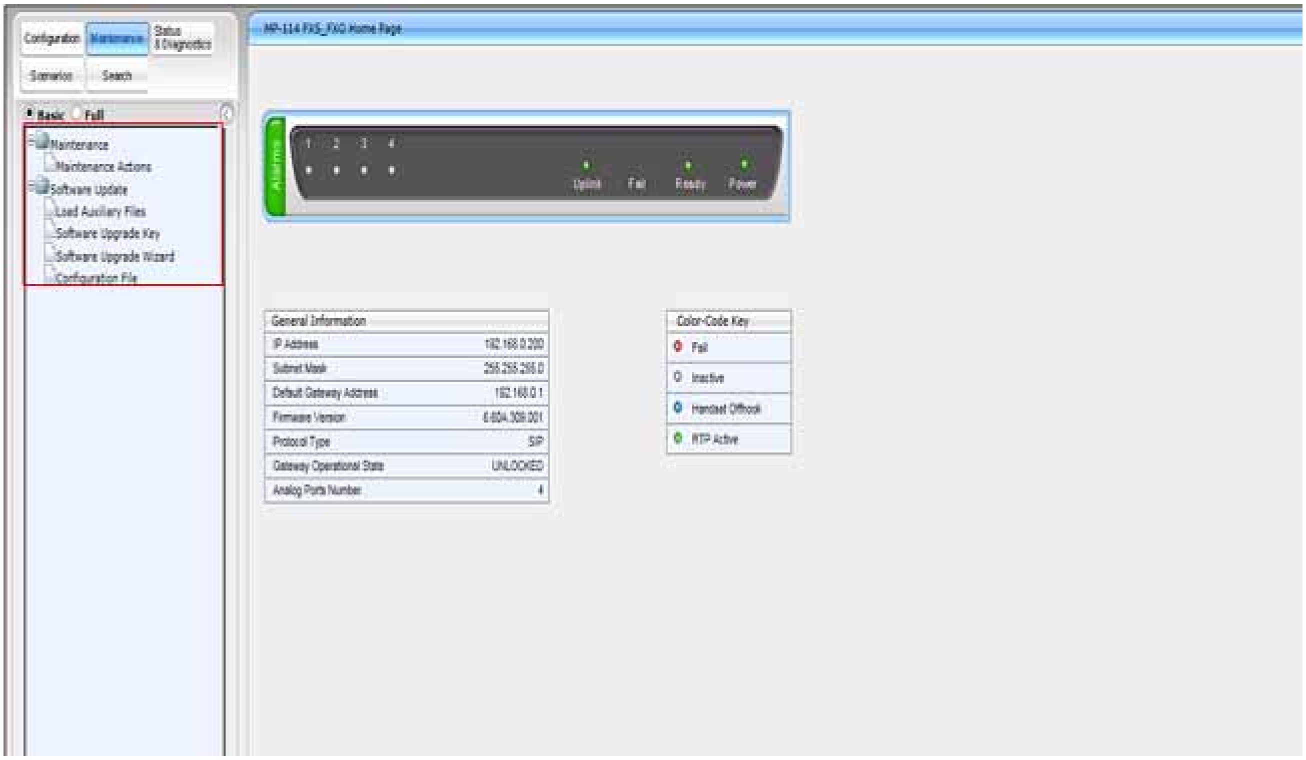
7. Management is advanced and not part of this setup.

8. Logging will not be used unless advised by Engineering.

9. Test call is advanced and not part of this setup.

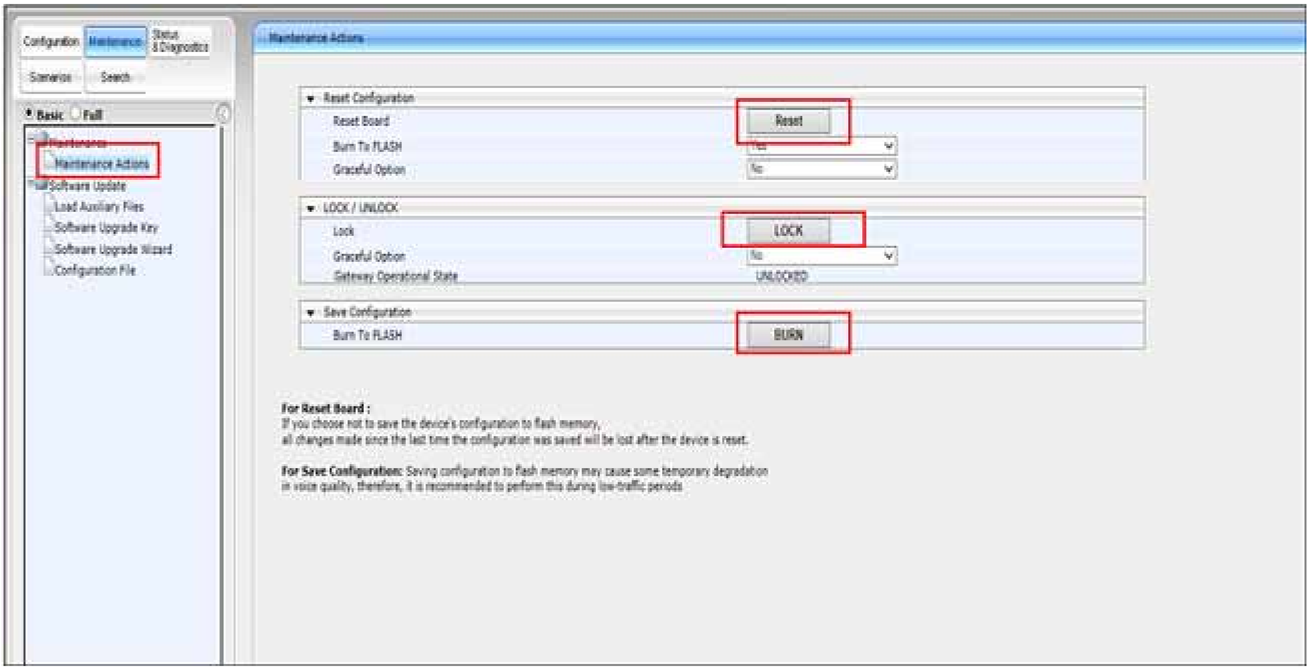
10. The **VOIP** tab has all of the configuration tabs and will be explained in more detail as we go.

**Figure 1-3 Maintenance Tab**




The Maintenance tab has both a **Maintenance** tab and a **Software Update** tab.

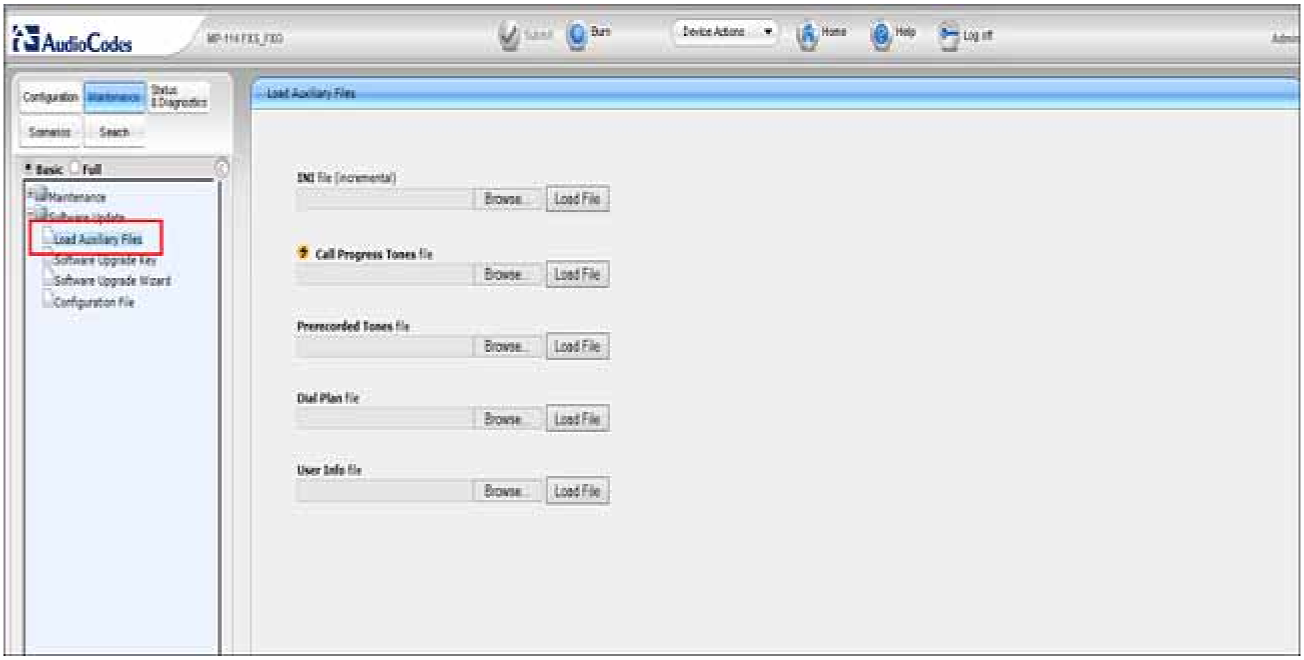
**Figure 1-4 Maintenance Actions**




The following actions may be performed from the **Maintenance** tab.

1. Reset Configuration can be set to **Burn to Flash** prior to the reset and also allows a Graceful option.
2. The Gateway can be locked. Normal operation is unlocked.
3. Save configuration Burns settings to **Flash**. You should always burn to flash prior to resetting.

**Figure 1-5 Load Auxiliary Files**



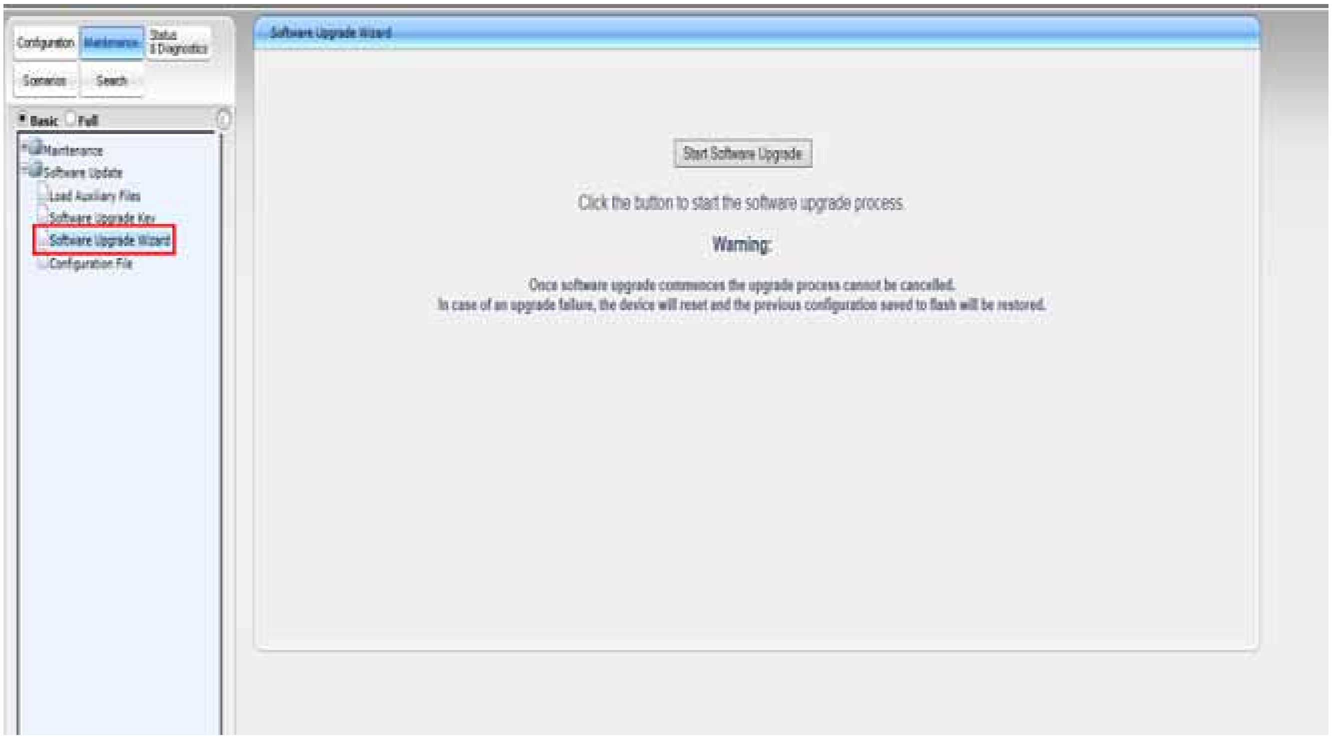
The Load Auxiliary Files tab can be found under the Software Update tab. This tab can be used to load an **INI** file, **Call Progress Tones**, **Prerecorded Tones**, **Dial Plan** and **User Info**.

Refer to the User's manual for configuration details.

**Figure 1-6 Software Upgrade Key**


The Software Upgrade Key is used to load additional license keys. Refer to the User's manual for configuration details.

**Figure 1-7 Software Upgrade Wizard**




The Upgrade Wizard will walk you through an upgrade if needed.

**Figure 1-8 Configuration File**

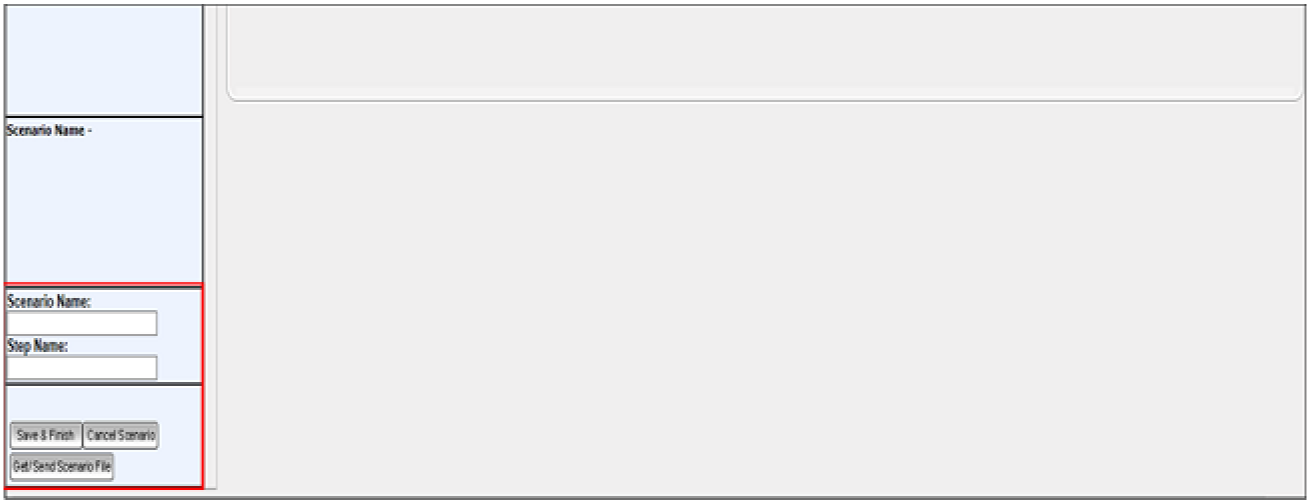

The Configuration tab is used for saving or loading an INI file. The INI file can be downloaded and saved as a backup. It is recommended to save the INI after any major changes. Changes can also be made to the INI file and reloaded. Refer to the User's manual for additional information.

**Figure 1-9 Statistics and Diagnostics**

The Statistics & Diagnostics tab consists of **Message log**, **Device Information**, **Ethernet Port Information**, and **Carrier Grade Alarms**.

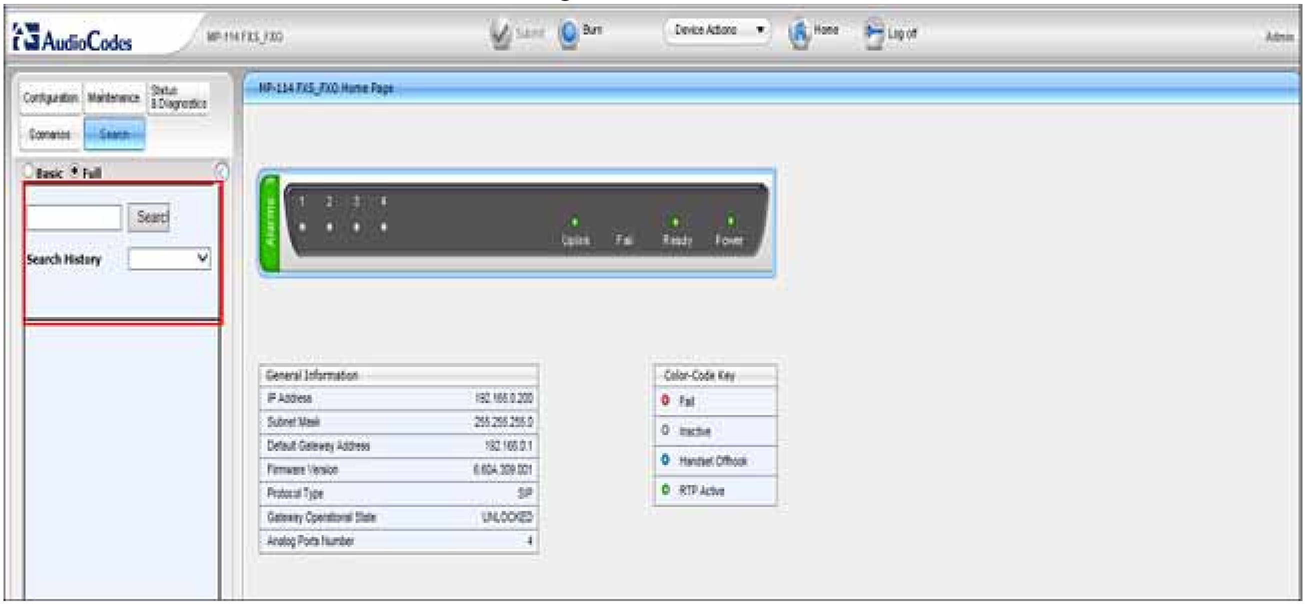
The VOIP Status tab consists of **IP Interface Status**, **Performance Statistics**, **IP to Tel Calls Count**, **Tel to IP Calls Count**, **Call Routing Status**, and **Registration Status**. Refer to the User's manual for additional information.

**Figure 1-10 Scenarios**




The Scenario tab will have a wizard to assist in building scenarios. Refer to the User's manual for additional information.

**Figure 1-11 Search**



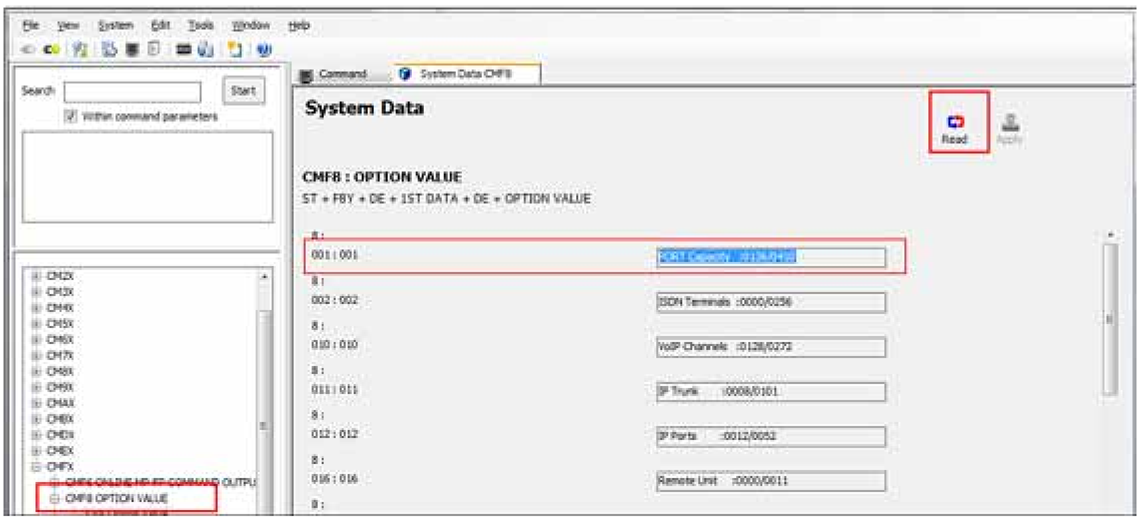

The Search tab provides a way to search for any particular item in the AudioCodes.

# Chapter 2 *SV9300 Configuration*

## SECTION 1 PROGRAM SV9300 FOR STATION CONNECTIVITY TO AUDIOCODES

The SV9300 will need to be programmed to use Standard SIP stations for connectivity to the AudioCodes.

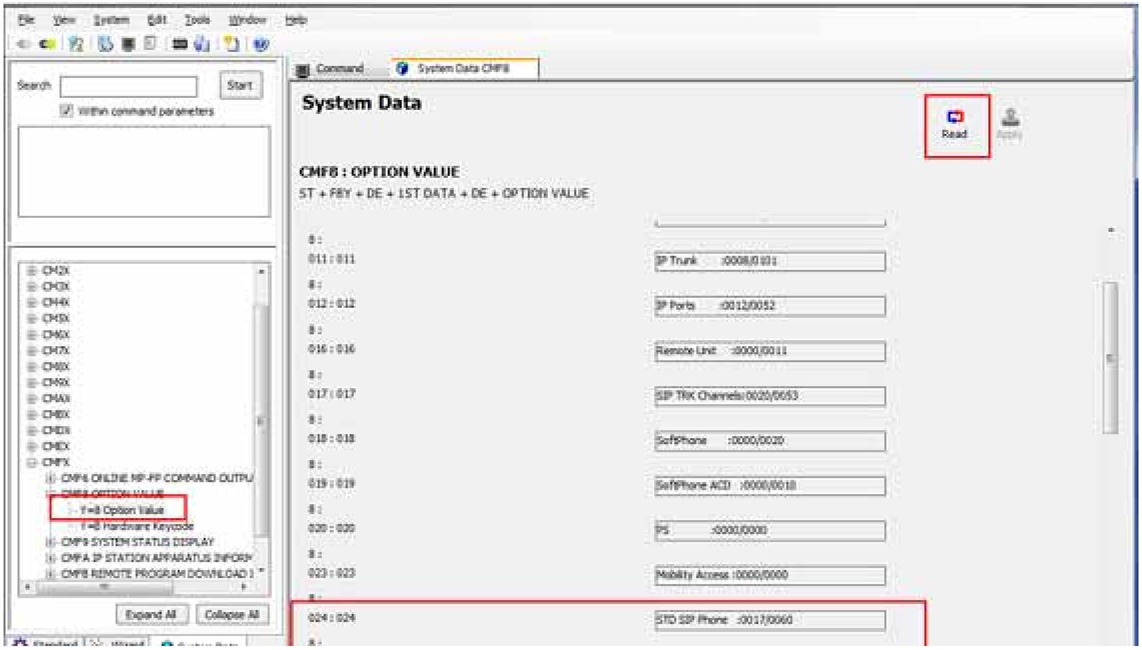
**Figure 2-1 CM F88 Port Capacity**




1. Confirm port license are available. Read is automatic for this command. Example: **CMF8 Y=8>001>0126/0410**

2. Click **Apply**.

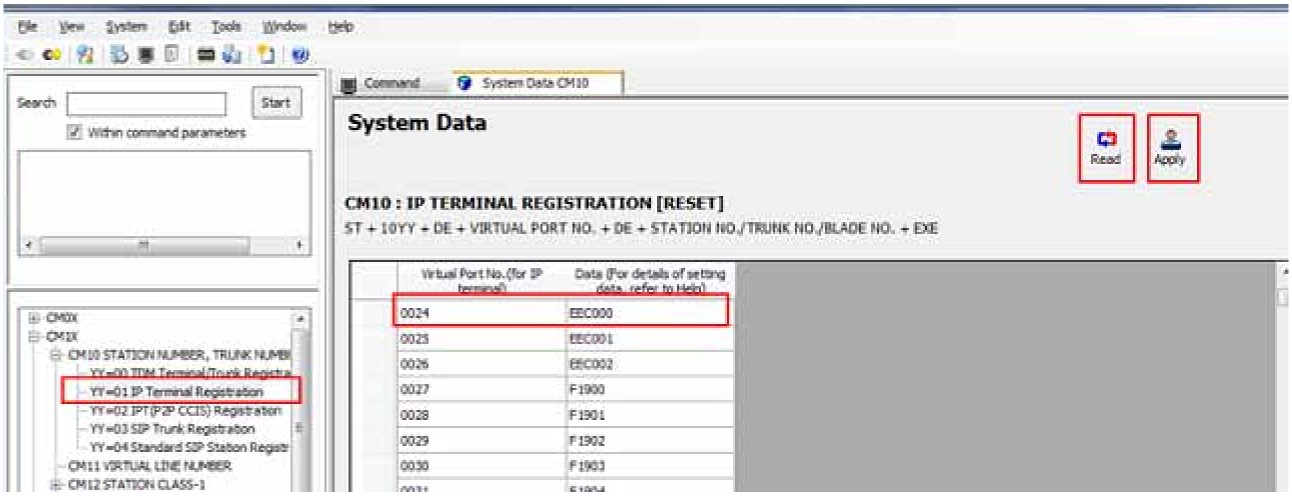
**Figure 2-2 CM F88 STD SIP Station Capacity**




1. Confirm STD SIP phone license are available. Read is automatic for this command.
   Example: **CMF8 Y=8>024>0017/0060**

2. Click **Apply**.

**Figure 2-3 CM 1001 SIP Converter No.**



1. Assign the **SIP Converter No**. to the Virtual port number for Standard SIP station.

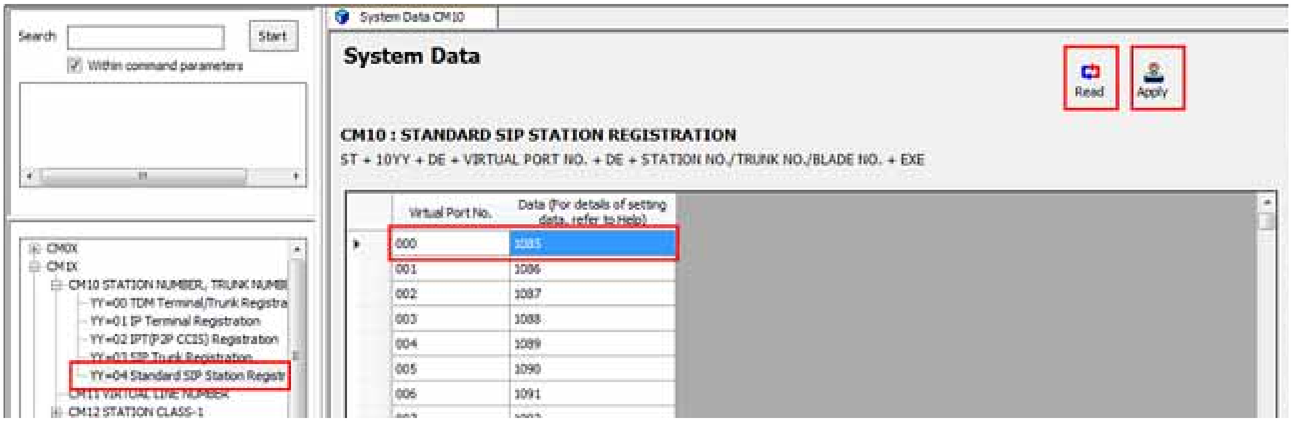
2. Click **Read** to get the current settings.
   Example: **CM10 Y=01>0024>EEC000**

3. Click **Apply**.

✎ *SIP converter supports 3 Simultaneous calls by the Standard SIP stations. A system reset is required.*

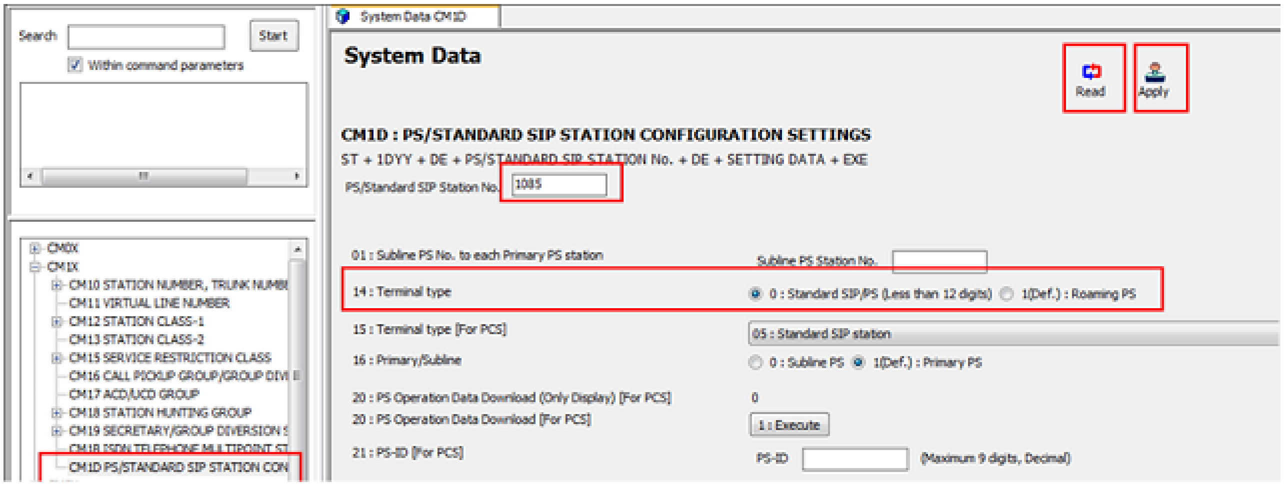
✎ *A reset of the Standard SIP station is required after the system reset.*

**Figure 2-4 CM 1004 Standard SIP Station**




1. Assign the **Standard SIP station number** to the virtual port number.
2. Click **Read** to get the current settings.
   Example: **CM10 Y04>000>1085**
3. Click **Apply**.

**Figure 2-5 CM 1DXX Standard SIP Station Terminal Type**



1. Set the terminal type to **Standard SIP station** for the station number assigned by CM10 Y=04.

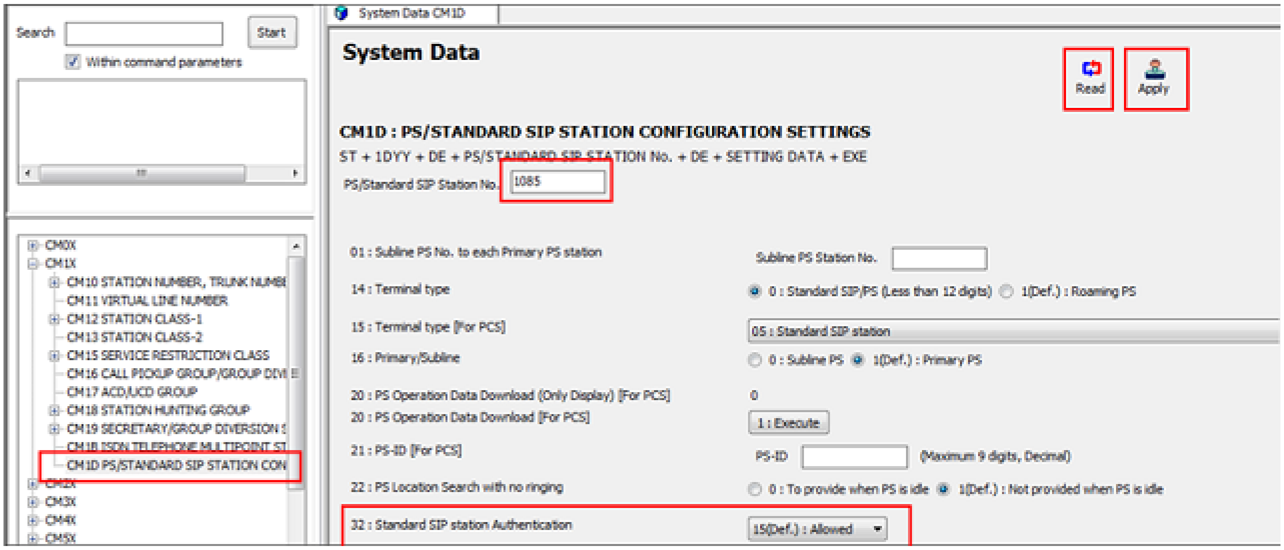
2. Click **Read** to get the current settings.
   Example: **CM1D Y14>1085>0**

3. Set the terminal type to the **Standard SIP station** for the station the number assigned by CM10 Y=04.

4. Click **Read** to get the current settings.
   Example: **CM1D Y15>1085>05**

5. Click **Apply**.

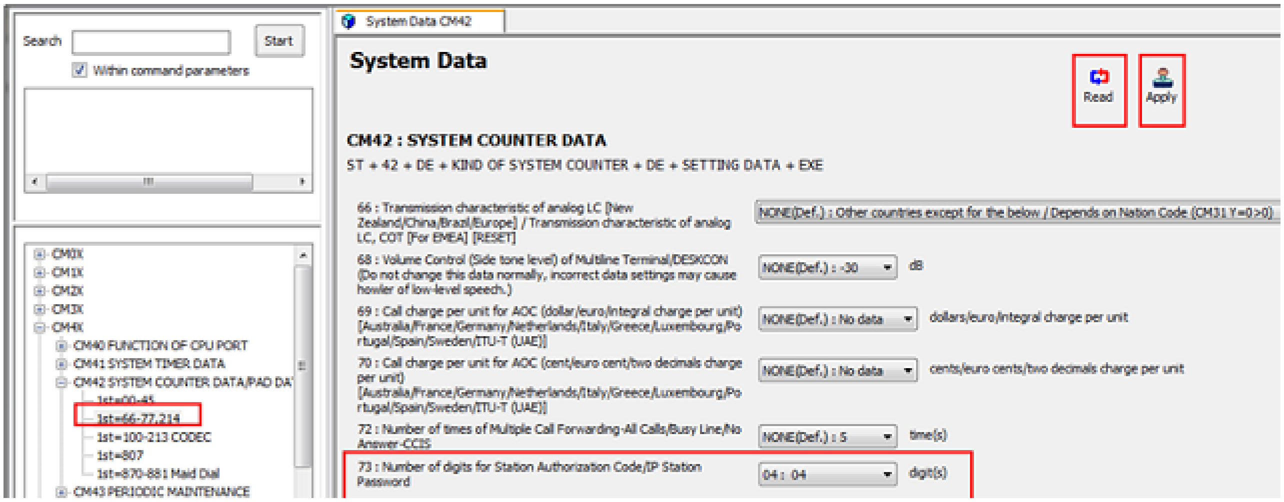
**Figure 2-6 CM 1DXX Standard SIP Station Authentication**




1. Allow Authentication to the Standard SIP phone. Click **Read** to get the current settings.
   Example: **CM1D Y=32>1085>15** (Default).

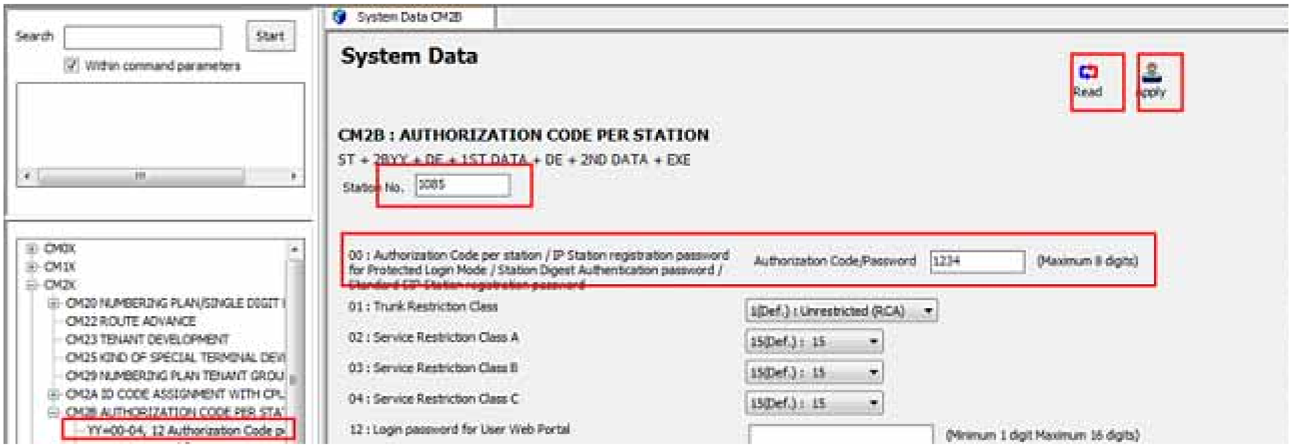
2. Click **Apply** if changes are made.

**Figure 2-7 CM 42XX Maximum number of digits for Standard SIP registration password**




1. Specify the maximum number of digits for Standard SIP registration password.
2. Click **Read** to get the current settings.
   Example: **CM42>73>04** (Default).
3. Click **Apply** if changes are made.

**Figure 2-8 CM 2B00 Station Authorization Code to each Standard SIP station**

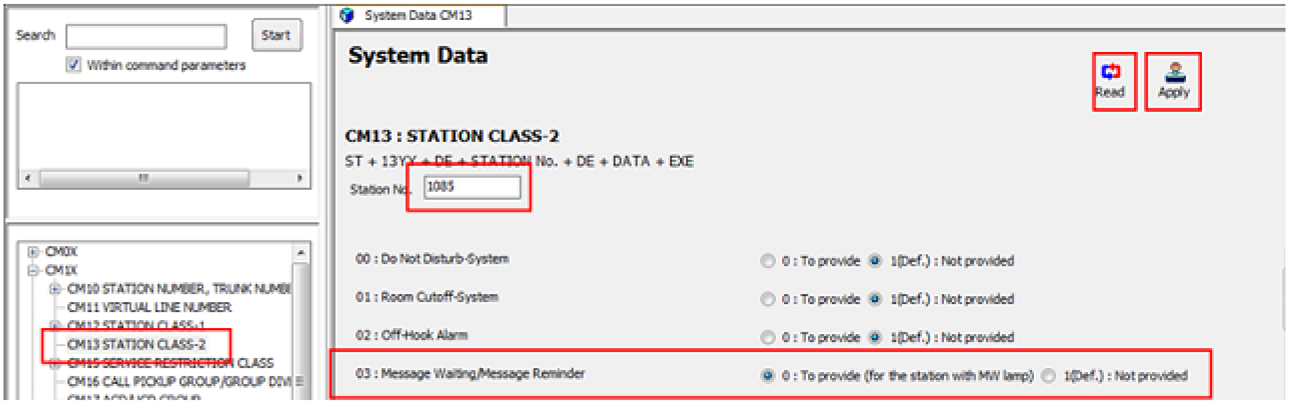


1. Assign a Station Authorization Code to each Standard SIP station.

2. Click **Read** to get the current settings.
   Example: **CM2B Y=00>1085>1234**.

3. Click **Apply** if changes are made.

**Figure 2-9 CM 1303 Message Waiting is provided**

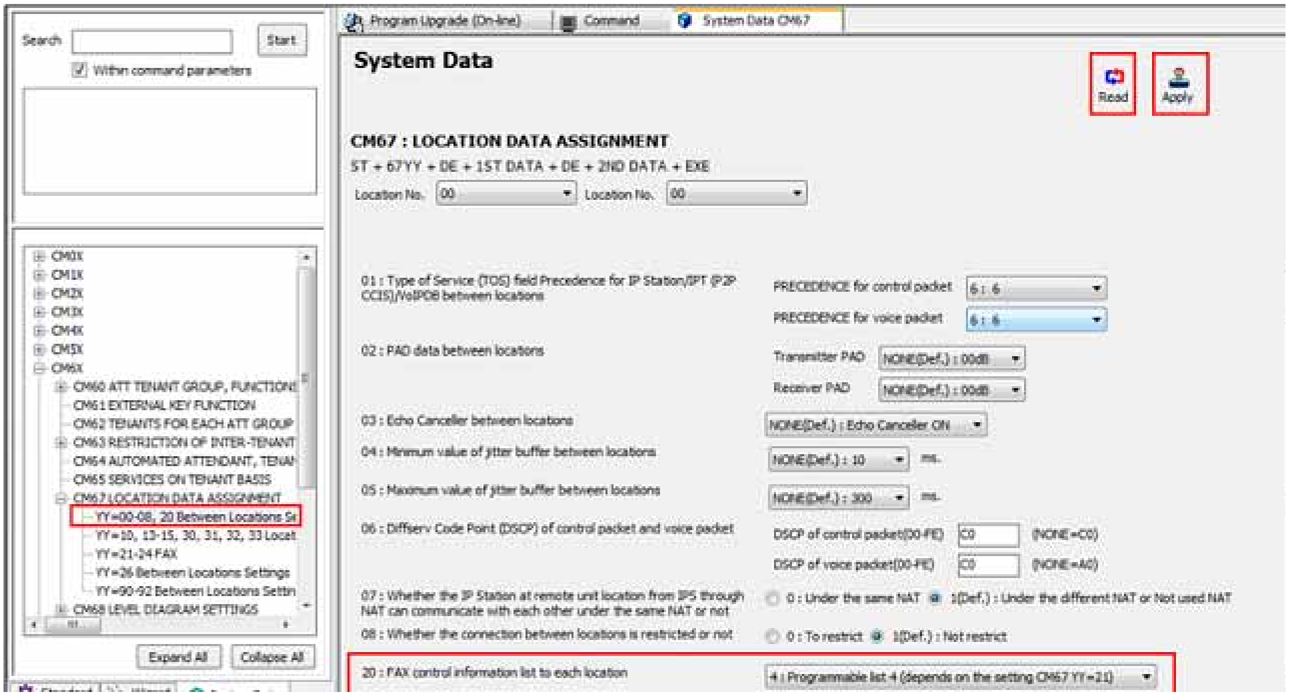



1. Specify whether the Message Waiting is provided to the Standard SIP station assigned by CM10 Y=04.

2. Click **Read** to get the current settings.
   Example: **CM13 Y=03>1085>0**

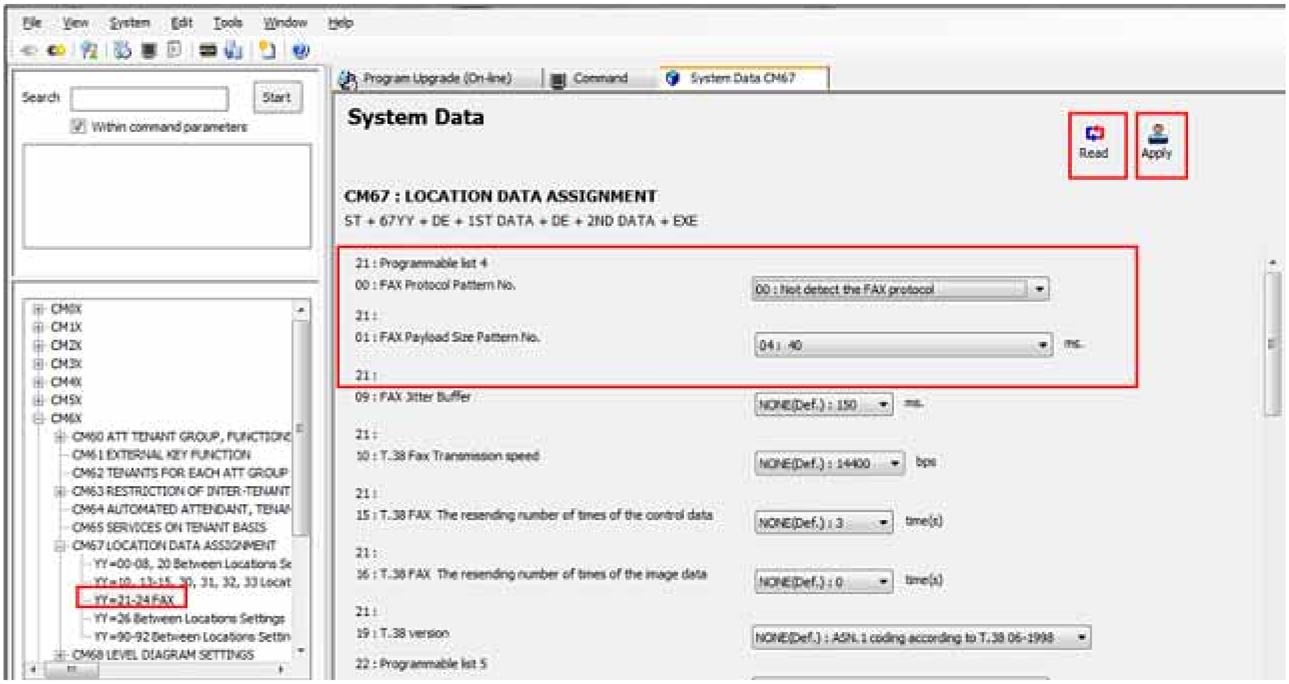
3. Click **Apply**.

**Figure 2-10 CM 67XX Assign Fax control information list**




1. Enter the location group number in both areas.
2. Click **Read** to get the current settings.
3. Assign it as programmable list 4.
   Example: **CM67 Y=20>0000>**
4. Click **Apply**.

**Figure 2-11 CM 67XX Assign Fax Protocol**




1. Assign the fax protocol to the programmable list.

2. Click **Read** to get the current settings.

3. Assign Fax Control pattern No. as **Not detect the Fax protocol**.
   Example: **CM67 Y=21>00>00**

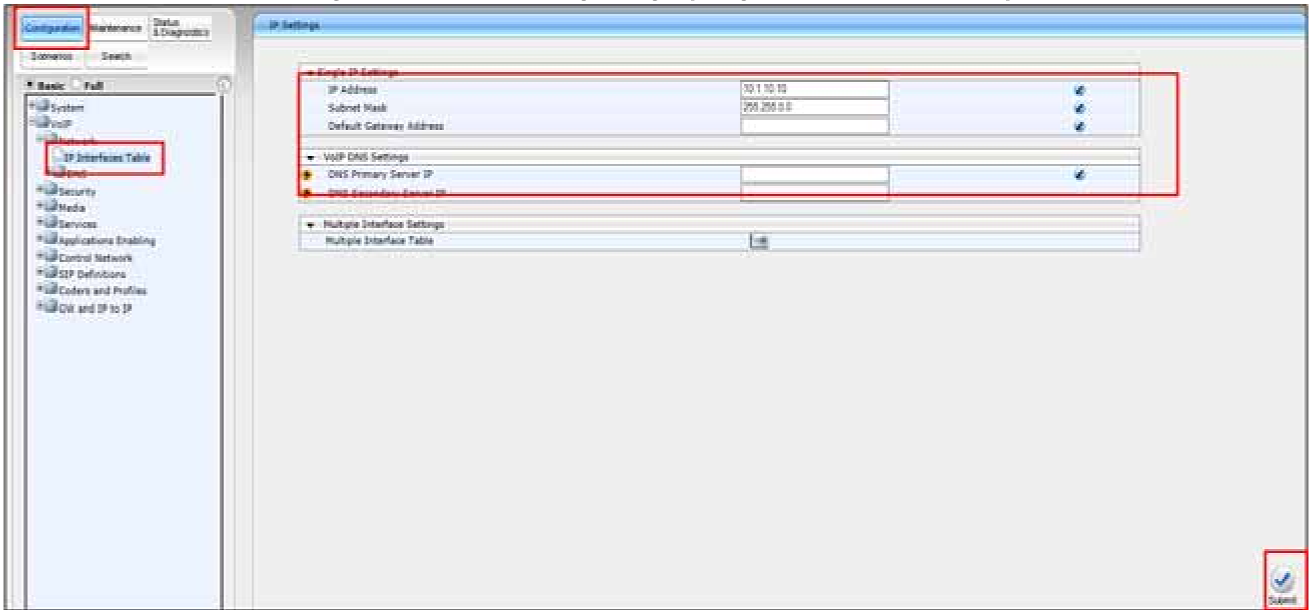
4. Assign Fax Payload to 40 msc.
   Example: **CM67 Y=21>01>04**

5. Click **Apply**.

   Refer to the **SV9300 System Manual** for additional programming for the Standard SIP Terminal.

## SECTION 2 AUDIO CODES STATION CONFIGURATION

**Figure 2-12 IP Settings Page (Single Network Interface)**



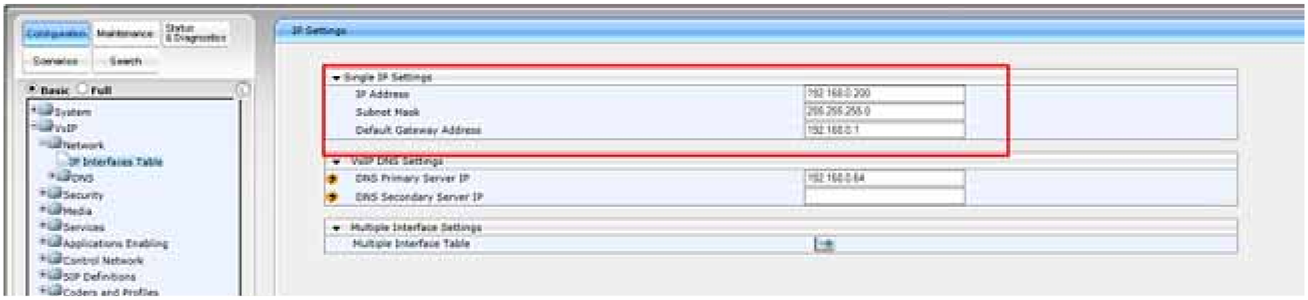

1. Open the **IP Settings** page (**Configuration tab-VoIP menu-Network-IP settings**).

2. The initial IP address is **10.1.10.10**.

3. Enter the **IP Address**, **Subnet Mask**, **Gateway** and **DNS Address** (if required).

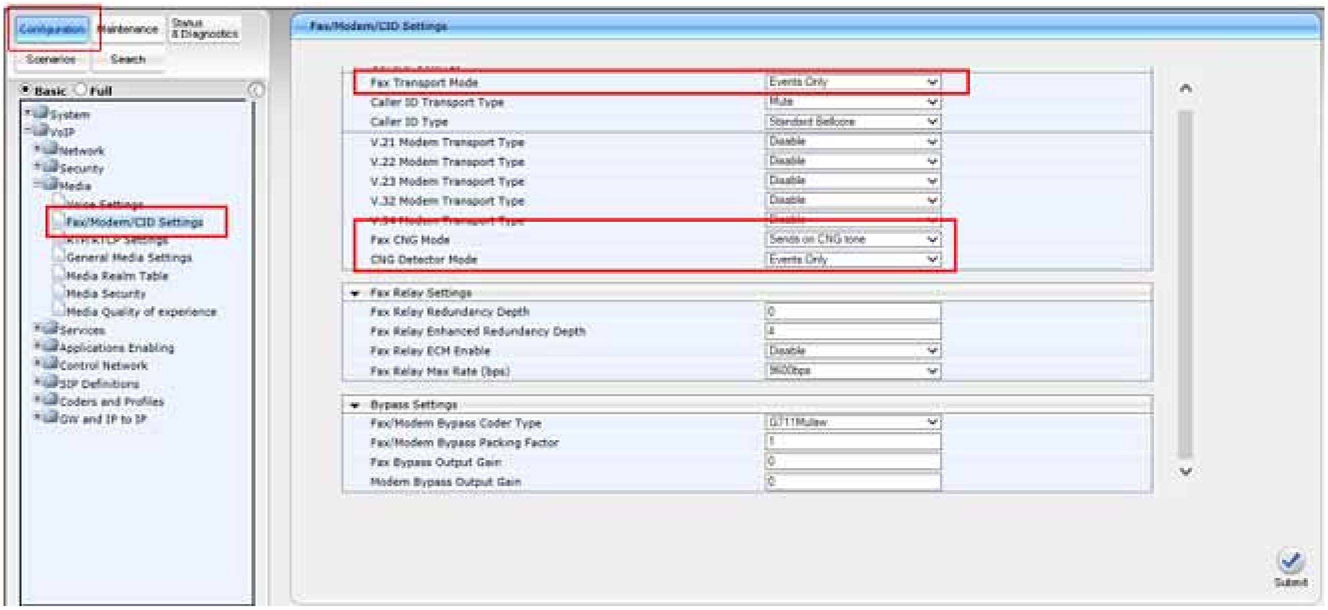
4. Click **Submit**.

**Figure 2-13 Test Configuration Example**



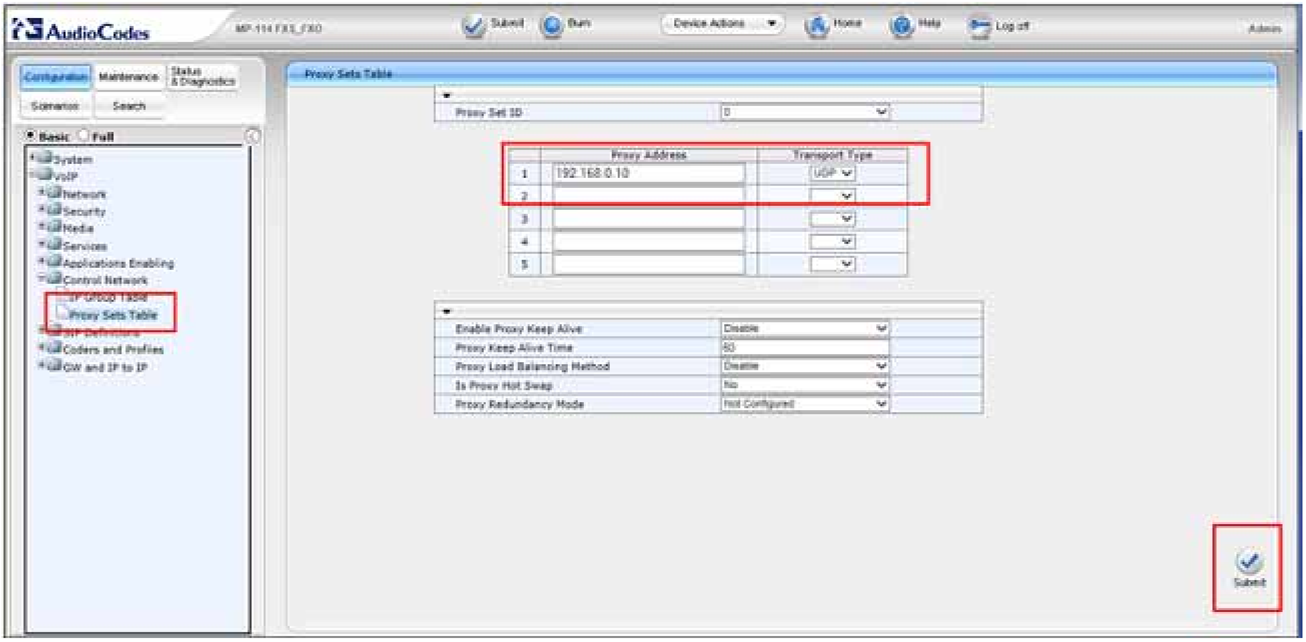

1. Assigned IP address **192.168.0.200**.
2. Assigned Subnet Mask of **255.255.255.0**.
3. Assigned Gateway **192.168.0.1**.
4. Assigned DNS address of **192.168.0.64** (only if required).

**Figure 2-14 Fax/Modem/CID Settings Page**



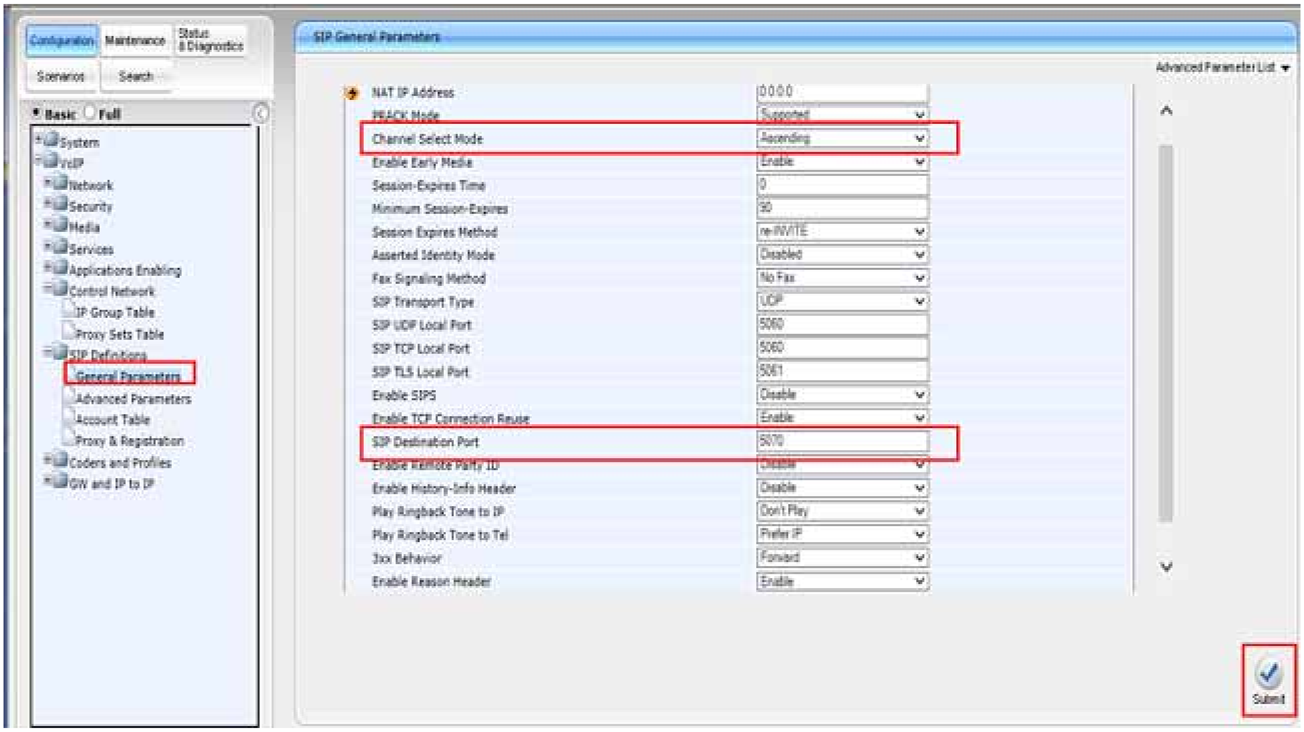

1. Open the **Fax/Modem/CID Settings** page (**Configuration tab-VoIP menu-Media- Fax/Modem/CID settings**).
2. Assign **Fax transport Mode** to **Events Only**.
3. Assign **Fax CNG Mode** to **Sends on CNG tone**.
4. Assign **CNG detector Mode** to **Events Only**.
5. Click **Submit**.

**Figure 2-15 Proxy Sets Table Page**



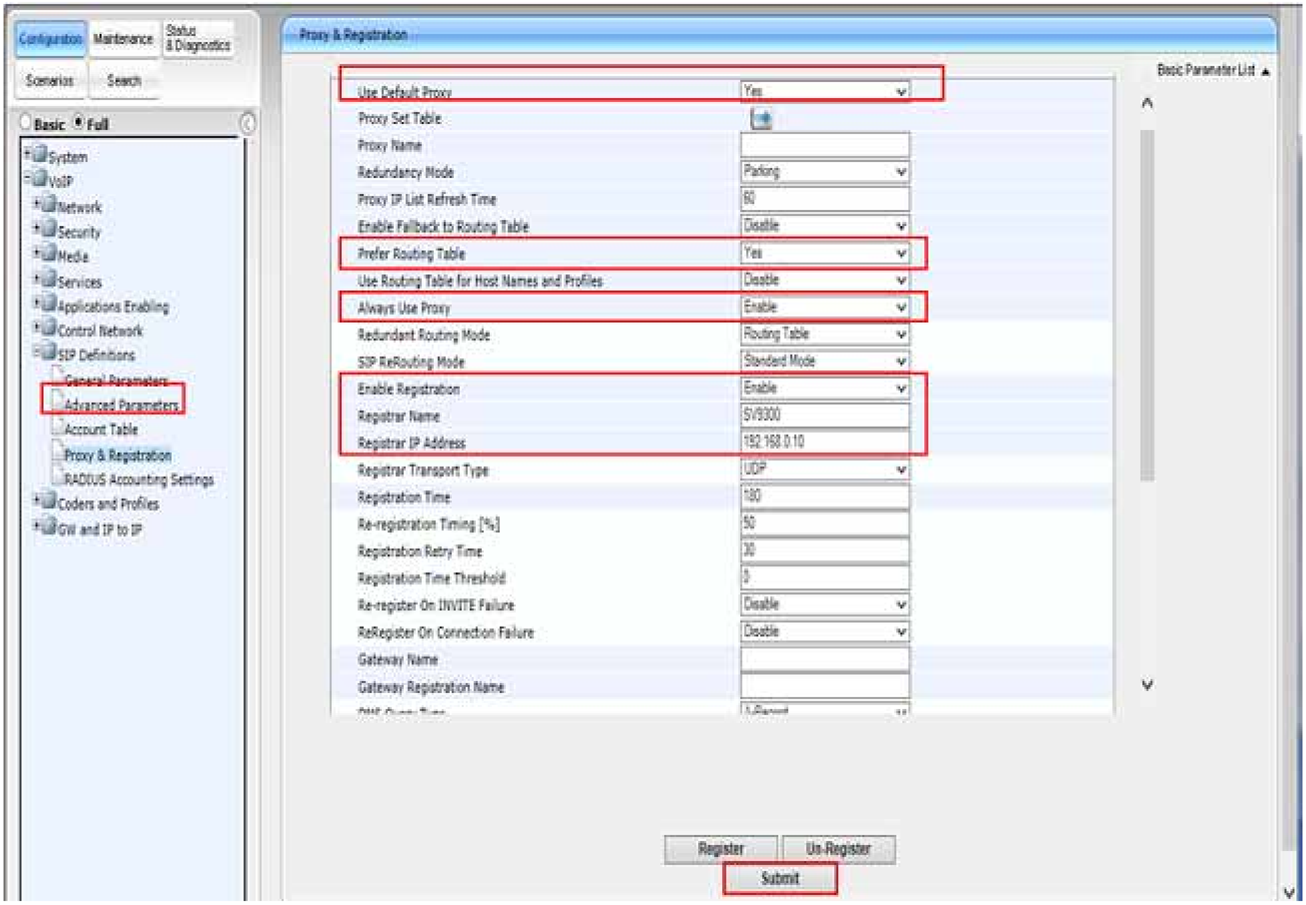

1. Open the **Proxy Sets Table** page (**Configuration-VoIP Menu-Control Network-Proxy Sets Table**).
2. Select a **Proxy Set ID** from the drop-down list under **Transport Type**.
3. Configure the **Proxy** as required (IP address of the SV9300).
4. Leave all other settings at default.
5. Click **Submit**.

**Figure 2-16 SIP Definitions General Parameters**



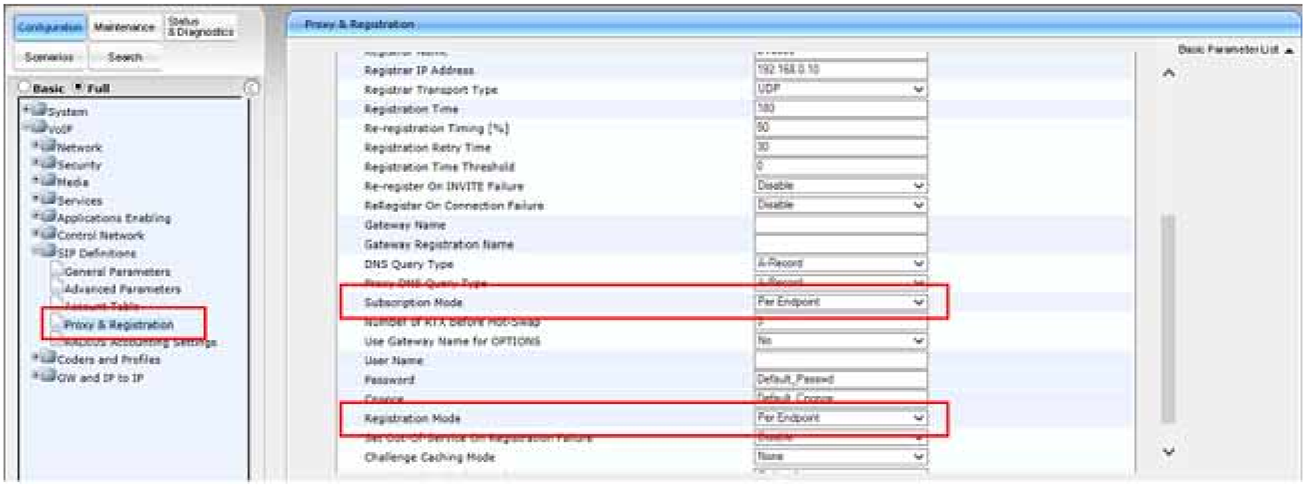

1. Open **SIP Definitions General Parameters** (**Configuration tab-VoIP menu-SIP Definitions-General Parameters**).
2. Set the **Channel Select Mode** to **Ascending**.
3. Set the **SIP Destination Port** to **5070**.
4. Click **Submit**.

**Figure 2-17 SIP Definitions Proxy & Definitions**



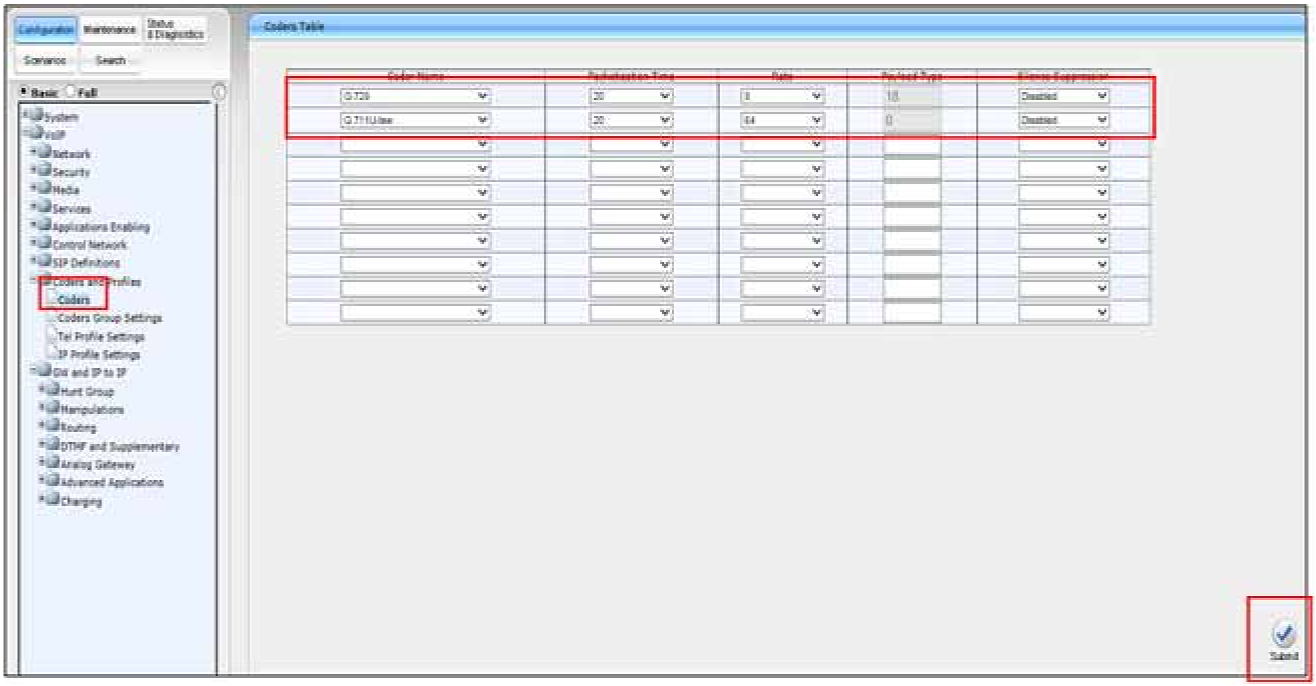

1. Open the **Proxy & Registration** page (**Configuration tab-VoIP menu-SIP Definitions-Proxy and Registration**).
2. Set **Use Default Proxy** to **Yes**.
3. Set **Prefer Routing Table** to **Yes**.
4. Set **Always Use Proxy** to **Enable**.
5. Set **Registrar Name** as **SV9300**.
6. Set **Registrar IP Address** to the SV9300 IP address (EX: 192.168.0.10).
7. Click **Submit**.

**Figure 2-18 SIP Definitions Proxy & Definitions Continued**



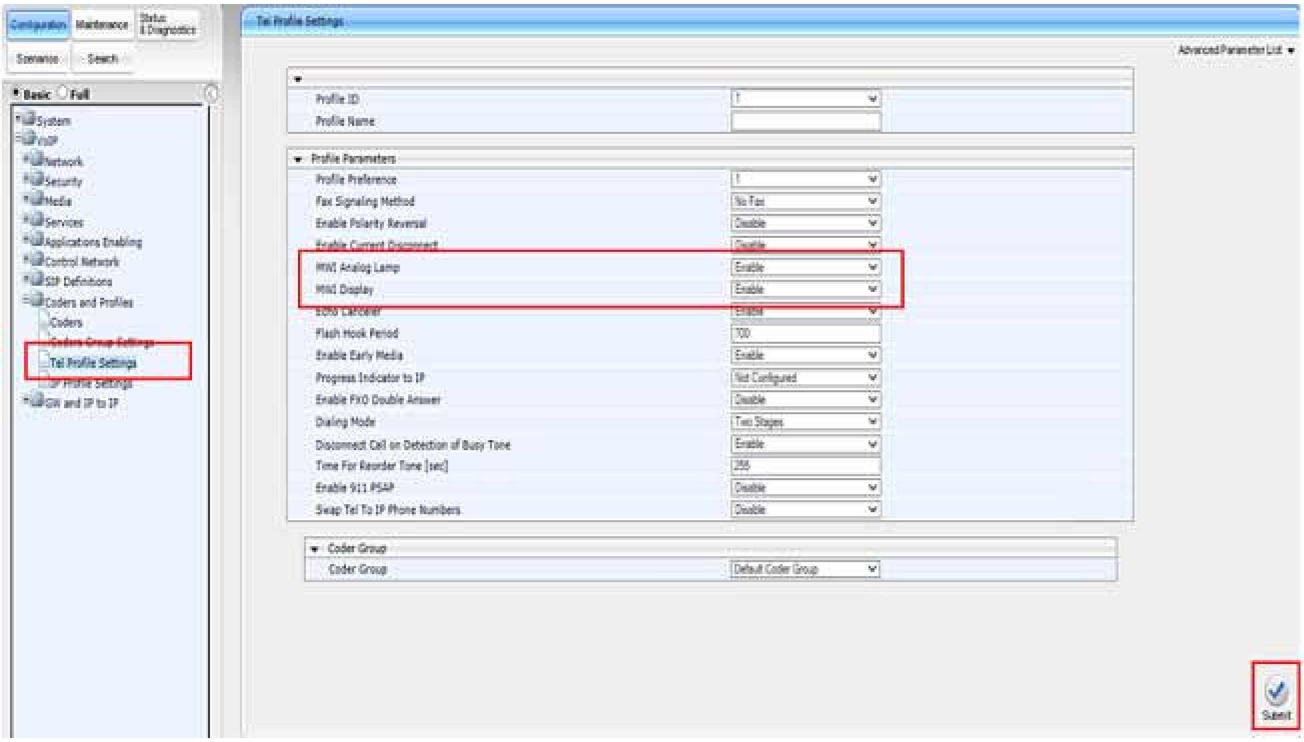

1. Open the **Proxy & Registration** page (**Configuration tab-VoIP menu-SIP Definitions-Proxy and Registration**).
2. Set **Subscription Mode** to **Per Endpoint**.
3. Set **Registration Mode** to **Per Endpoint**.
4. Click **Submit**.

**Figure 2-19 Coders Table Page**



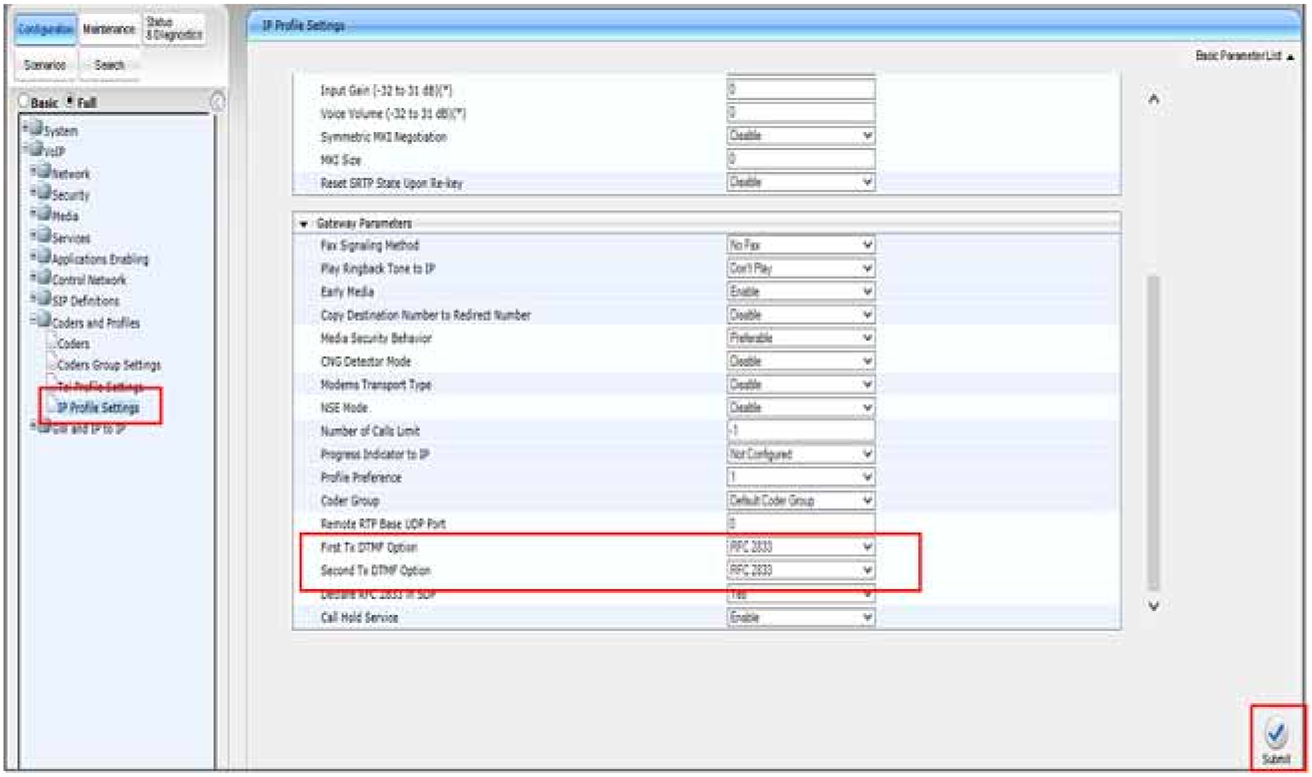

1. Open the **Coders** page (**Configuration- VoIP menu-Coder** and **profiles-Coders**).
2. Under the **Coder Name** drop-down list, select the required coder.
3. Under the **Packetization Time** drop-down list, select the packet size.
4. Under the **Silence Suppression** drop-down list select the desired option.
5. Repeat steps 2 through 6 for the next optional coders.
6. Click **Submit**.

**Figure 2-20 Tel Profile Settings**



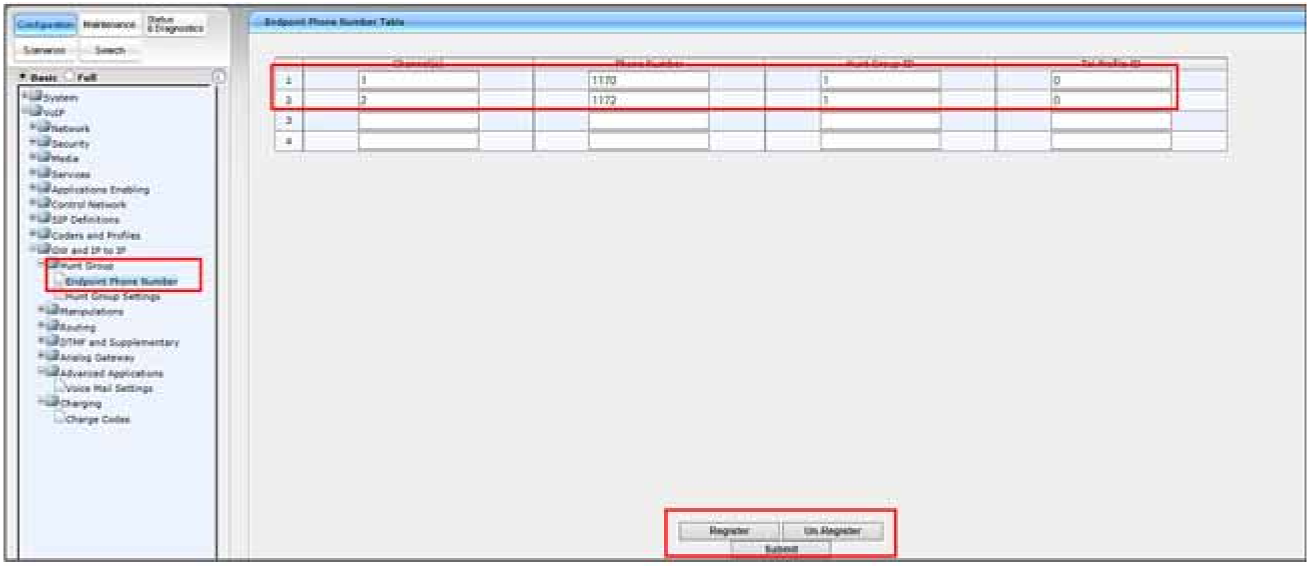

1. Open the **Tel Profile Settings** page (**Configuration tab-VoIP-Coders** and **Profiles- Tel Profile Settings**).
2. Set **MWI Analog Lamp** to **Enable**.
3. Set **MWI Display** to **Enable**. (Only required if terminal has a display.).
4. Click **Submit**.

**Figure 2-21 IP Profile Settings**



1. Open the **IP Profile Settings** page (**Configuration tab-VoIP menu- Coders** and **Profile- IP Profile Settings**).
2. Assign **First Tx DTMF Option** to **RFC 2833**.
3. Assign **Second Tx DTMF Option** to **RFC 2833**.
4. Click **Submit**.

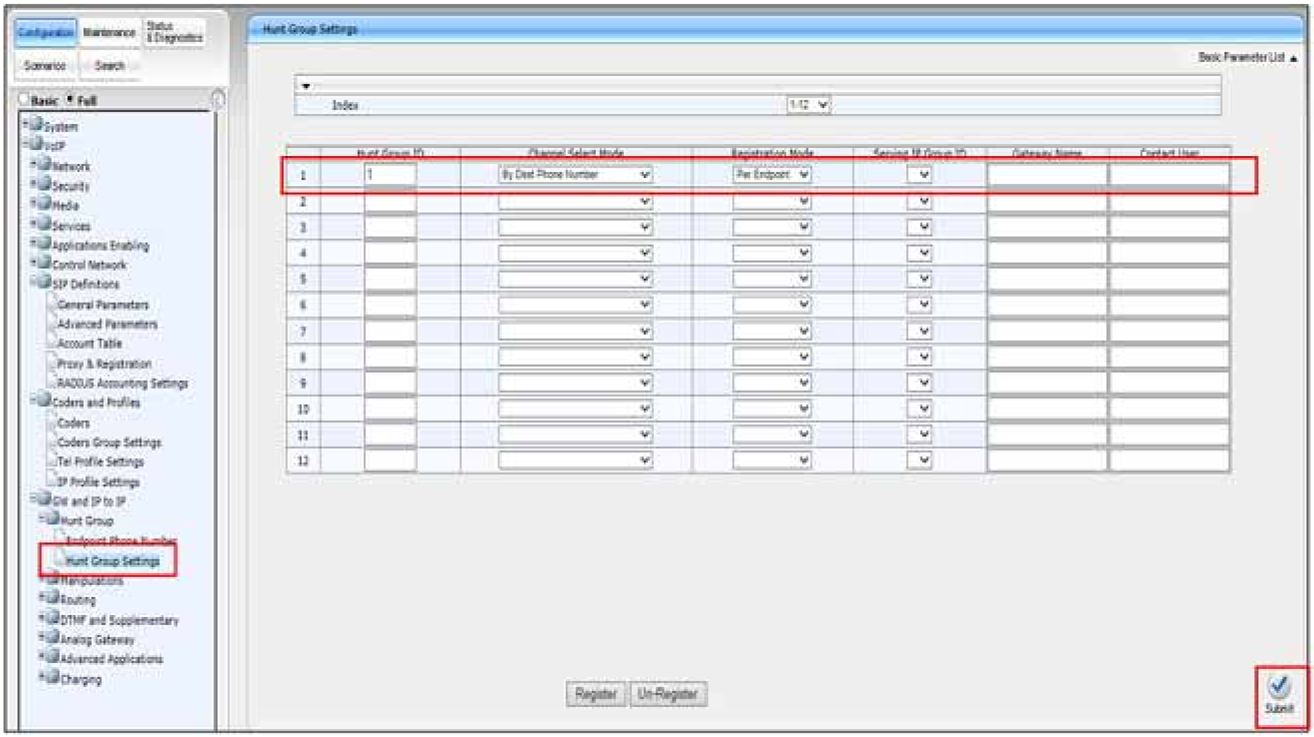
**Figure 2-22 Endpoint Phone Number Table Page**



1. Open the **Endpoint Phone Number Table** page (**Configuration tab- VoIP- GW** and **IP to IP- Hunt Group- Endpoint Phone Number**).
2. Configure the **Endpoint Phone Numbers** to the Channels. (EX: 1170, 1171).
3. Channels 3 and 4 will not be used in this configuration.
4. Assign the **Hunt Group ID** (Default is 1).
5. Assign **Tel Profile ID**. (Optional)
6. Click **Submit**.

✎ *To register an endpoint to a Proxy/Registrar server, click the **Register** button; to un-register an endpoint, click **Un-Register**.*

**Figure 2-23 Hunt Group Settings Page**



1. Open the **Hunt Group Settings** page (**Configuration-VoIP- GW** and **IP to IP-Hunt Group-Hunt Group settings**).

2. From the **Index** drop-down list, select the range.

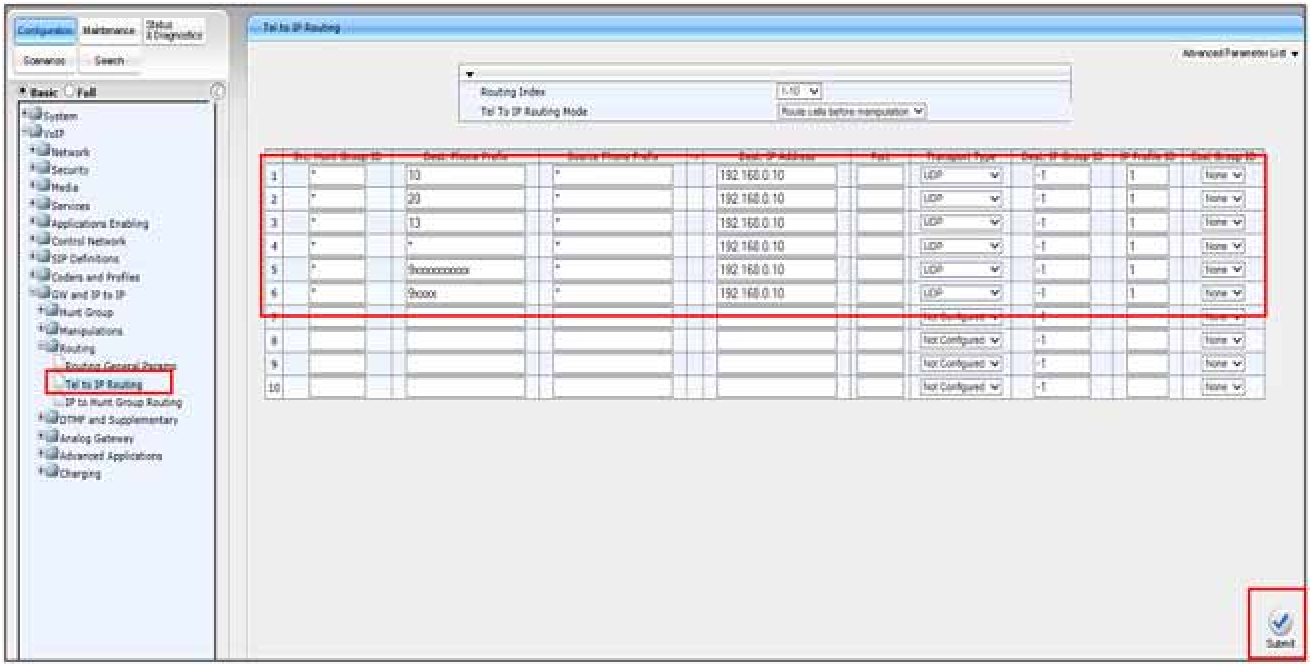
3. Assign **Hunt group ID**: 1.

4. Assign the **Channel Select Mode** to: **By Dest Phone Number**.

5. Assign **Registration Mode** to **Per Endpoint**.

6. Click **Submit**.

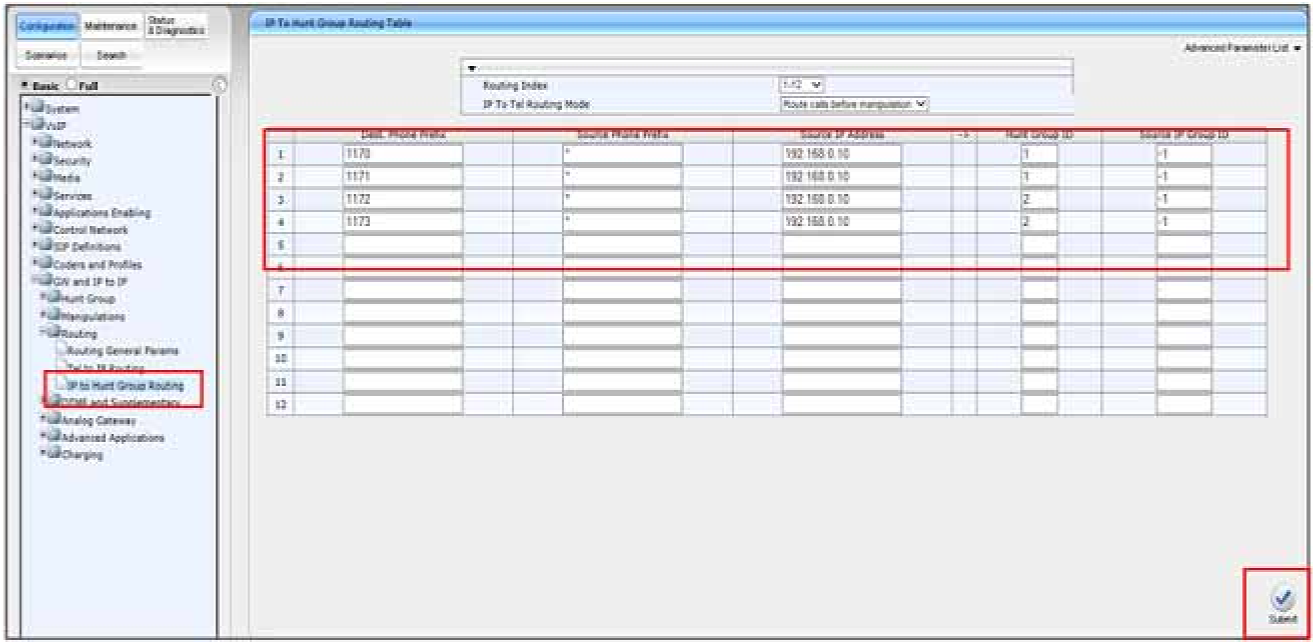
**Figure 2-24 Configure Tel to IP Routing**



1. Open **Tel to IP Routing** (**Configuration- VoIP menu- GW** and **IP to IP-Routing- Tel to IP**).
2. From the **Routing Index** drop-down list, select the range of entries that you want to add.
3. Configure the routing rule as required.
4. Click **Submit** to apply your changes.

✎ *Refer to the User's manual Tel to IP section for a detailed explanation of the routing parameters.*

**Figure 2-25 Configure IP to Hunt Group Routing**



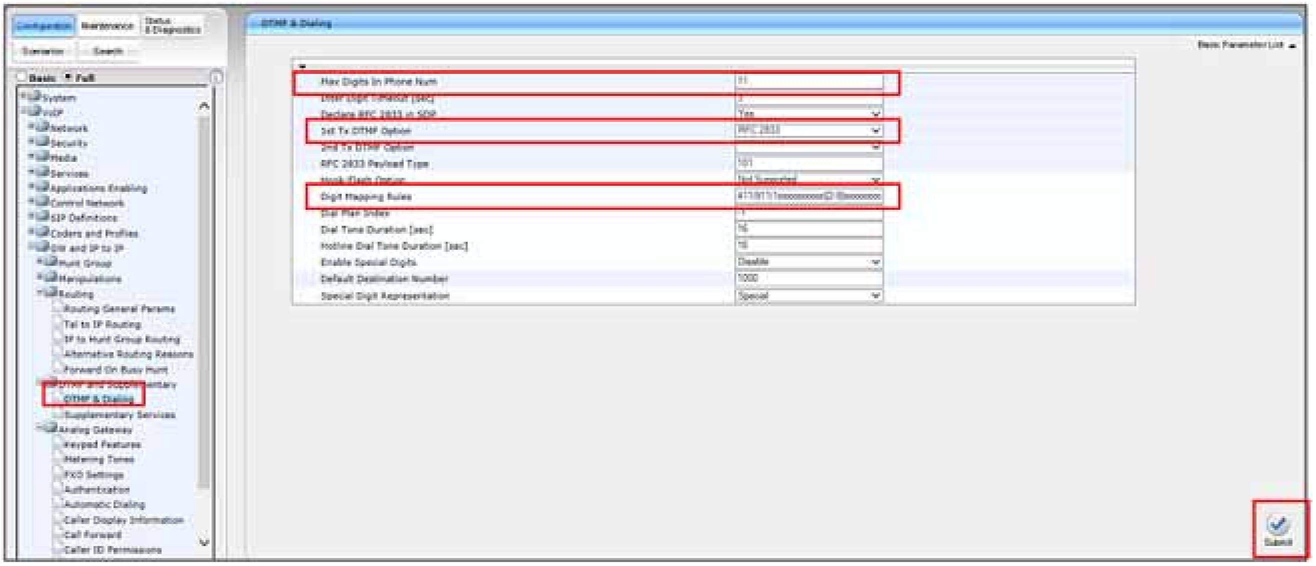
1. Open the **IP To Hunt Group Routing Table** page (**Configuration tab- VoIP menu- GW** and **IP to IP- Routing- IP to hunt group routing**).

2. Configure the routing rule as required.

3. Click **Submit**.

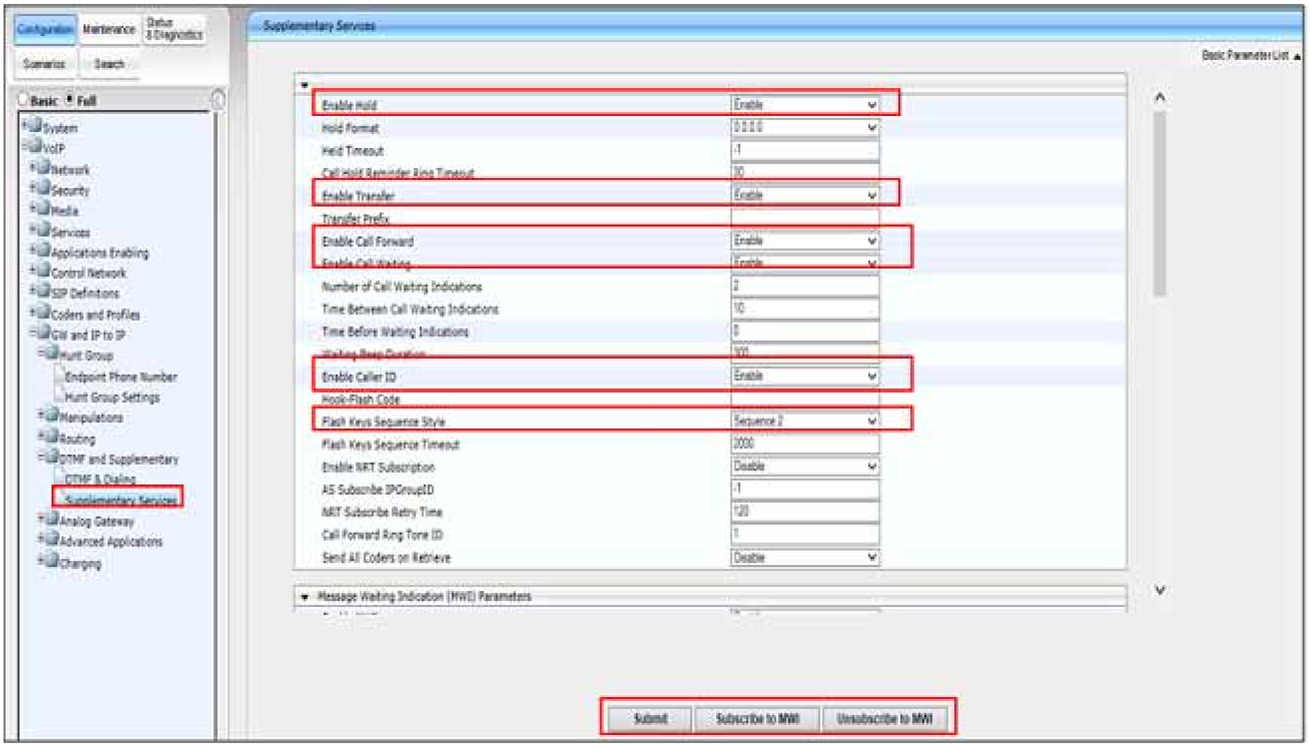
✎ *Refer to the User's Manual IP to Hunt group section for a detailed explanation of the routing parameters.*

**Figure 2-26 DTMF & Dialing**



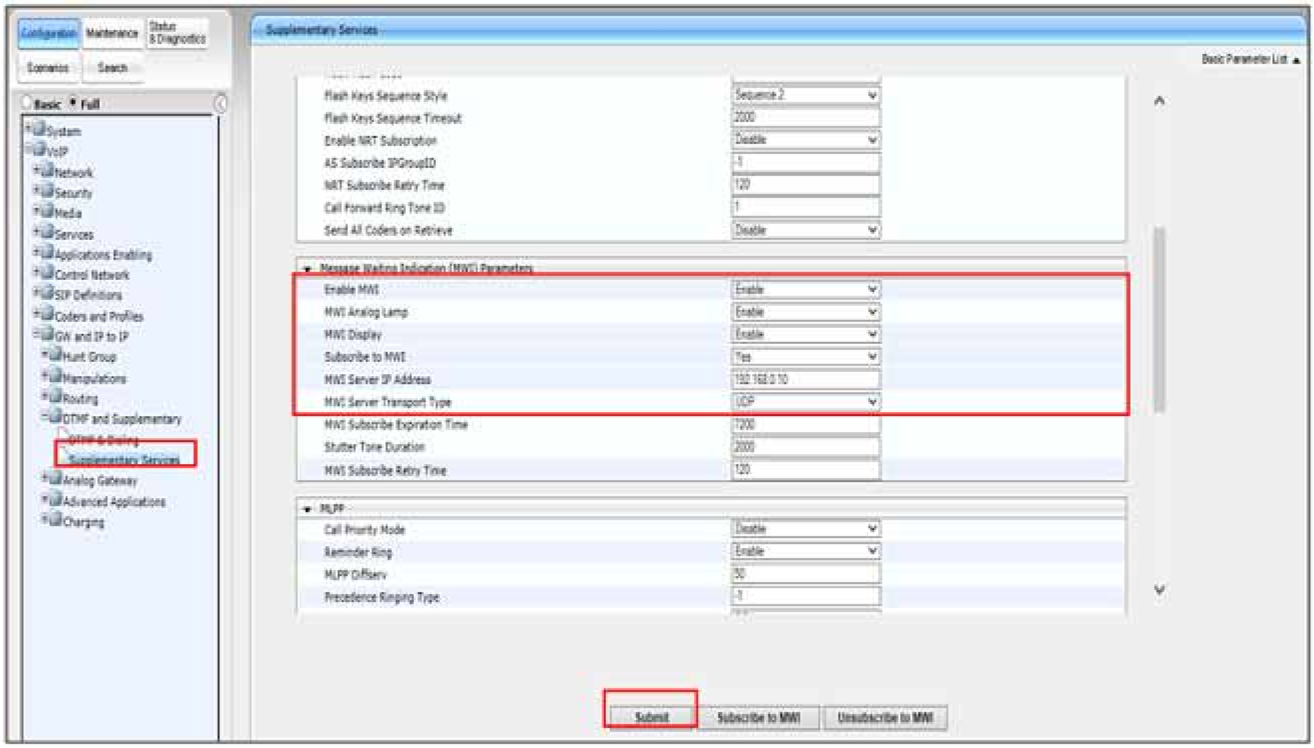
1. Open the **DTMF & Dialing** page (**Configuration- VoIP menu- GW** and **IP to IP- DTMF & Supplementary- DTMF & Dialing**).
2. Set **MAX Digits in Phone Num** to 11.
3. Digit mapping rules can be added to make dialing more efficient.
4. Click **Submit**.

**Figure 2-27 DTMF Supplementary**



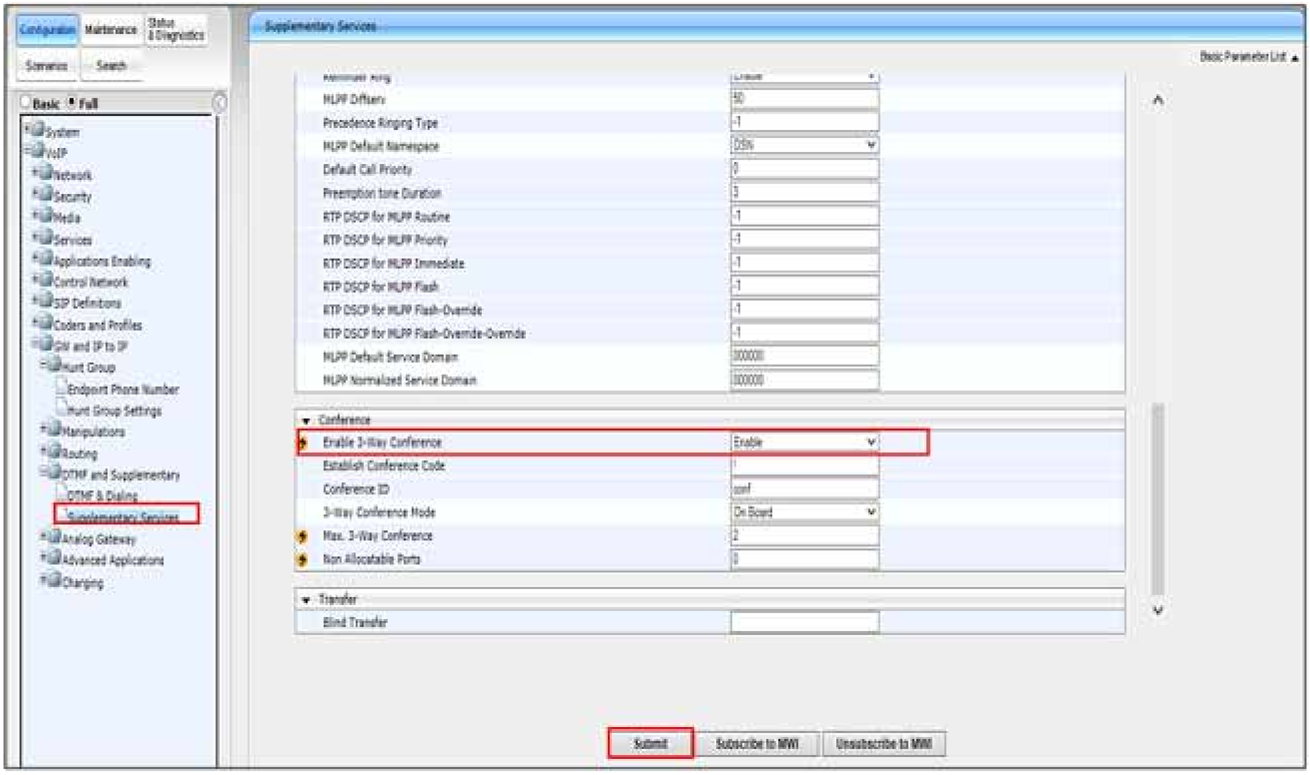
1. Open the **Supplementary Services** page (**Configuration- VoIP menu- GW** and **IP to IP- DTMF & Supplementary- Supplementary Services**).
2. Set **Enable Hold** to **Enable**.
3. Set **Enable Transfer** to **Enable**.
4. Set **Enable Call Forward** to **Enable**.
5. Set **Enable Call Waiting** to **Enable**.
6. Set **Enable Caller ID** to **Enable**.
7. Set **Flash Keys Sequence Style** to **Sequence 2**. (This is required for 3 party conferences).
8. Click **Submit**.
9. Click **Subscribe to MWI** or **Unsubscribe to MWI**.

**Figure 2-28 DTMF Supplementary Continued**



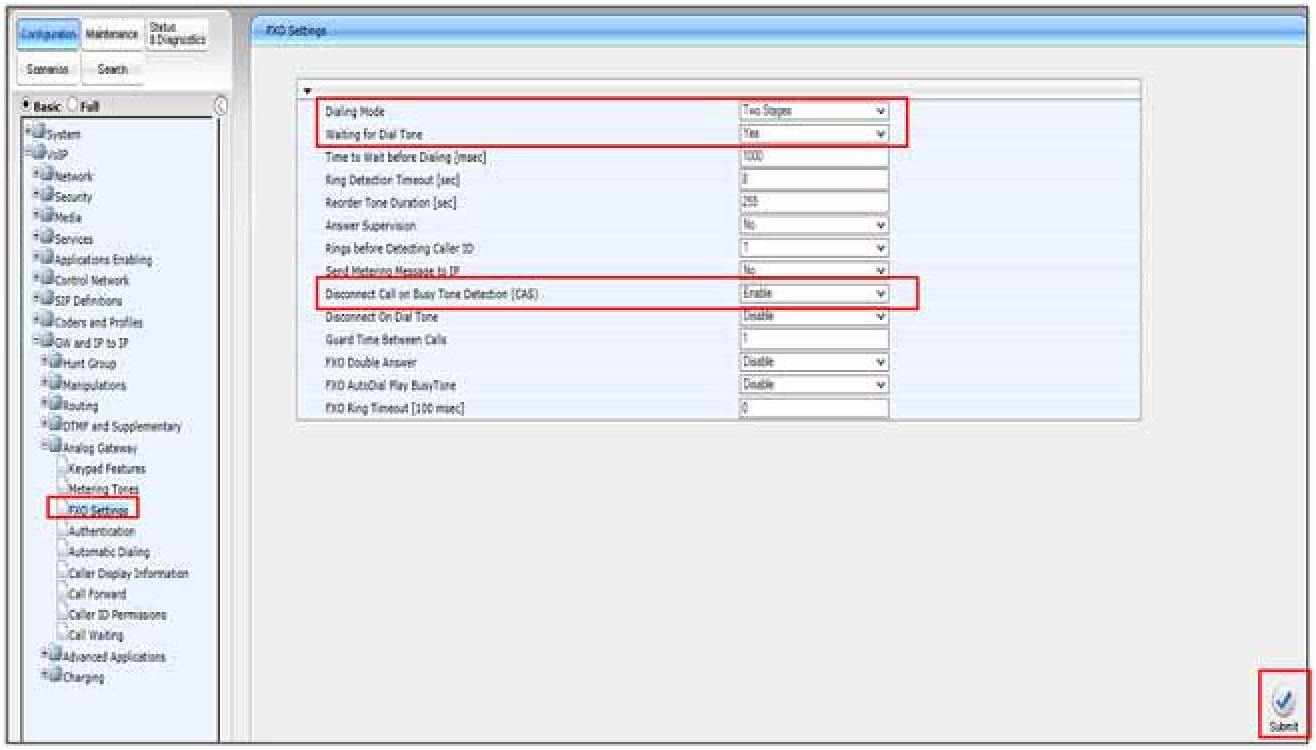

1. Open the **Supplementary Services** page (**Configuration- VoIP menu- GW** and **IP to IP- DTMF & Supplementary- Supplementary Services**).
2. Set **Enable MWI** to **Enable**.
3. Set **MWI Analog Lamp** to **Enable**.
4. Set **MWI Display** to **Enable**. (Terminal must have a display.)
5. Set **Subscribe to MWI** to **Enable**.
6. Set **MWI Server IP address** of MWI server. (SV9300 IP address).
7. Set **MWI Server Transport Type** to **UDP**.
8. Click **Submit**.

**Figure 2-29 DTMF Supplementary Continued**



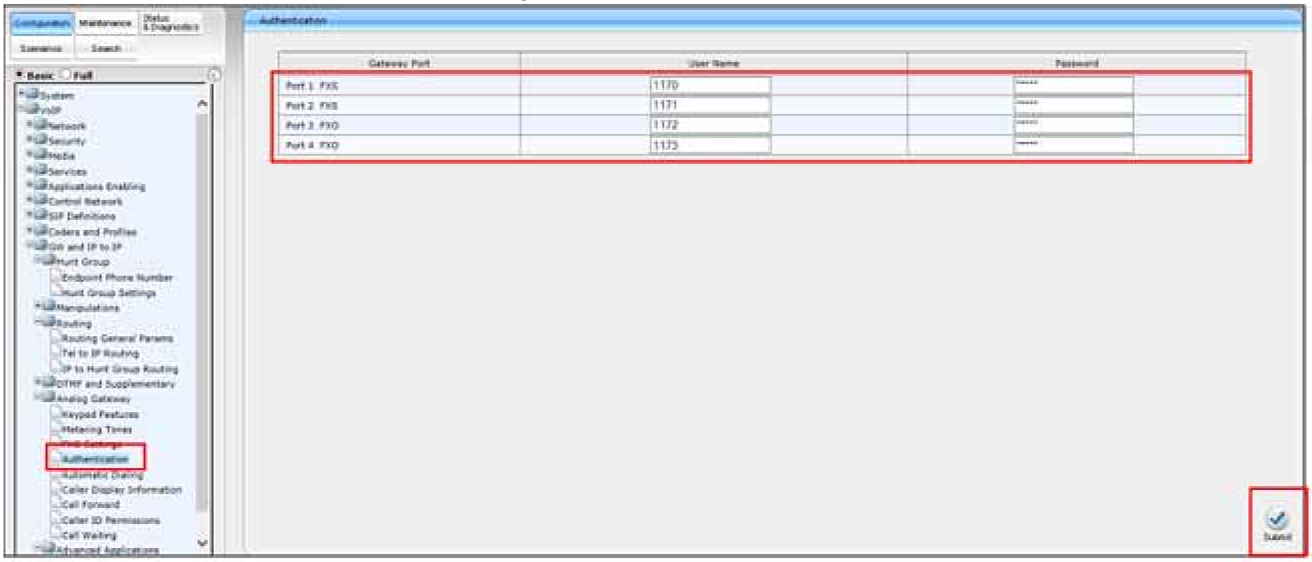
1. Open the **Supplementary Services** page (**Configuration- VoIP menu- GW and IP to IP- DTMF & Supplementary- Supplementary Services**).
2. Set **Enable 3-Way Conference** to **Enable**.
3. Click **Submit**.

**Figure 2-30 FXO Settings**



1. Open the **FXO Settings** page (**Configuration- VoIP- GW** and **IP to IP- Analog gateway- FXO settings**).
2. Set **Dialing Mode** to **Two Stages**.
3. Set **Waiting for Dial Tone** to **Yes**.
4. Set **Disconnect Call on Busy Tone Detection** to **Enable**.
5. Click **Submit**.

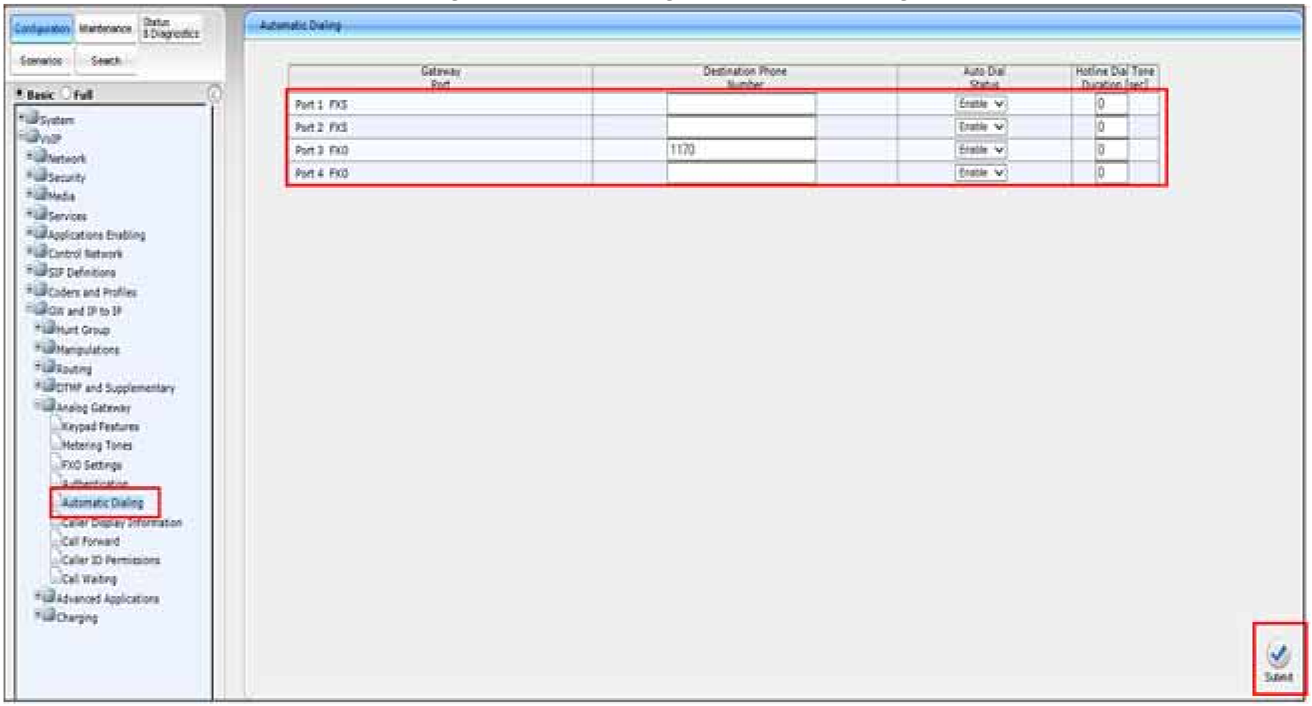
**Figure 2-31 Authentication**




1. Open the **Authentication** page (**Configuration- VoIP- GW** and **IP to IP-Analog Gateway Authentication**).
2. Assign the **User Name** and **Password** to the appropriate ports.
3. Click **Submit**.

✎ *To configure authentication credentials per port: Set the parameter **Registration Mode** (**Authentication Mode**) to **Per Endpoint**. This can be configured in any of the following pages: **Proxy** and **Registration** pages.*

**Figure 2-32 Analog Automatic Dialing**



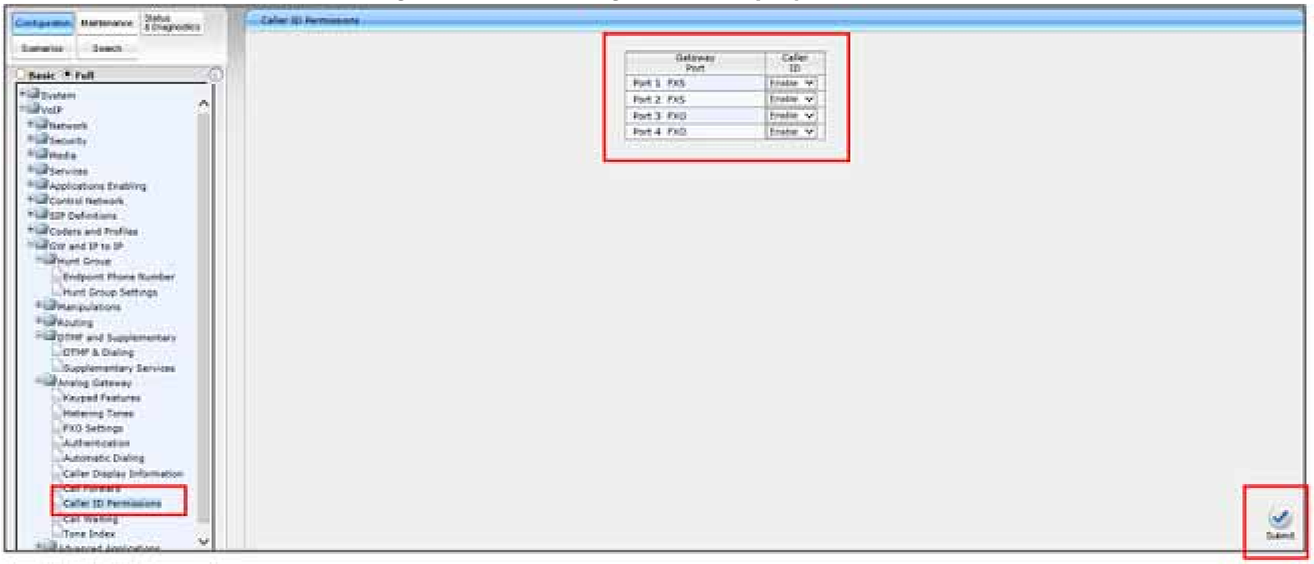
1. Open the **Automatic Dialing** page (**Configuration- VoIP- GW** and **IP to IP- Analog gateway- Automatic Dialing**).

2. Configure Automatic dialing on a per port basis. This is used for direct termination and Hotline assignments.

3. Click **Submit**.

The above configuration sends any incoming call on port 3 (FXO) to 1170.

**Figure 2-33 Analog Caller Display Information**



1. Open the **Caller ID Permissions** page (**VoIP- GW** and **IP to IP- Analog Gateway- Caller ID display**).
2. Select **Enable** to allow Call display information.
3. Click **Submit**.

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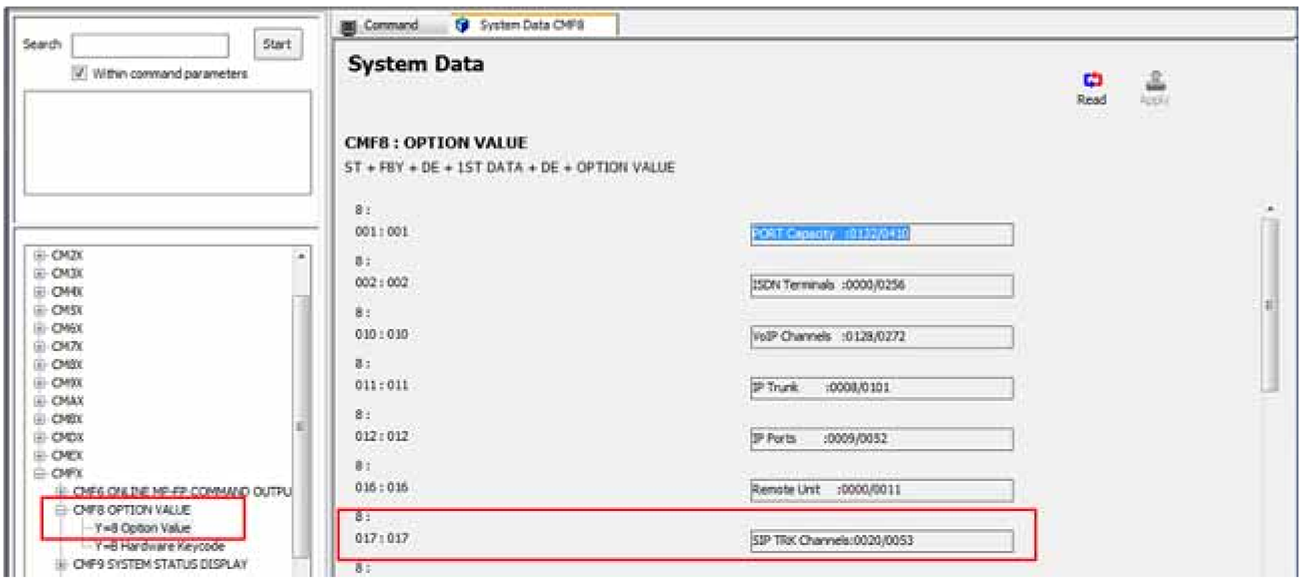
# Chapter 3 *Audio Codes for Trunk Configuration*

## SECTION 1 PROGRAM SV9300 TO USE SIP TRUNKS FOR CONNECTIVITY TO AUDIOCODES

The SV9300 will need to be programmed to use SIP trunks for connectivity to the AudioCodes.

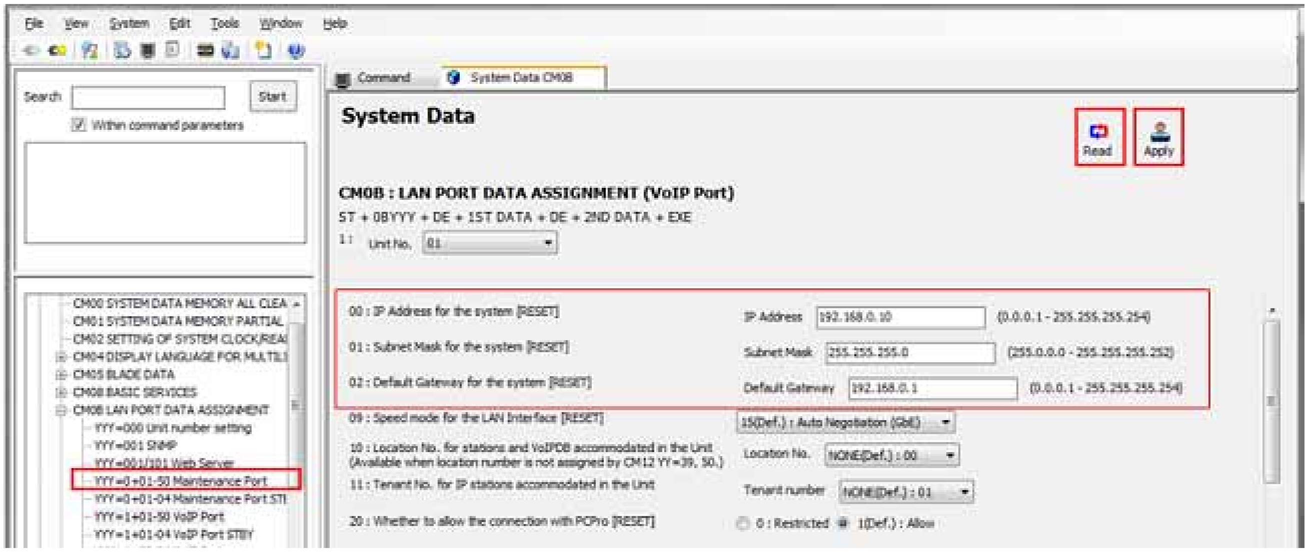
**Figure 3-1 CM F88 SIP Trunk Port License Capacity**




1. Confirm SIP trunk license are available. **Read** is automatic for this command. Example: **CMF8 Y=8>017>0020/0053**

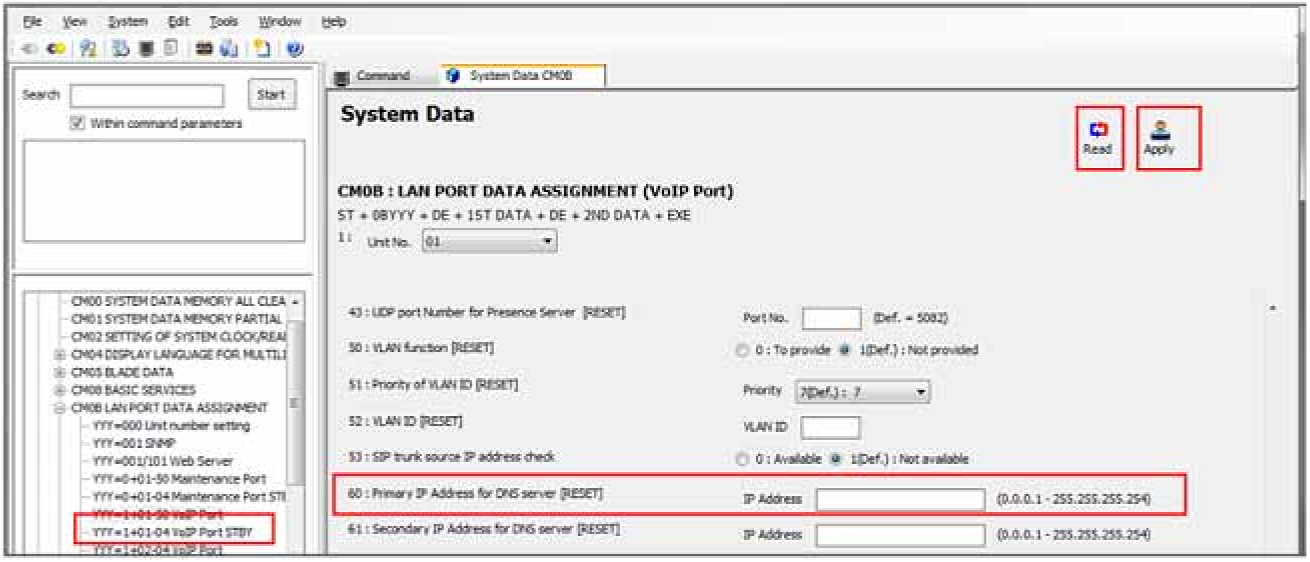
2. This is a read only command.

**Figure 3-2 CM 0B1XX Network Configuration**




1. Select **SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels**, then click **Read** to get the current data settings.
2. Assign the SV9300 **VoIPDB IP Address**.
   Example: **0B101>00>192.168.0.10**
3. Assign the SV9300 **VoIPDB Subnet Mask**.
   Example: **0B101>01>255.255.255.0**
4. Assign the SV9300 **VoIPDB Default Gateway Address**.
   Example: **0B101>02>192.168.0.1**
5. Click **Apply**.

**Figure 3-3 CM 0B1XX DNS Address (If Required)**

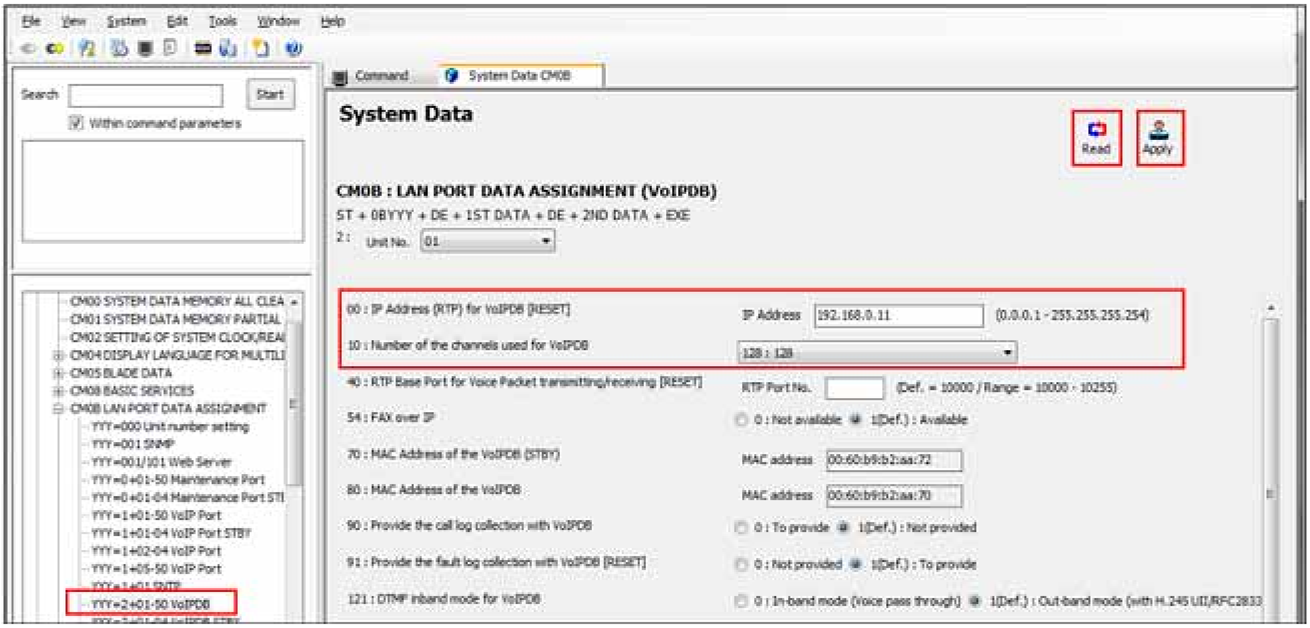



1. Select **SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels**, then click **Read** to get the current data settings.

2. Assign the primary DNS server IP Address (optional).
Example: **0B101>60>IP address of DNS server**

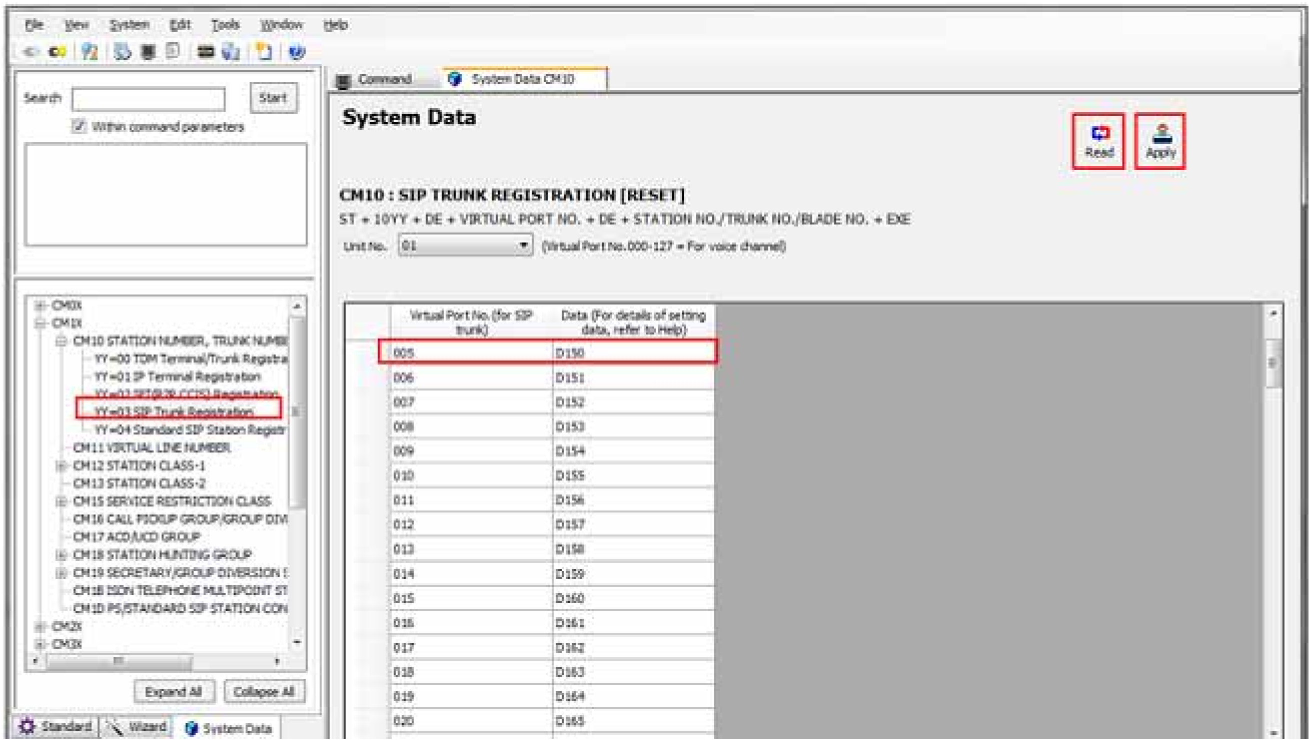
3. Click **Apply**.

**Figure 3-4 CM 0B2xx IP PAD Network Settings**



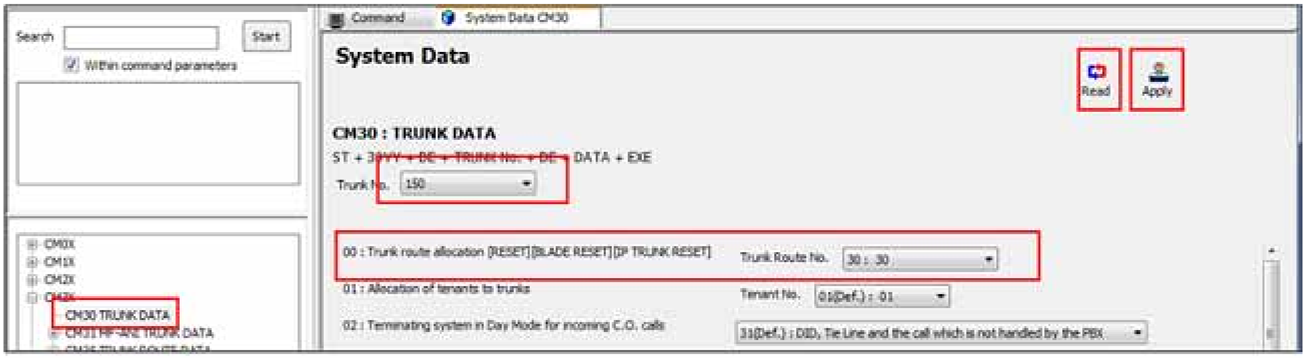

1. Select **SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels**, then click **Read** to get the current data settings.
2. Enter VoIP IPPAD IP address.
   Example: **0B201>00>192.168.0.11**
3. Assign the number of VoIP IPPAD channels.
   Example: **0B201>10>128**
4. Click **Apply**.

**Figure 3-5 CM 1003 SIP Trunk Port Allocation**



1. Select **SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels**, then click **Read** to get the current data settings.
2. Enter Trunk Numbers used for the voice channels.
   Example: **1003>005>D150**, repeat until all trunks are assigned.
3. Click **Apply**.

**Figure 3-6 CM 30XX SIP Trunk Port Settings**

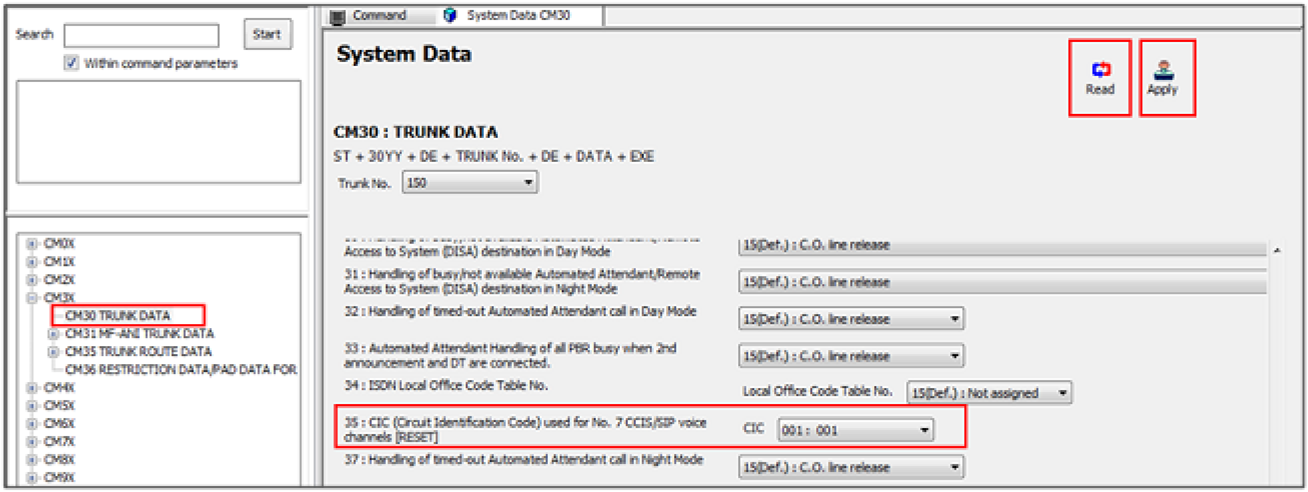



1. Select each **Voice Channel** and click the **Read** button to get the current data settings.
   Example: **CM3000>150>30**

2. Assign the same Trunk Route Number to each voice channel.

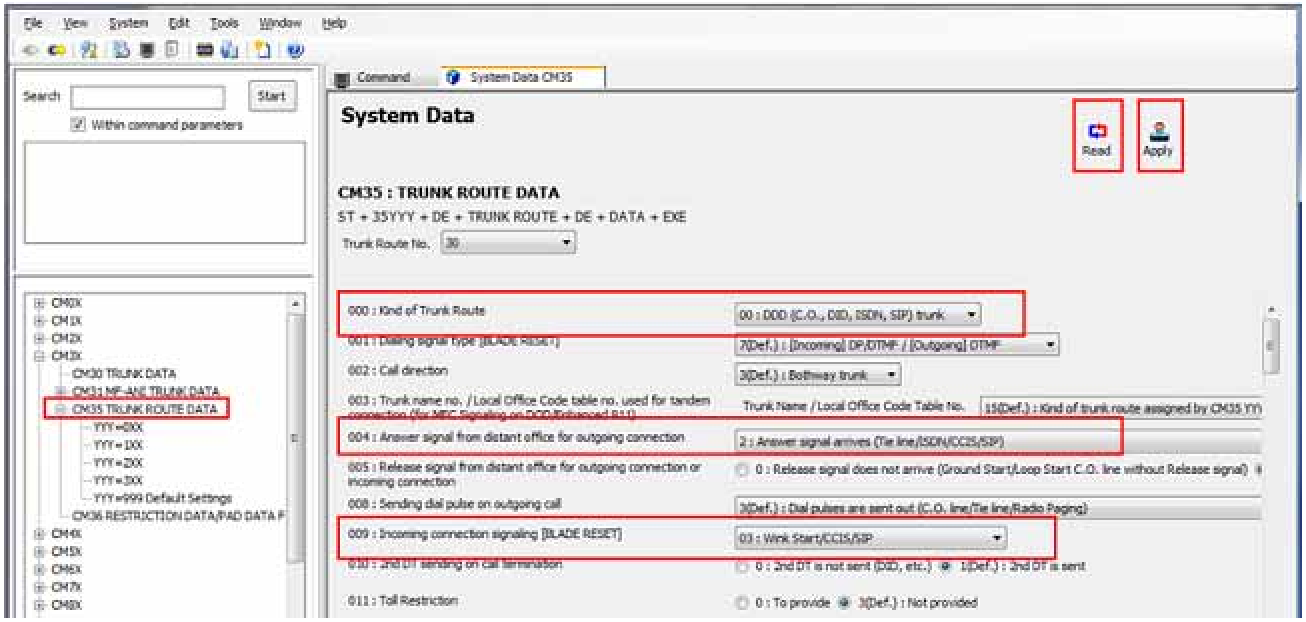
3. Click **Apply**.

**Figure 3-7 CM 30XX SIP Trunk Port Settings Continued**




1. Select each **Voice Channel** and click the **Read** button to get the current data settings.
2. Assign a CIC number to each voice trunk.
   Example: **CM3035>150>001**
3. Click **Apply**.

**Figure 3-8 CM 35XX SIP Trunk Route Settings**




1. Select the **Trunk Route Number** assigned to voice channels and click the **Read** button to get current data settings.

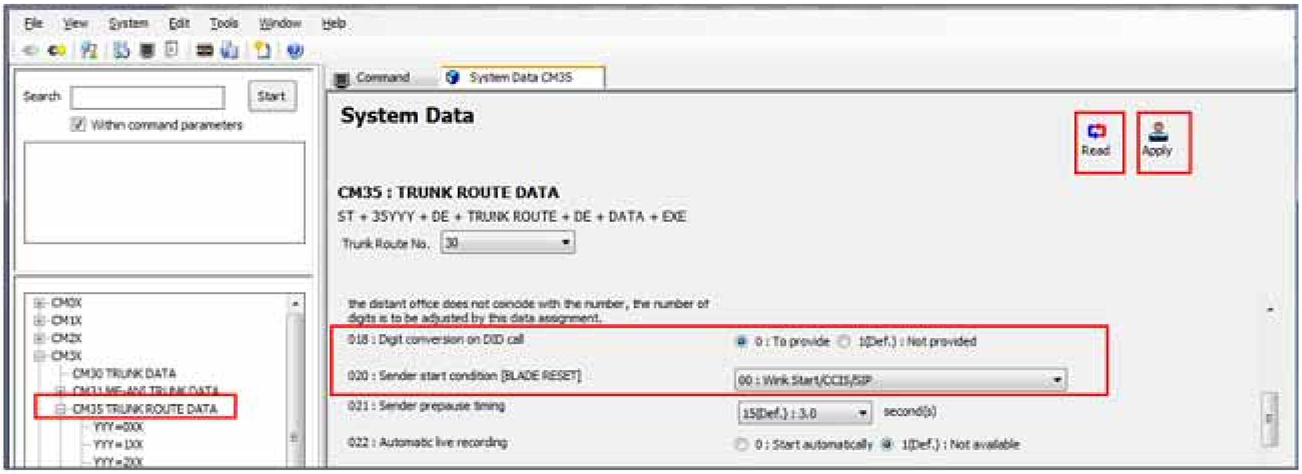
2. Assign **00** for SIP trunk service.
   Example: **CM3500>30>00**

3. Assign **2** for SIP trunk service.
   Example: **CM3504>30>2**

4. Assign **03** for SIP trunk service.
   Example: **CM3509>30>03**

5. Click **Apply**.

**Figure 3-9 CM 35XX SIP Trunk Route Settings Continued**



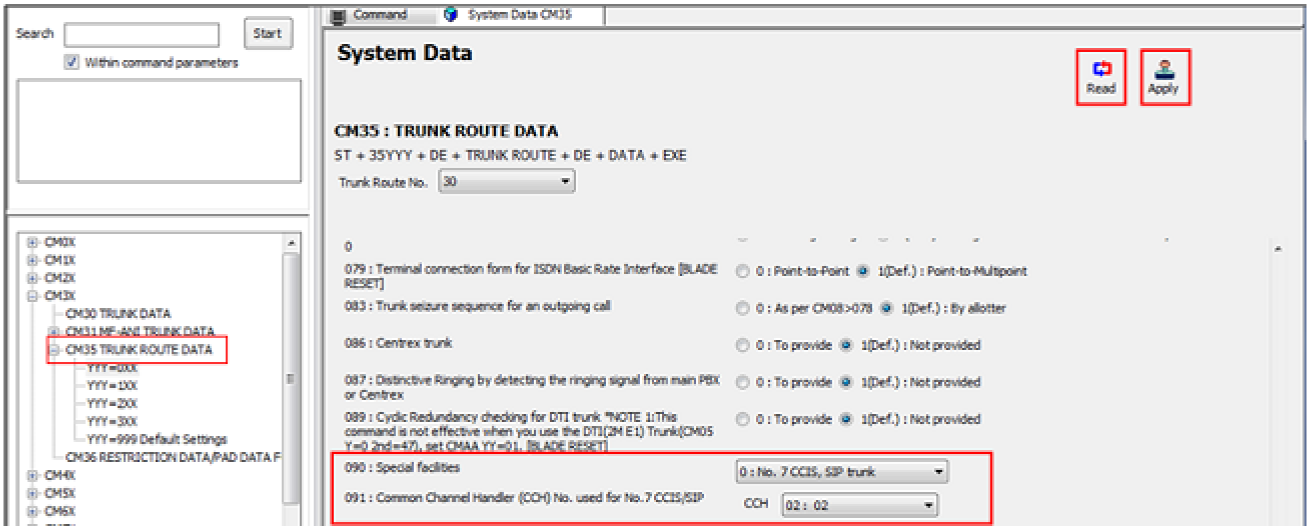

1. Select the **Trunk Route Number** assigned to voice channels and click the **Read** button to get current data settings.

2. Assign **0** for DID digit conversion.
Example: **CM3518>30>0**

3. Assign **00** for SIP trunk service.
Example: **CM3520>30>00**

4. Click **Apply**.

**Figure 3-10 CM 35XX SIP Trunk Route Settings Continued**




1. Select the **Trunk Route Number** assigned to voice channels and click the **Read** button to get current data settings.

2. Assign **0** for SIP trunk service.
   Example: **CM3590>30>0**

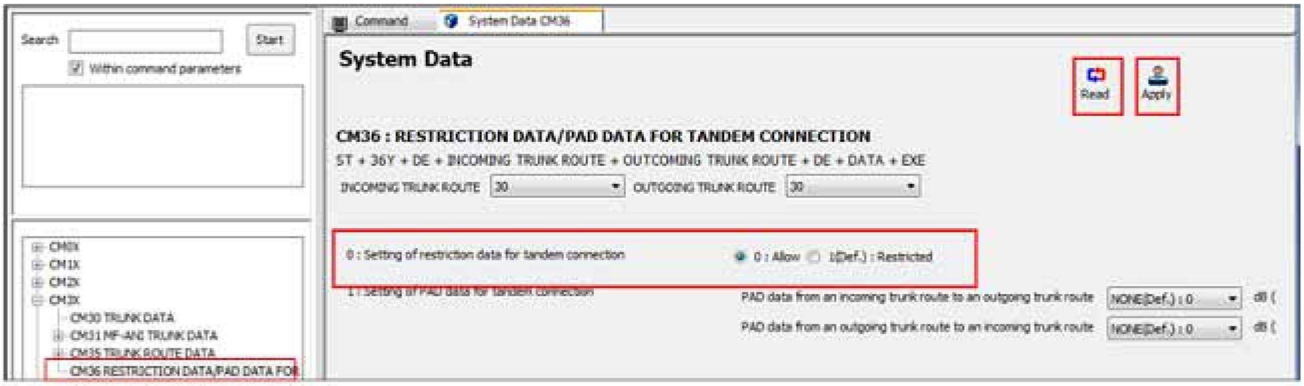
3. Assign **CCH** used for SIP trunk.
   Example: **CM3591>30>02**

4. Click **Apply**.

   ❑ CCH 00 should not be assigned for SIP trunk; P-P CCIS must use CCH 00.

   ❑ Assign a different CCH to each SIP trunk server voice route.

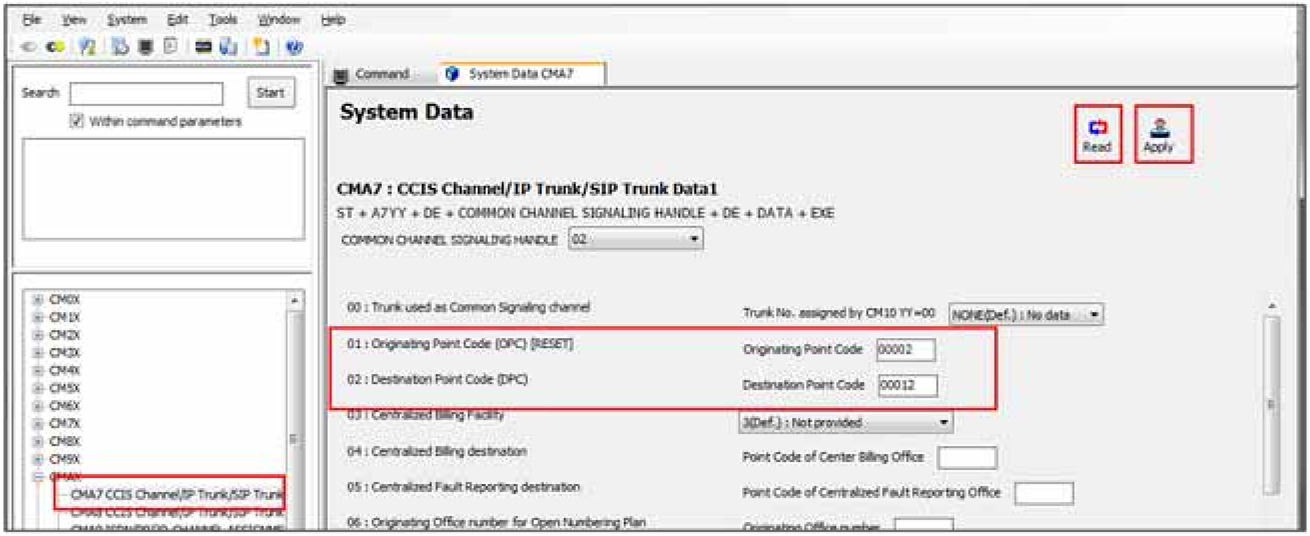
**Figure 3-11 CM 36 Route to Route Connection Settings**




1. Select **Incoming Trunk Route**.
2. Assign **Outgoing Trunk Route** and click the **Read** button to get current data settings.
3. Select **0** to allow route to route connection.
   Example: **CM360>3030>0**
4. Click **Apply**

   Allow route to route connection (i.e. SIP route to PRI route, PRI route to SIP route, and SIP route to SIP route).

**Figure 3-12 CM A7 SIP Trunk Control Channel Settings**




1. Select **CCH assigned to SIP trunk** and click the **Read** button to get current data settings.

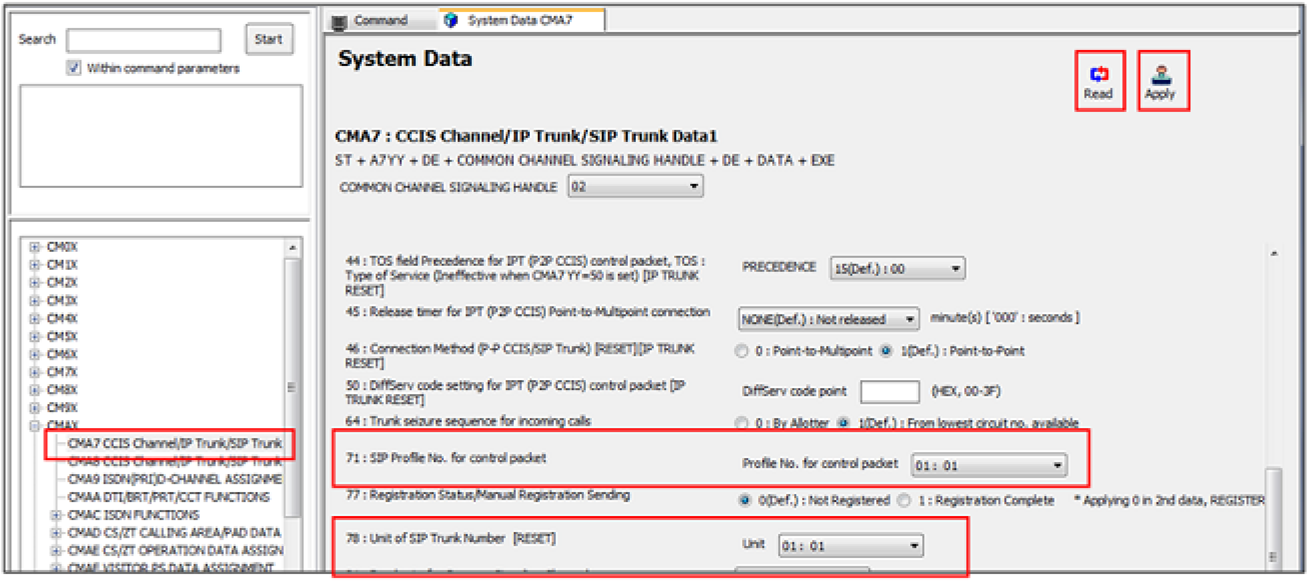
2. Assign an arbitrary Originating Point Code.
   Example: **CMA701>02>00002**
   The same originating point code can be used for each SIP trunk server.

3. YY=02 Assign an arbitrary Destination Point Code.
   Example: **CMA702>02>00012**

   Different destination point code must be used for each SIP trunk server.

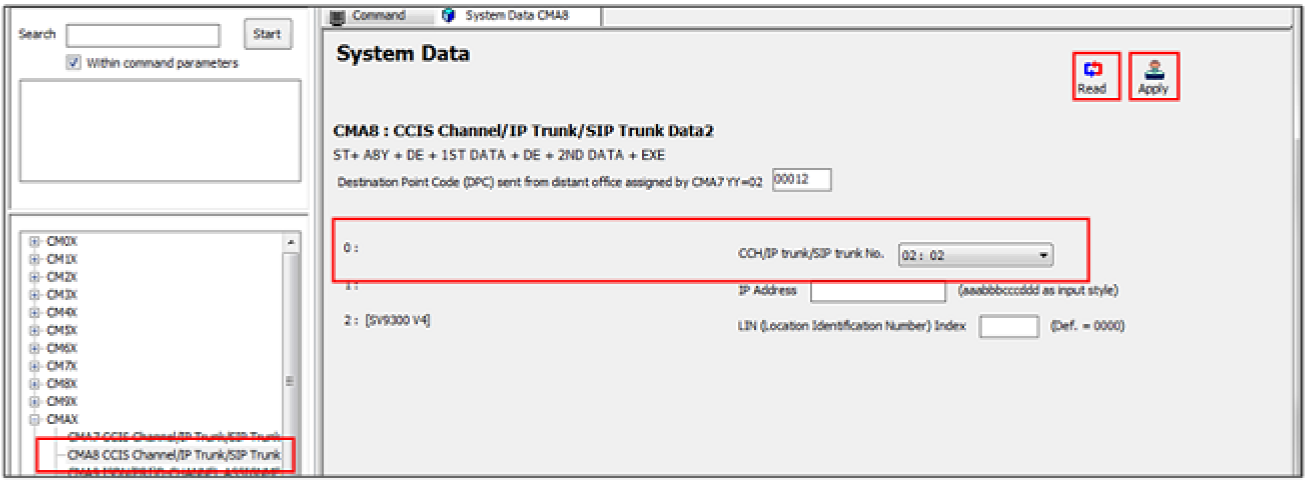
4. Click **Apply**.

**Figure 3-13 CM A7 SIP Trunk Control Channel Settings Continued**



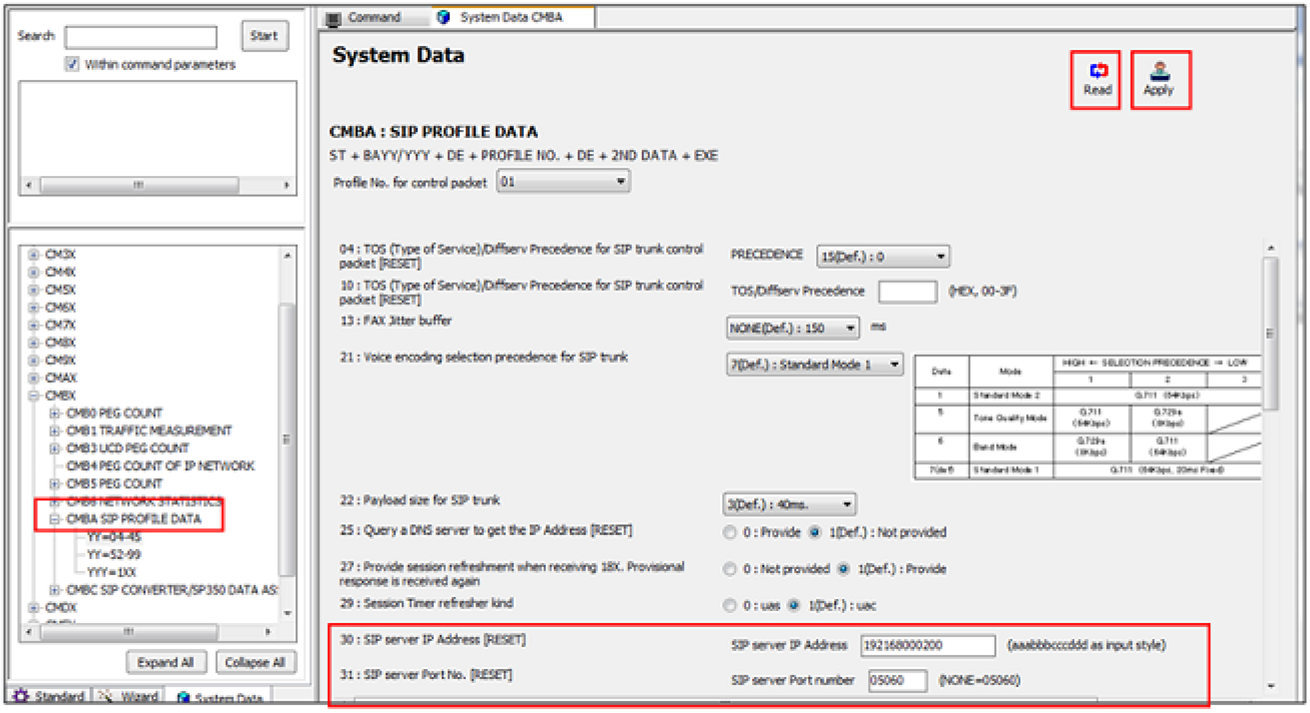
1. Select **CCH assigned to SIP trunk** and click the **Read** button to get current data settings.
2. Assign an unused SIP Trunk Profile Number.
   Example: **CMA771>02>01**
3. Assign Unit No. accommodating SIP trunk channels.
   Example: **CMA778>02>0**
4. Click **Apply**.

**Figure 3-14 CM A8 SIP Trunk Destination Point Code Settings**




1. Select **Destination Point Code** assigned by CMA7 YY=02 and click the **Read** button to get current data settings.
2. Assign CCH assigned to destination point code in CMA7 YY=02.
   Example: **A80>00012>02**
3. Click **Apply**.

**Figure 3-15 CM BA SIP Trunk Profile Settings**




1. Select **SIP Trunk Profile Number** assigned by CMA771 and click the **Read** button to get current data settings.

2. Assign AudioCodes IP address when not providing DNS server query. Example: **BA30>01>1921658000200**

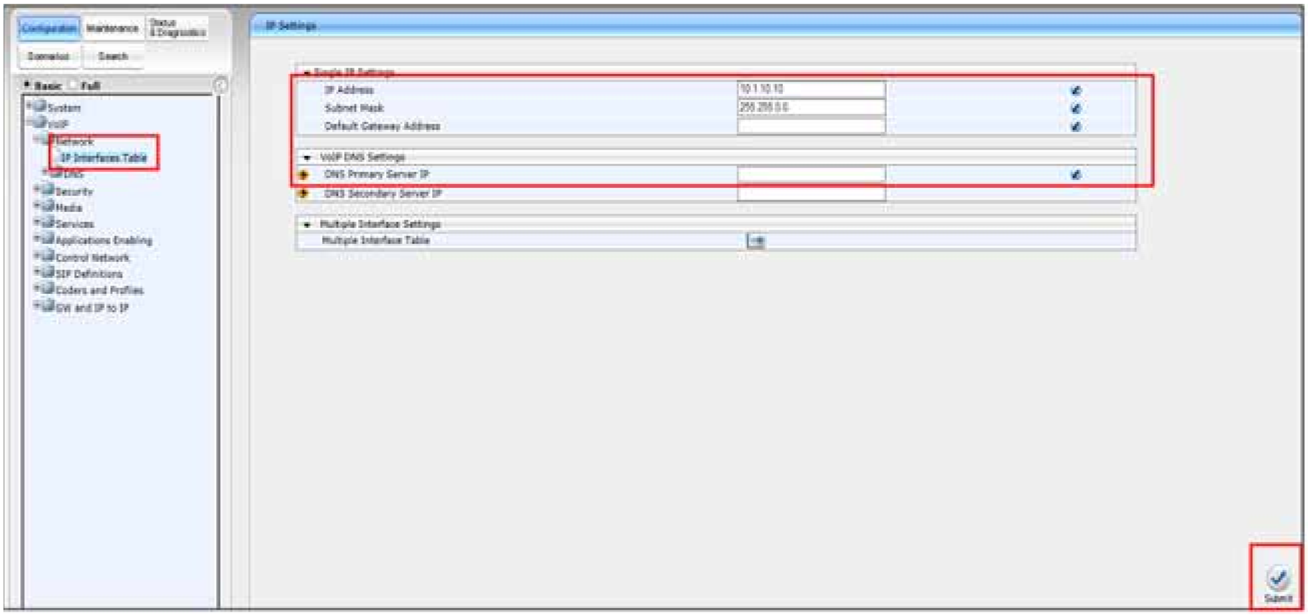
3. Assign SIP trunk server port. Assign port 05060 to use port 5. Example: **BA32>01>05060**

4. Click **Apply**.

✎ *This set up is with non-Registered trunks. Refer to the SV9300 Programming Manual for LCR programming.*

## SECTION 2 AUDIO CODES TRUNK CONFIGURATION

**Figure 3-16 IP Settings (Single Network Interface)**

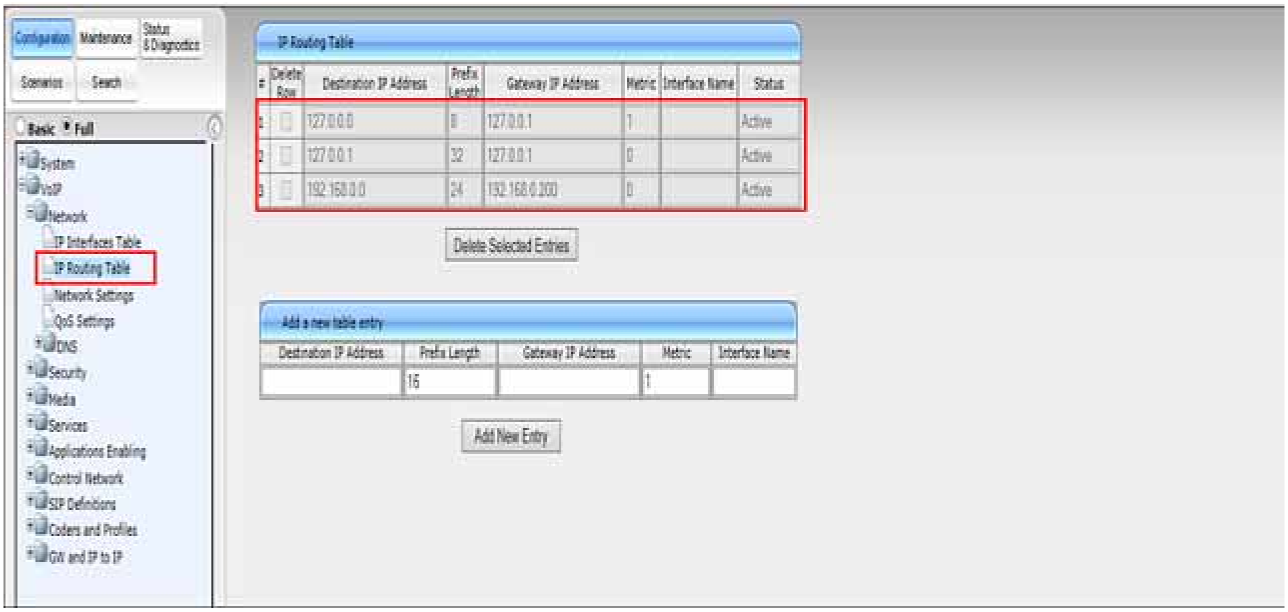


5. Open the **IP Settings** page (**Configuration tab-VoIP menu-Network-IP settings**).
   The initial IP address is 10.1.10.10.

6. Enter the **IP Address**, **Subnet Mask**, **Gateway** and **DNS Address** (if required).

7. Click **Submit**.

**Figure 3-17 IP Routing Table**




1. Open the **IP Routing Table** page (**Configuration- VoIP- Network- IP Routing table**).

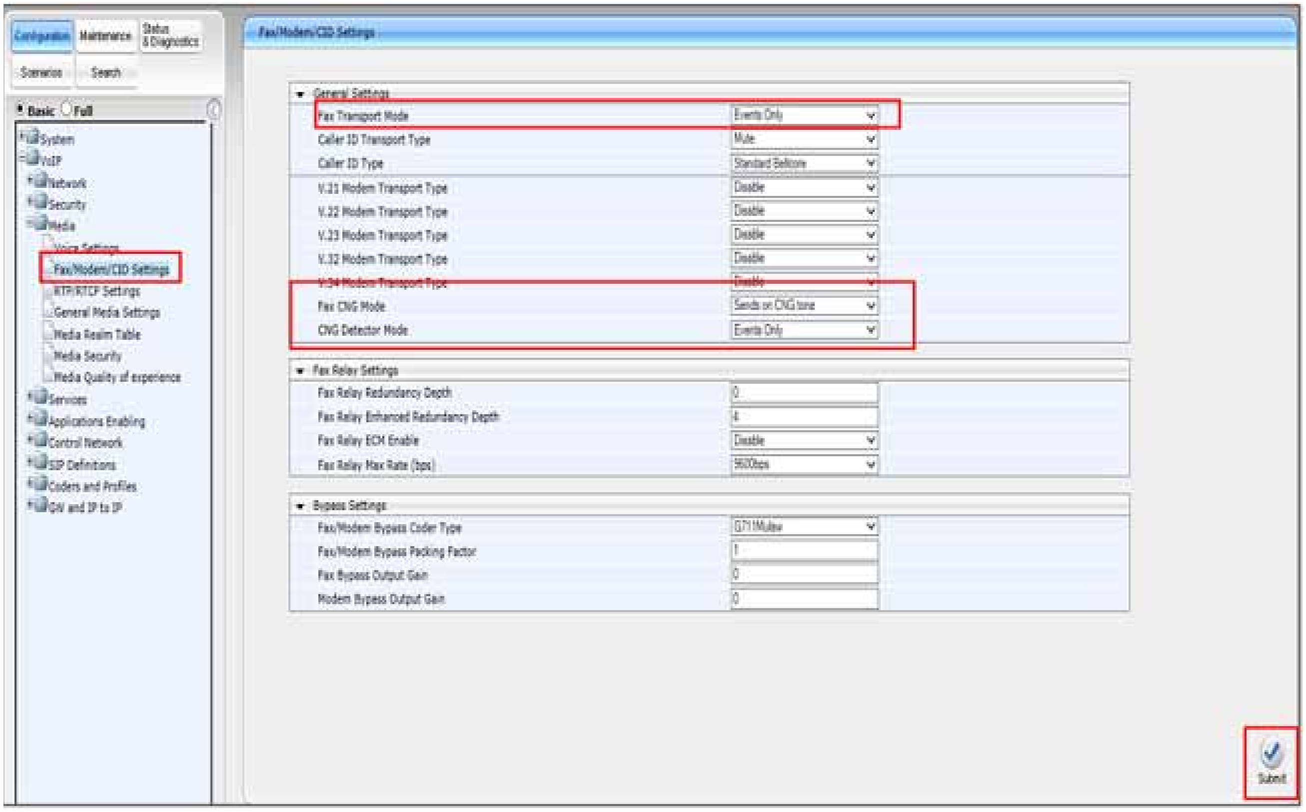
2. Click **Add New Entry** and enter a new static route.

3. Click **Add New Entry** and the new routing rule is added to the IP routing table.

4. To delete a routing rule, select the **Delete Row** check box and then click **Delete**.

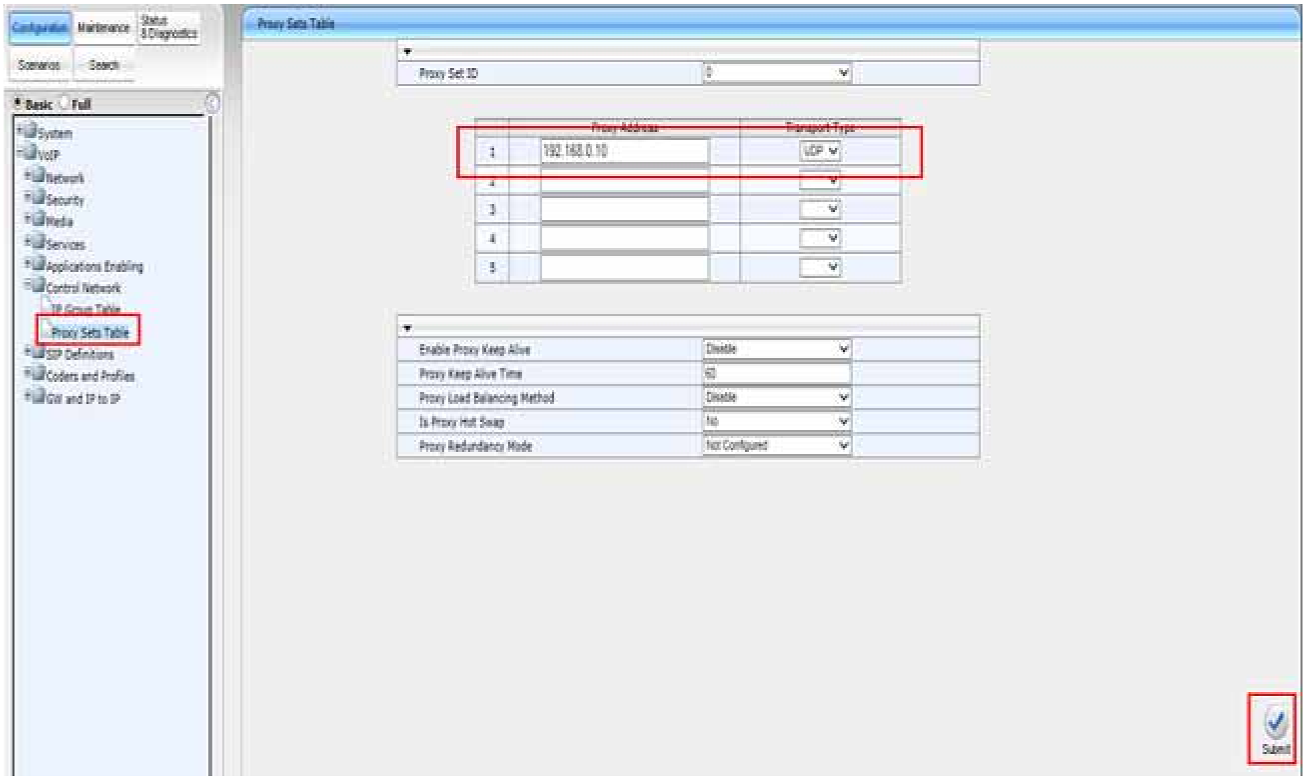
   Refer to the User's Manual for additional information.

**Figure 3-18 Fax Settings**



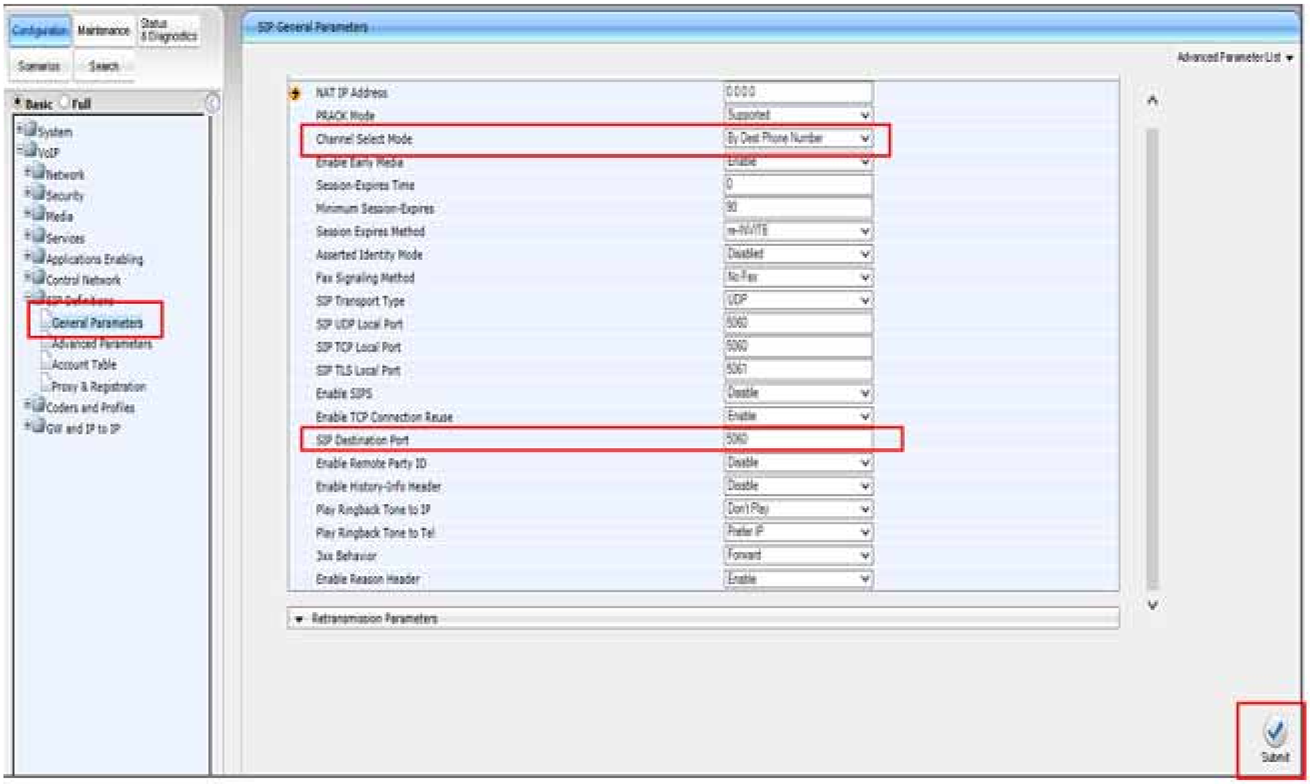
1. Open the **Fax/Modem/CID Settings** page (**Configuration tab-VoIP menu-Media- Fax/Modem/CID settings**).
2. Assign **Fax Transport Mode** to **Events Only**.
3. Assign **Fax CNG Mode** to **Sends on CNG tone**.
4. Assign **CNG Detector Mode** to **Events Only**.
5. Click **Submit**.

**Figure 3-19 Proxy Sets Table**



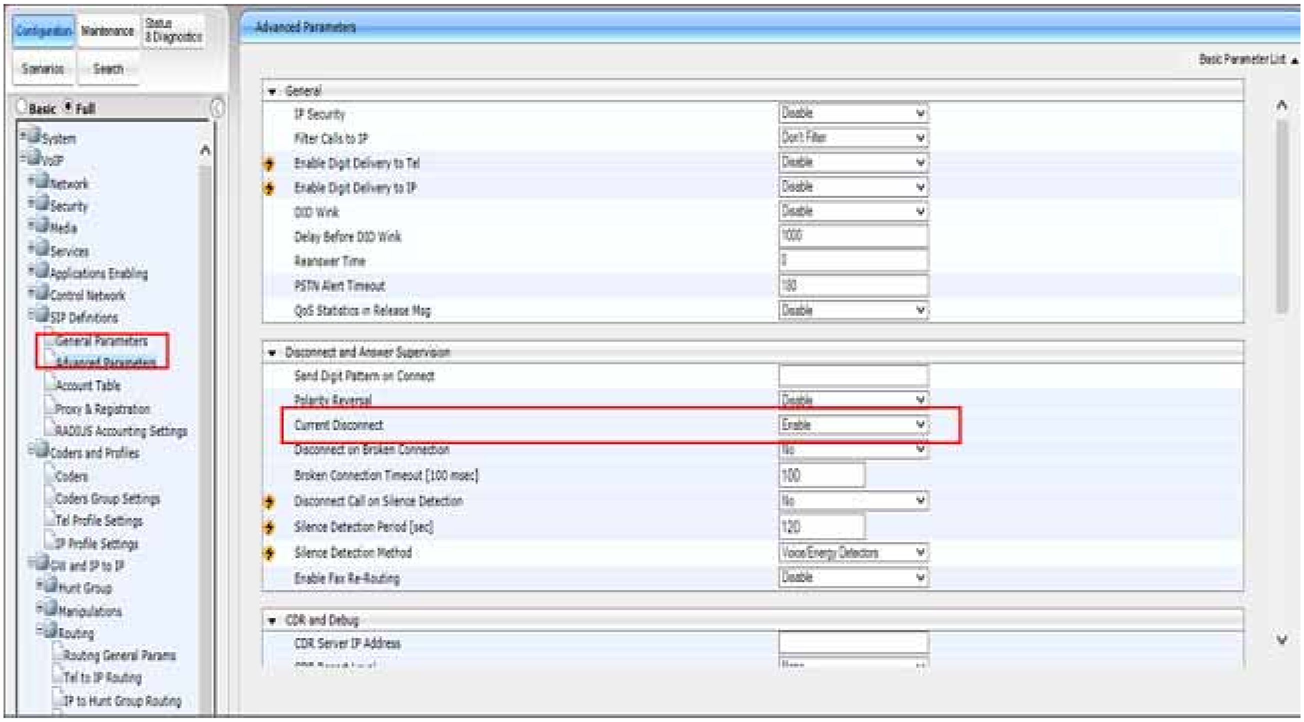
1. Open the **Proxy Sets** table page (**Configuration-VoIP Menu-Control Network-Proxy Sets Table**).
2. Select a **Proxy Set ID** from the drop-down list.
3. Configure the Proxy as required (IP address of the SV9300).
4. Leave all other settings at default.
5. 5. Click **Submit**.

**Figure 3-20 SIP Definitions General Parameters**




1. Open **SIP Definitions General Parameters** (**Configuration tab-VoIP menu-SIP Definitions-General Parameters**).
2. Set the **Channel Select Mode** to Destination Phone number.
3. Set the **SIP Destination Port** to **5060**.
4. Click **Submit**.

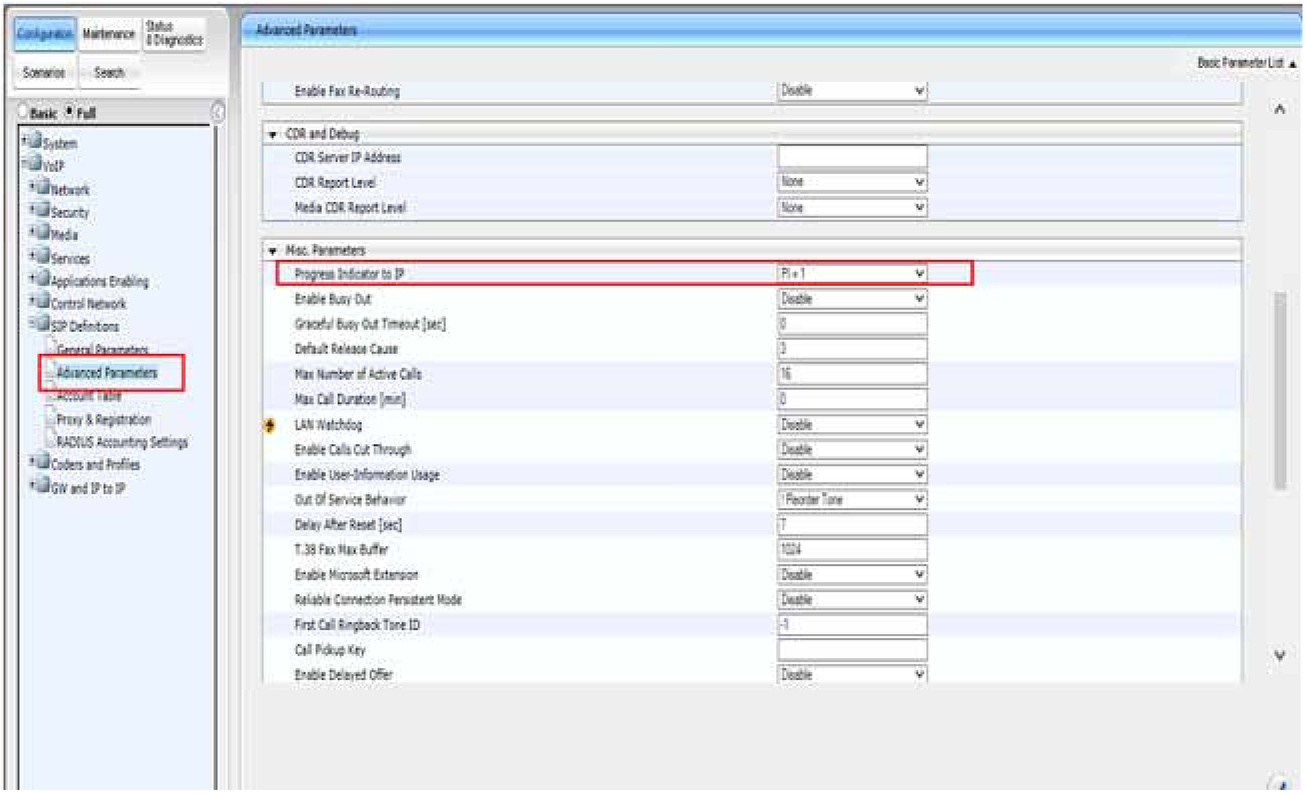
**Figure 3-21 SIP Definitions Advanced Parameters**



1. Open the **Advanced Parameters** page (**Configuration- VoIP- SIP Definitions- Advanced Parameters**).

2. Confirm that **Current Disconnect** is set to **Enable**.

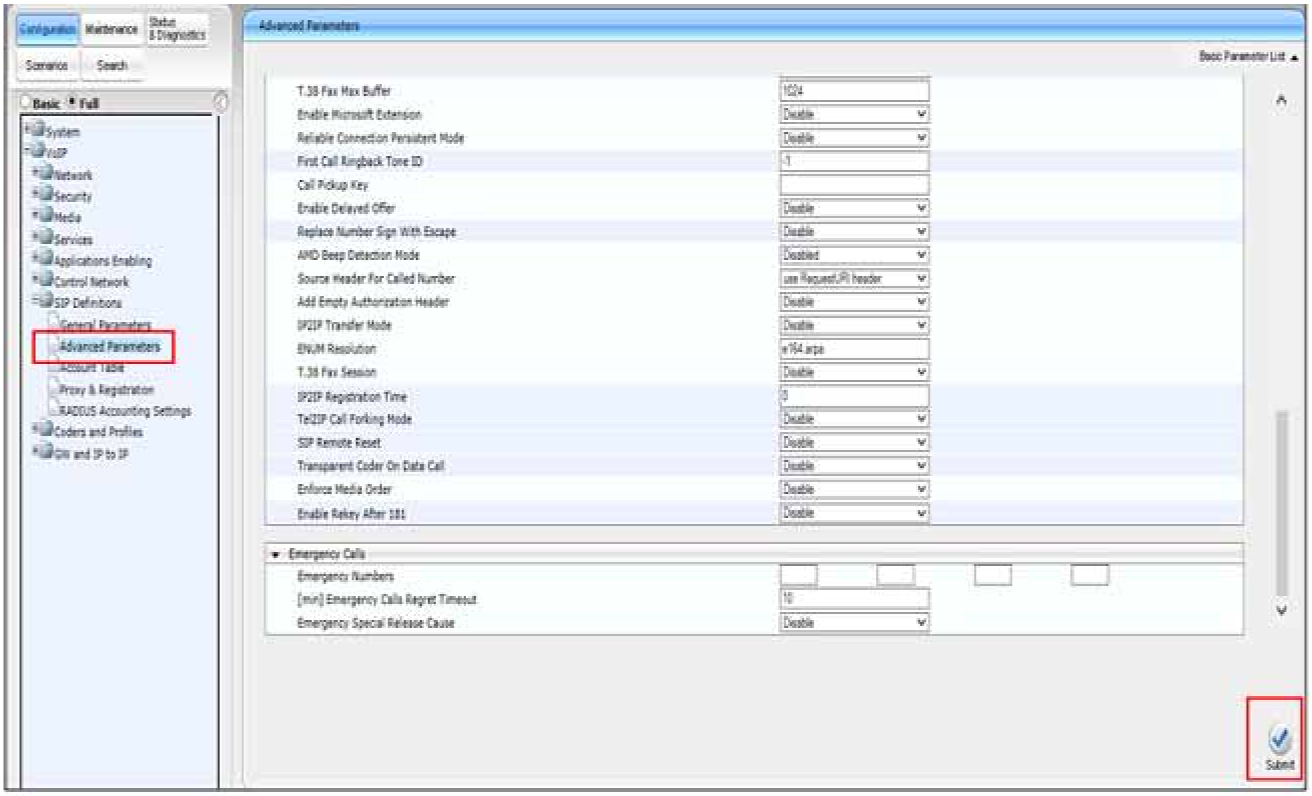
**Figure 3-22 SIP Definitions Advanced Parameters Continued**



1. Open the **Advanced Parameters** page (**Configuration- VoIP- SIP Definitions- Advanced Parameters**).

2. Confirm that **Progress Indicator to IP** is set to **PI=1**.

**Figure 3-23 SIP Definitions Advanced Parameters Continued**

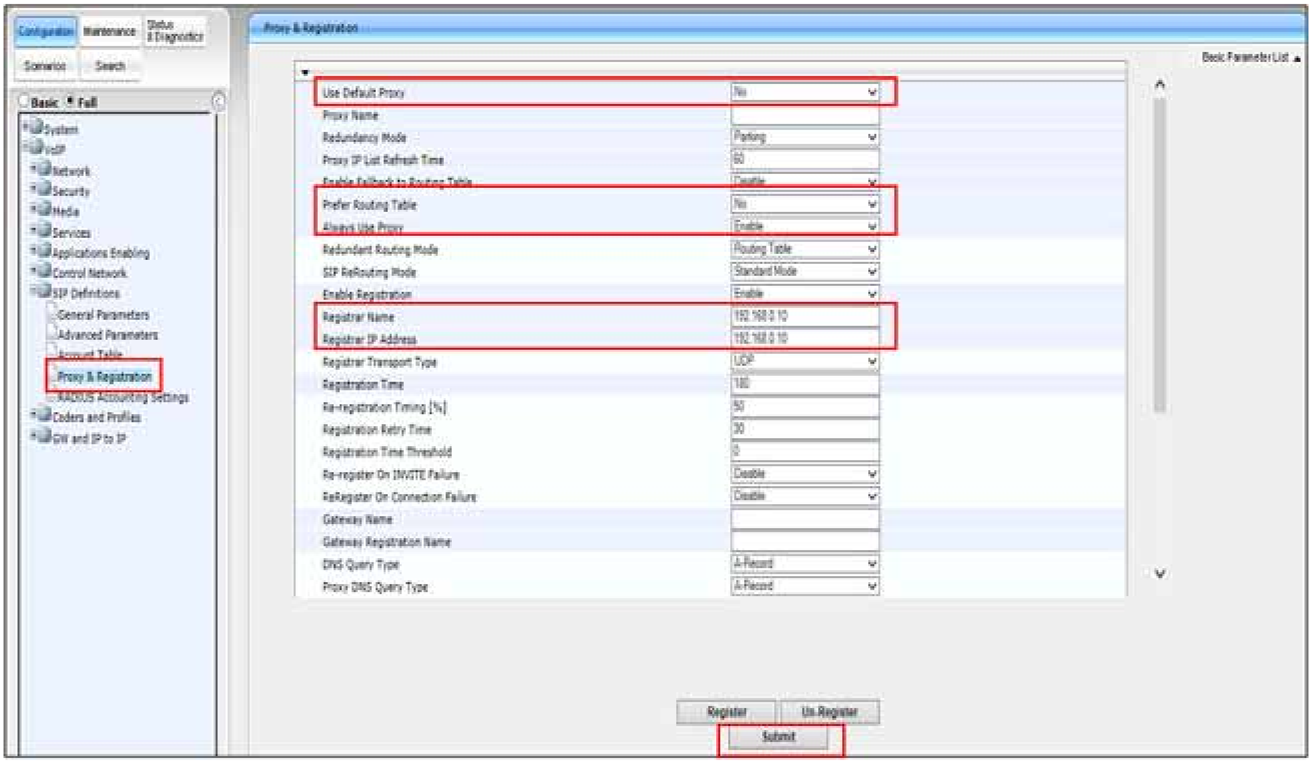



1. Open the **Advanced Parameters** page (**Configuration- VoIP- SIP Definitions- Advanced Parameters**).

   There should be no changes to this page

2. Click **Submit**.

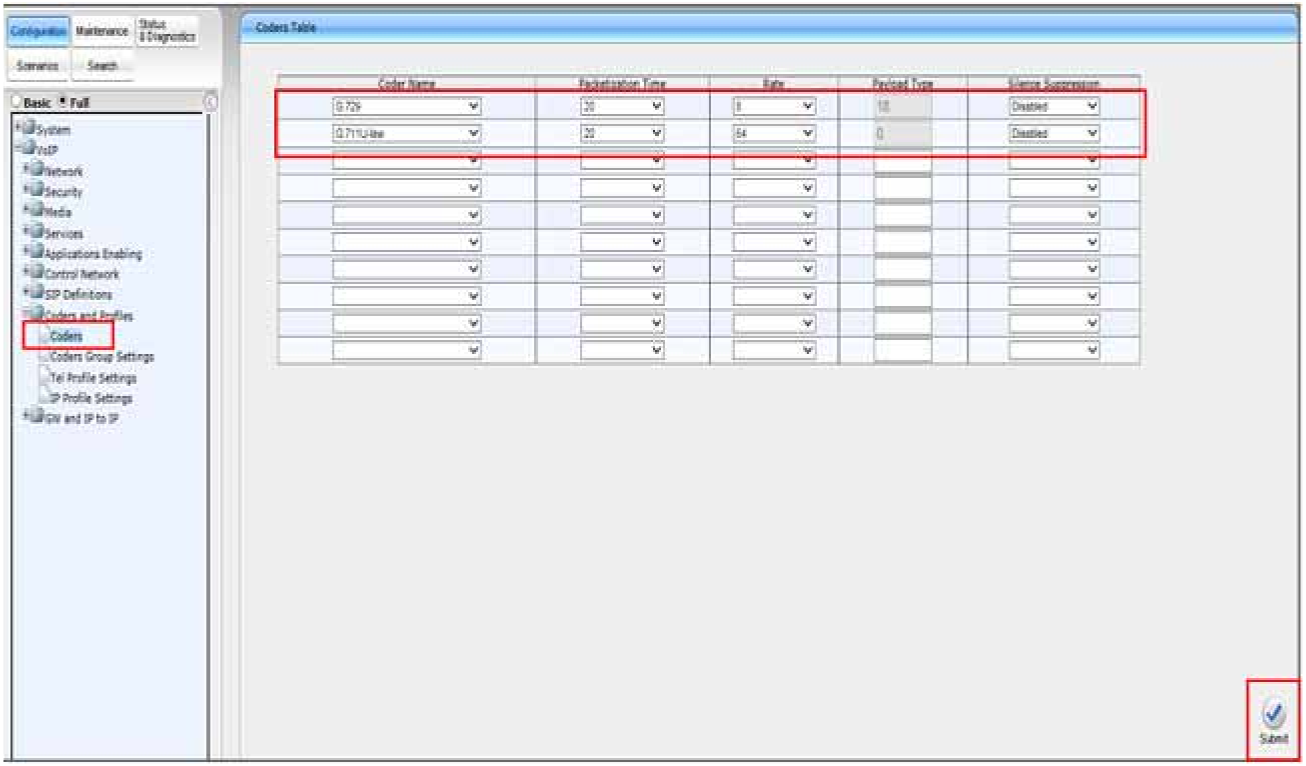
**Figure 3-24 SIP Definitions Proxy and Registration**




1. Open the **Proxy & Registration** page (**Configuration tab-VoIP menu-SIP Definitions-Proxy and Registration**).
2. Set **Use Default Proxy** to **No**.
3. Set **Prefer Routing Table** to **No**.
4. Set **Always Use Proxy** to **Enable**.
5. Set **Registrar Name** to the SV9300 IP address. (EX: 192.168.0.10)
6. Set **Registrar IP Address** to the SV9300 IP address (EX: 192.168.0.10)
7. Click **Submit**.

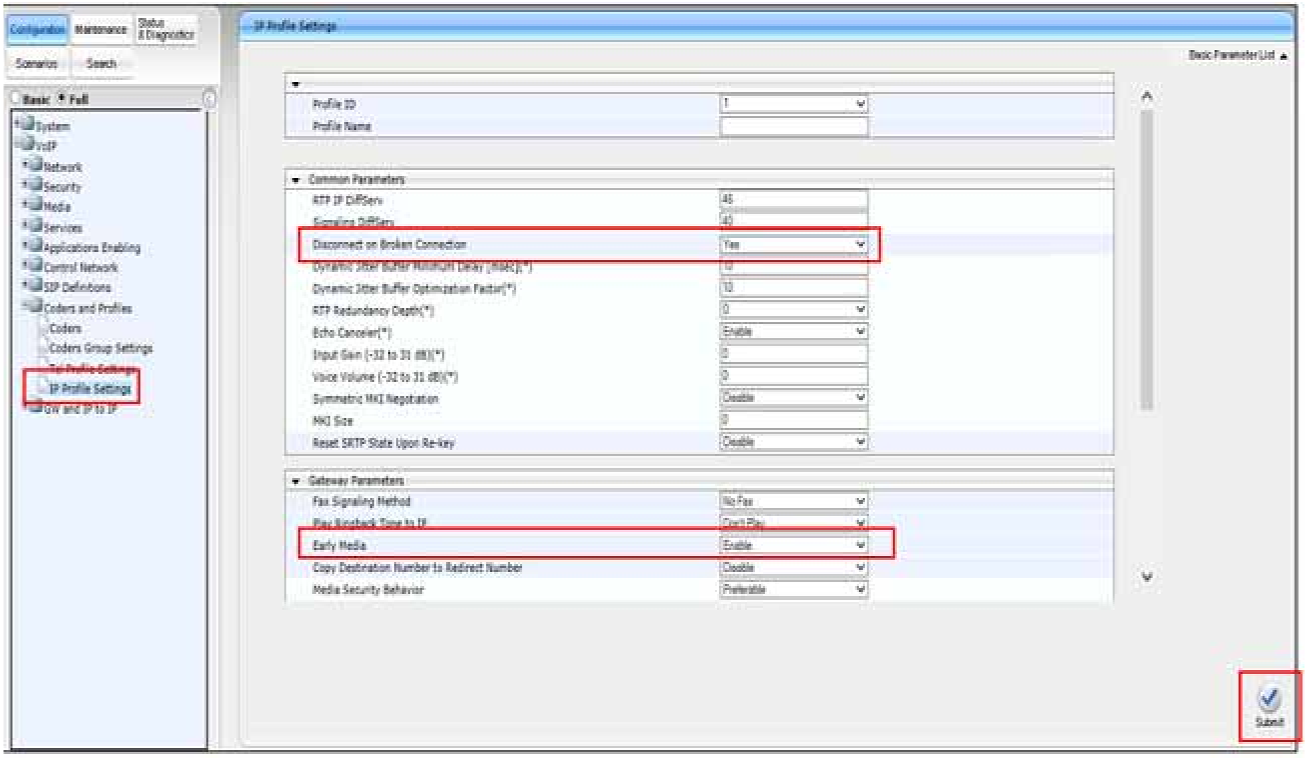
✎ *This is a non registered SIP trunk. There is no need to register.*

**Figure 3-25 Coders**



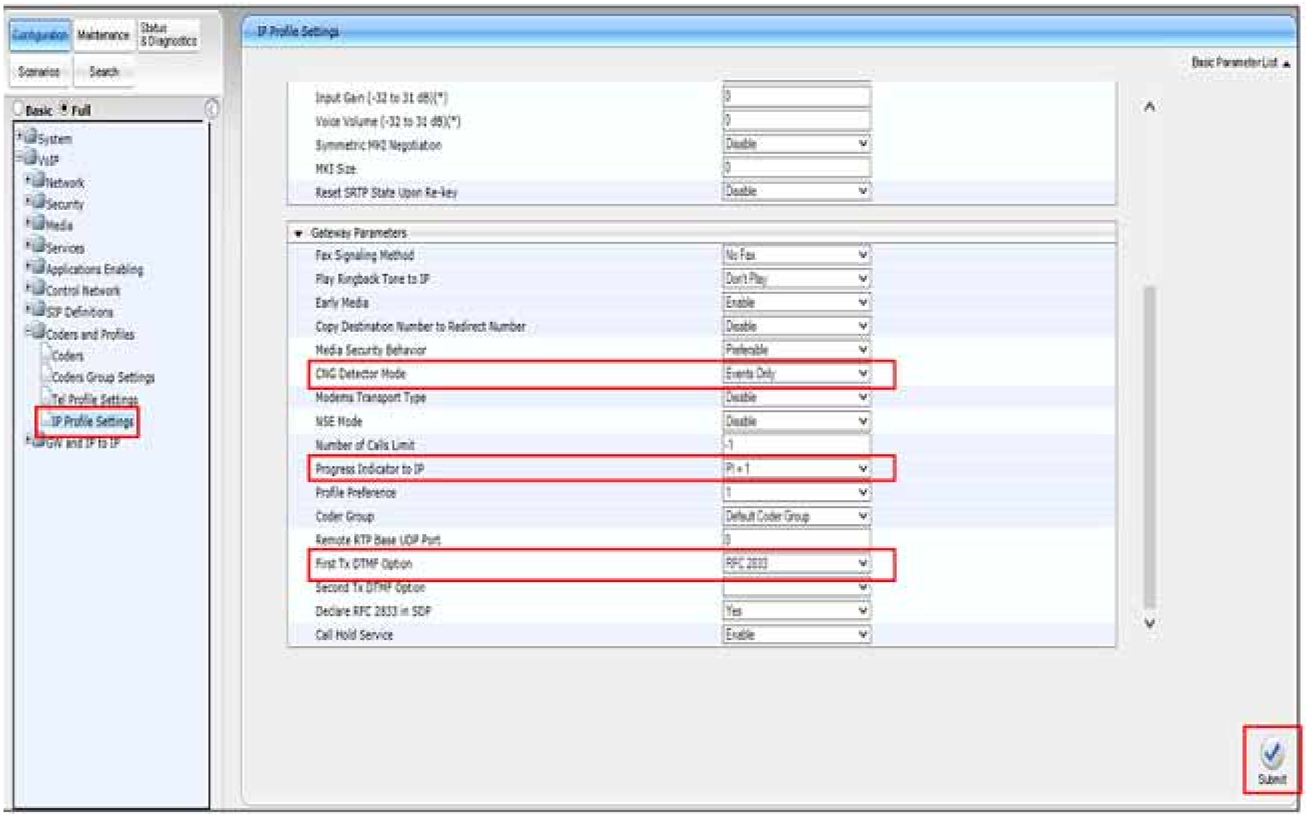
1. Open the **Coders Table** page (**Configuration- VoIP menu-Coder** and **profiles-Coders**).
2. Under the **Coder Name** drop-down list, select the required coder.
3. Under the **Packetization Time** drop-down list, select the packet size.
4. Under the **Silence Suppression** drop-down list select **Disabled**.
5. Repeat steps 2 through 6 for the next optional coders.
6. Click **Submit**.

**Figure 3-26 Coders IP Profiles Settings**



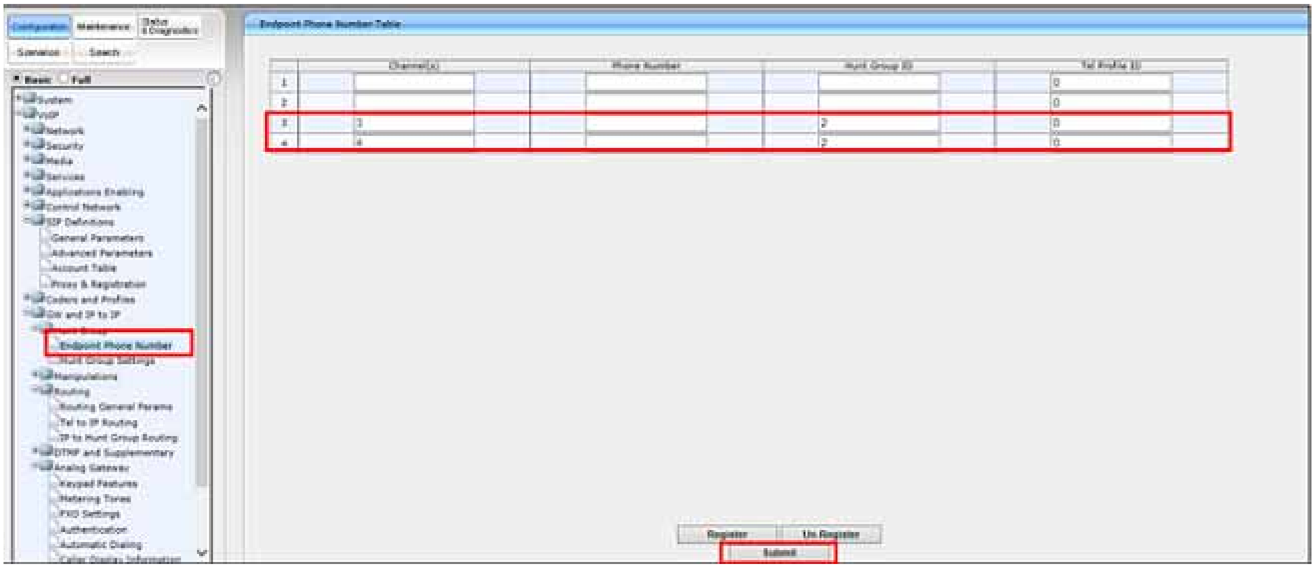

1. Open the **IP Profile Settings** page (**Configuration tab-VoIP menu- Coders** and **Profile- IP Profile Settings**).
2. Assign **Disconnect on Broken Connection** to **Yes**.
3. Assign **Early Media** to **Enable**.
4. Click **Submit**.

**Figure 3-27 Coders IP Profiles Settings Continued**



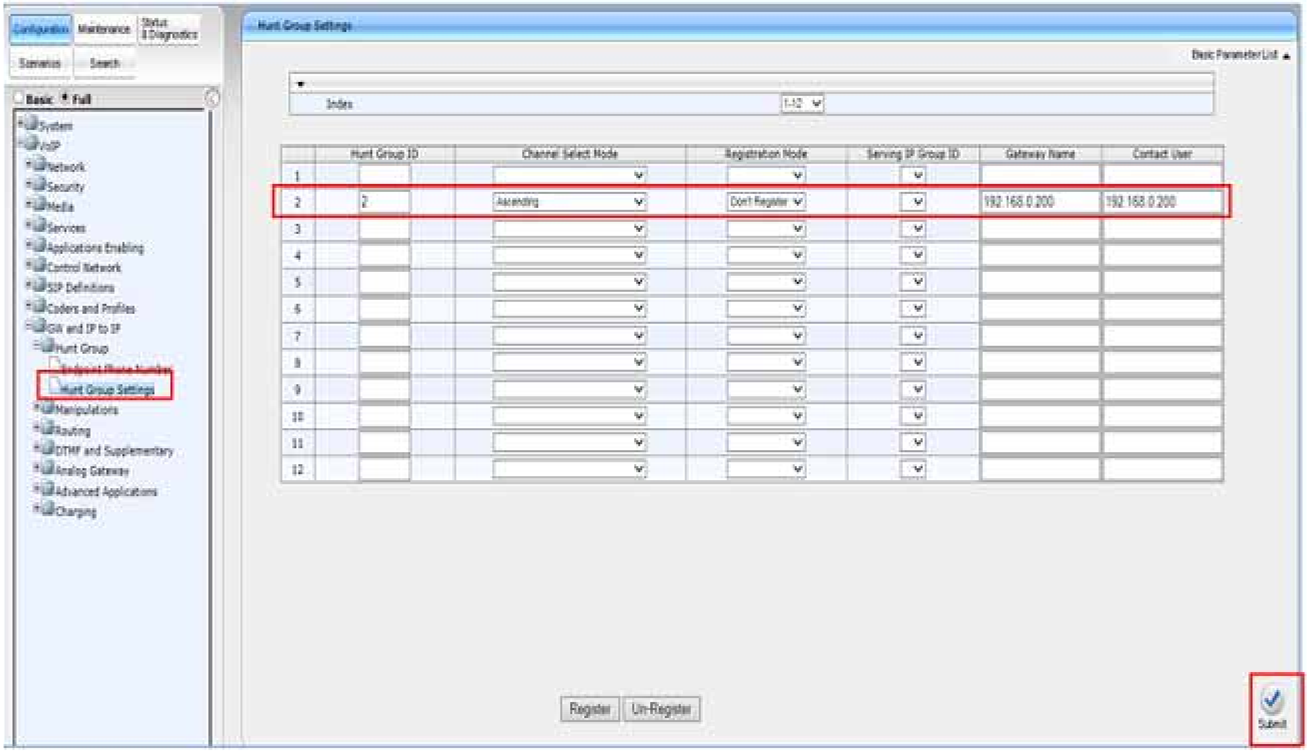

1. Open the **IP Profile Settings** page (**Configuration tab-VoIP menu- Coders** and **Profile- IP Profile Settings**).
2. Set **CNG Detector Mode** to **Events Only**.
3. Set **Progress Indicator IP** to **PI=1**.
4. Set **First TX DTMF Option** to **RFC 2833**.
5. Click **Submit**.

**Figure 3-28 Endpoint Phone Number Table Page**



1. Open the **Endpoint Phone Number Table** page (**Configuration tab- VoIP- GW** and **IP to IP- Hunt Group- Endpoint Phone Number**).
2. Assign ports **3** and **4** to channels **3** and **4**.
3. Assign the **Hunt Group** number to **02**.
4. Assign **Tel Profile ID**. (Optional)
5. Click **Submit**.

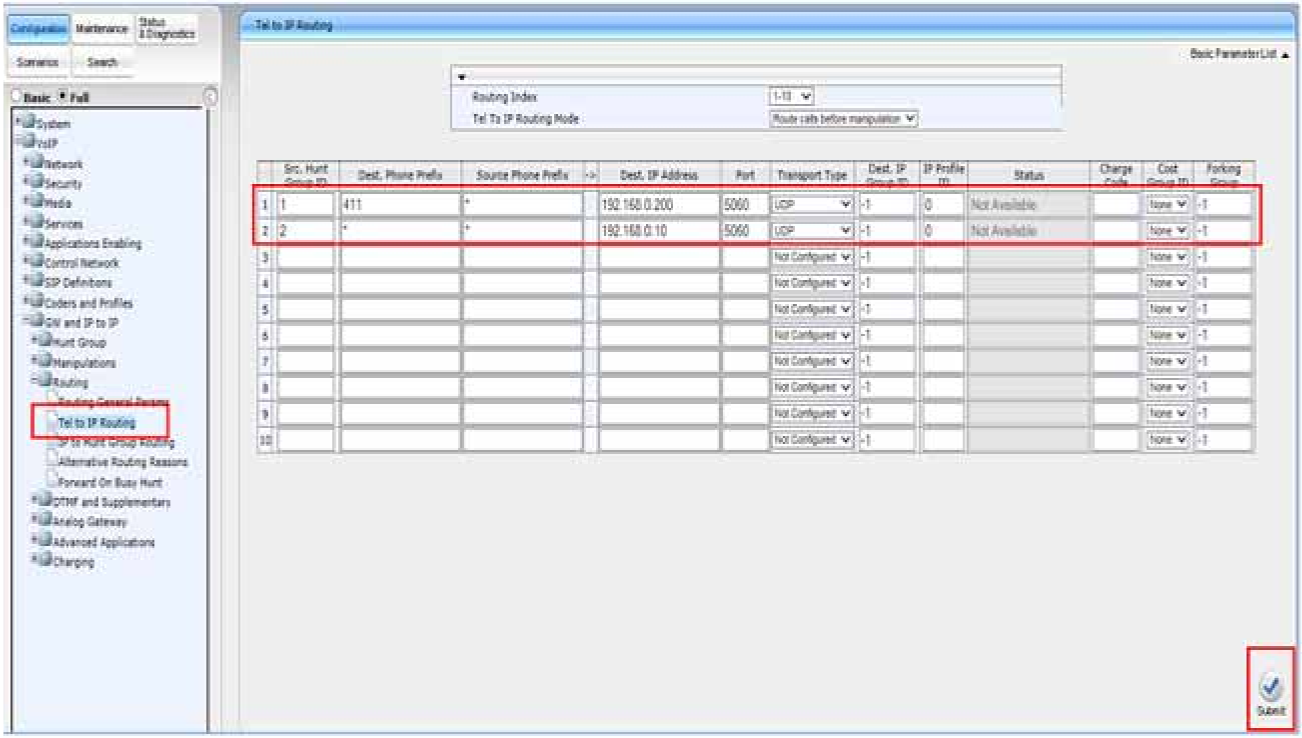
**Figure 3-29   Hunt Group Settings Page**




1. Open the **Hunt Group Settings** page (**Configuration-VoIP- GW** and **IP to IP-Hunt Group-Hunt Group settings**).
2. From the Index drop-down and select the range.
3. Assign **Hunt Group ID 2**.
4. Assign the **Channel Select Mode** to **Ascending**.
5. Assign **Registration Mode** to **Don't Register**.
6. Click **Submit**.

✎ *These are arbitrary numbers and will not register.*

**Figure 3-30 Configuring Tel to IP Routing**



1. Open **Tel to IP Routing** (**Configuration- VoIP menu- GW** and **IP to IP-Routing- Tel to IP**).

2. From the **Routing Index** drop-down list, select the range of entries that you want to add.

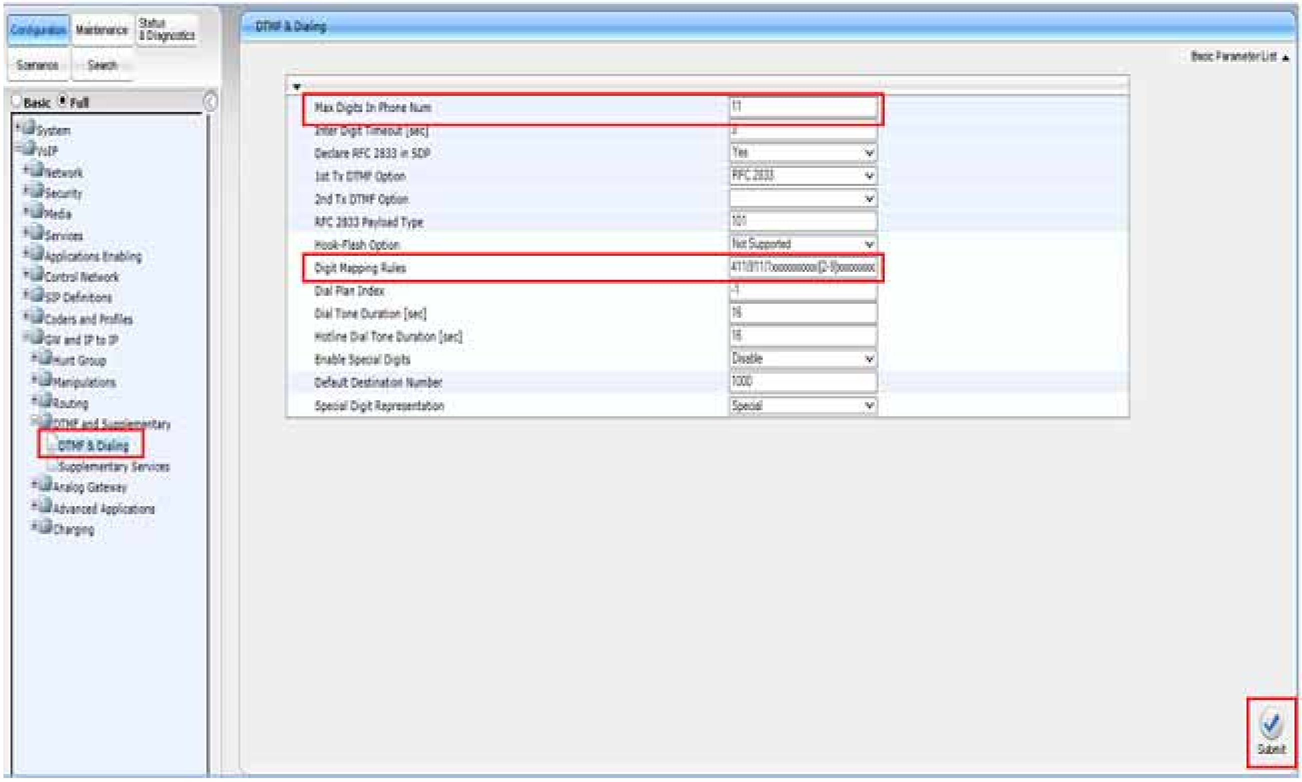
3. Configure the routing rule as required.

   The above example will send an incoming IP call to 411 out the FXO port. Any other incoming call will route out the IP network to the PBX.

4. Click **Submit** to apply your changes.

✎ *Refer to the User's Manual Tel to IP section for a detailed explanation of the routing parameters.*

**Figure 3-31 DTMF and Dialing**



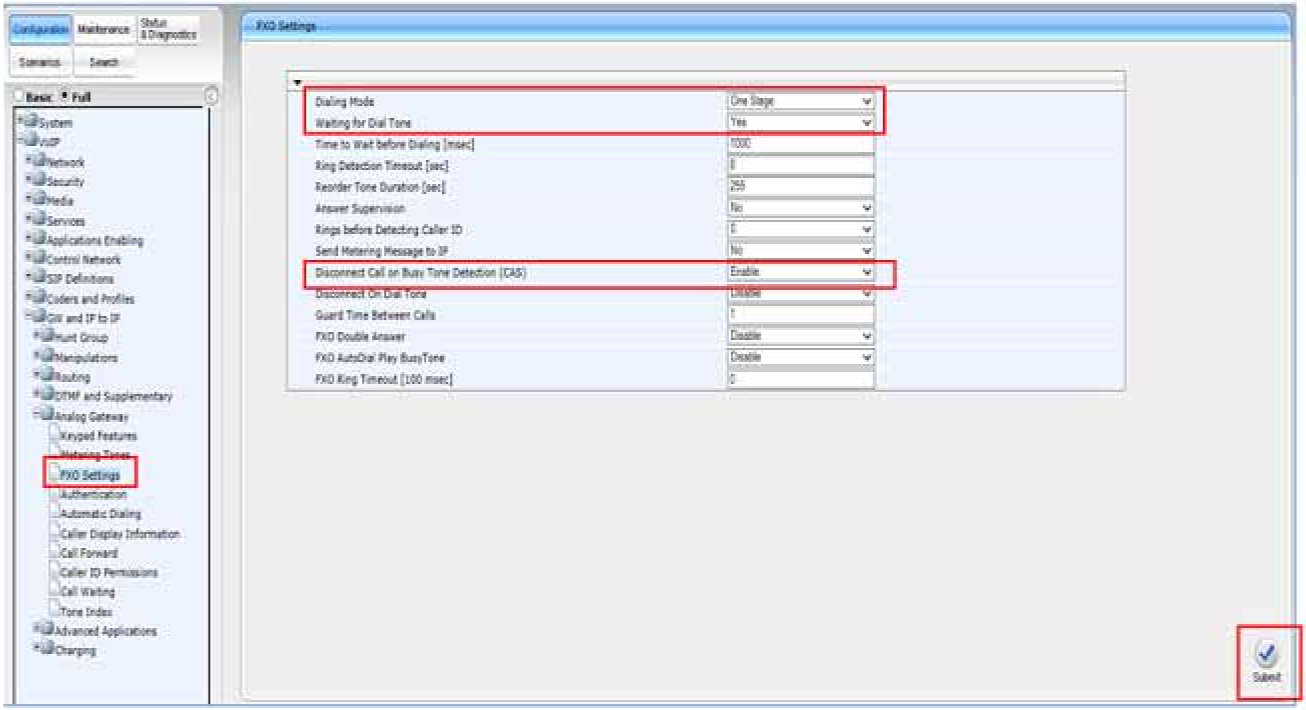
1. Open the **DTMF & Dialing** page (**Configuration- VoIP menu- GW** and **IP to IP- DTMF & Supplementary- DTMF & Dialing**).

2. Set **MAX Digits In Phone Num** to **11**.

   Digit Mapping rules can be added to make dialing more efficient.

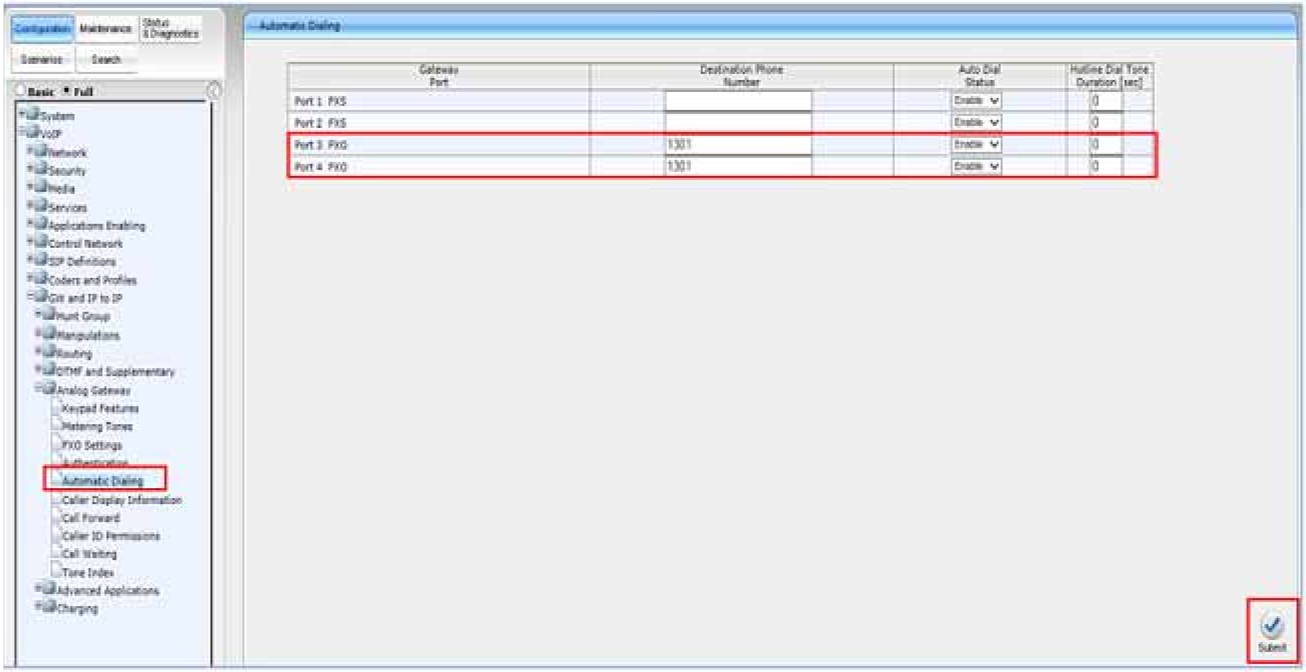
3. Click **Submit**.

**Figure 3-32 FXO Settings**




1. Open the **FXO Settings** page (**Configuration- VoIP- GW** and **IP to IP- Analog gateway- FXO settings**).
2. Set **Dialing Mode** to **One Stage**.
3. Set **Waiting for Dial Tone** to **Yes**.
4. Set **Disconnect Call on Busy Tone Detection** to **Enable**.
5. Click **Submit**.

**Figure 3-33 Analog Automatic Dialing**




1. Open the **Automatic Dialing** page (**Configuration- VoIP- GW** and **IP to IP- Analog gateway- Automatic Dialing**).

2. Configure Automatic dialing on a per port basis. This is used for direct termination and Hotline assignments.

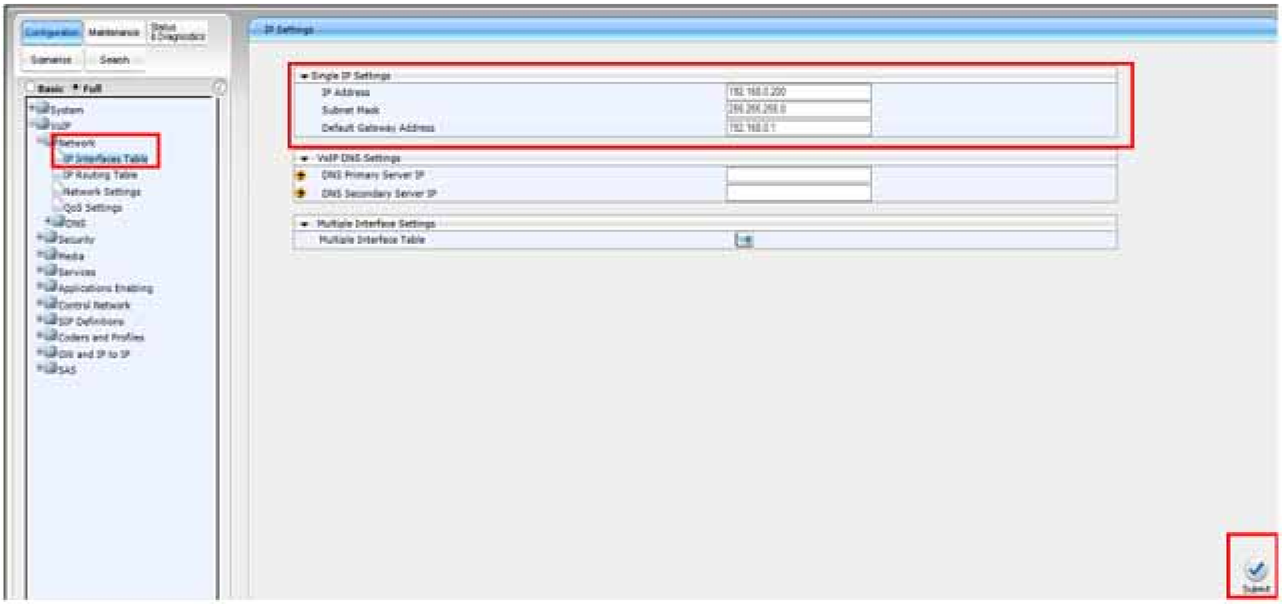
3. Click **Submit**.

   The above configuration sends any incoming call on FXO port 3 and 4 to 1301.

✎ *Refer to the User's Manual for additional information.*

## SECTION 3 COMBINATION FXS/FXO WITH SIP TRUNK CONNECTIVITY

**Figure 3-34 IP Settings Page (single network interface)**

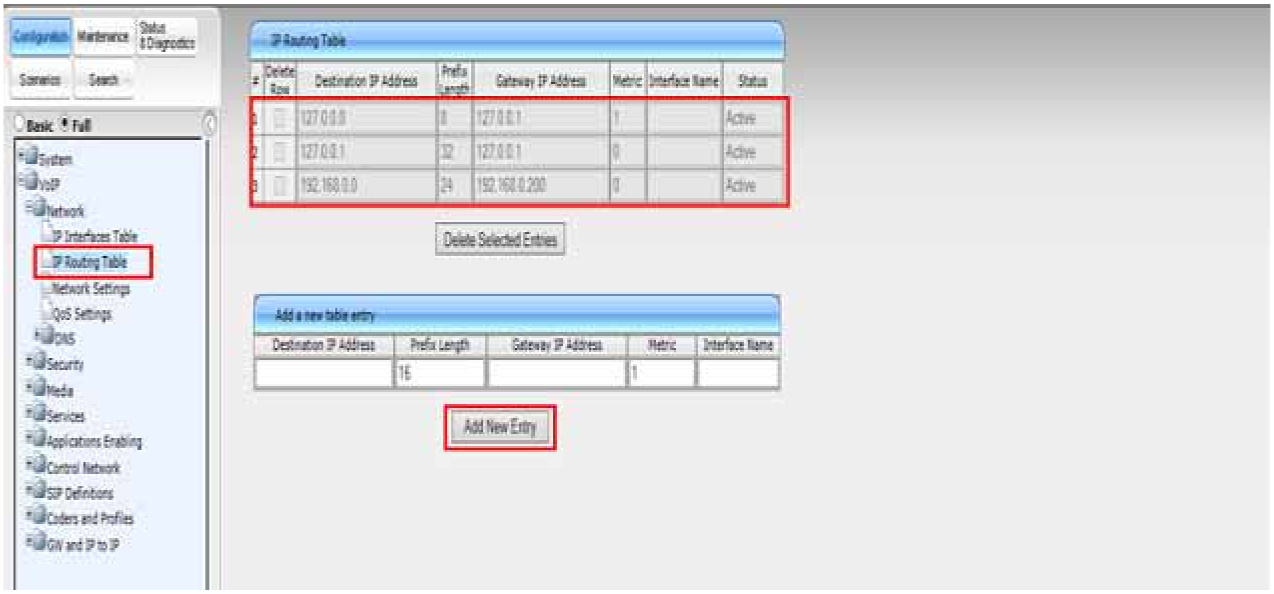


1. Open the **IP Settings** page (**Configuration tab- VoIP menu- Network-IP settings**).

   The initial IP address is 10.1.10.10.

2. Enter the **IP Address**, **Subnet Mask**, **Gateway** and **DNS Address** (if required).
3. Click **Submit**.

**Figure 3-35 IP Routing Table**



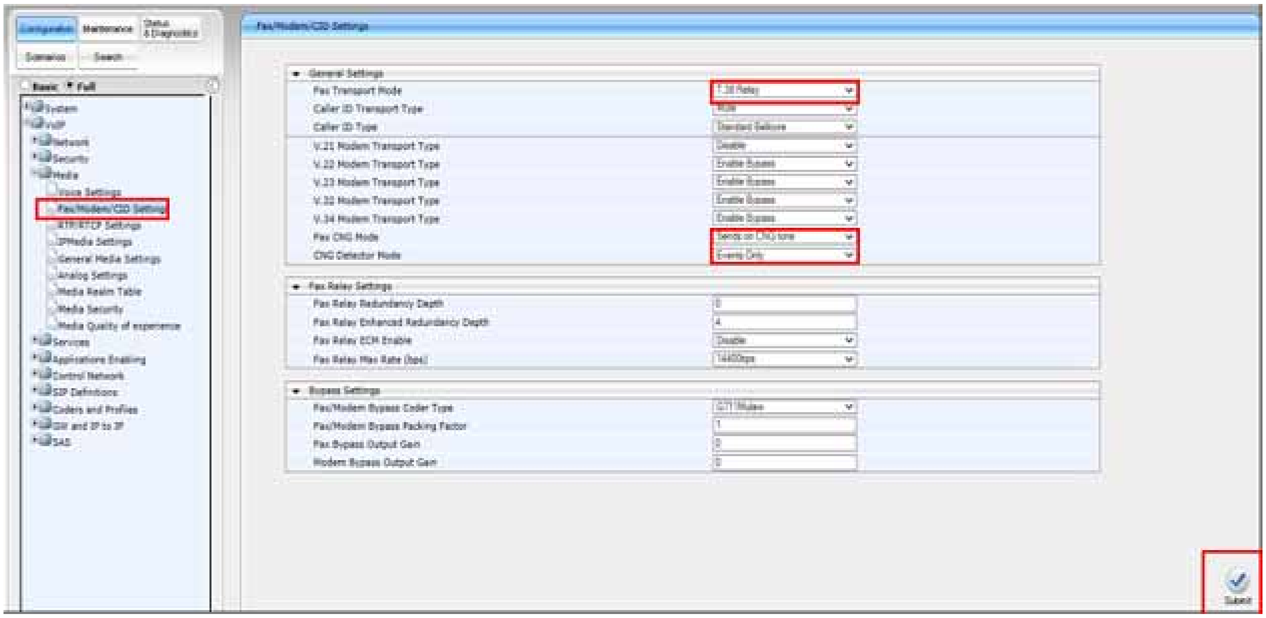

1. Open **IP Routing Table** page (**Configuration- VoIP- Network- IP Routing table**).

2. Enter route information under **Add a new table entry**.

3. Click **Add New Entry** and the new routing rule is added to the IP routing table.

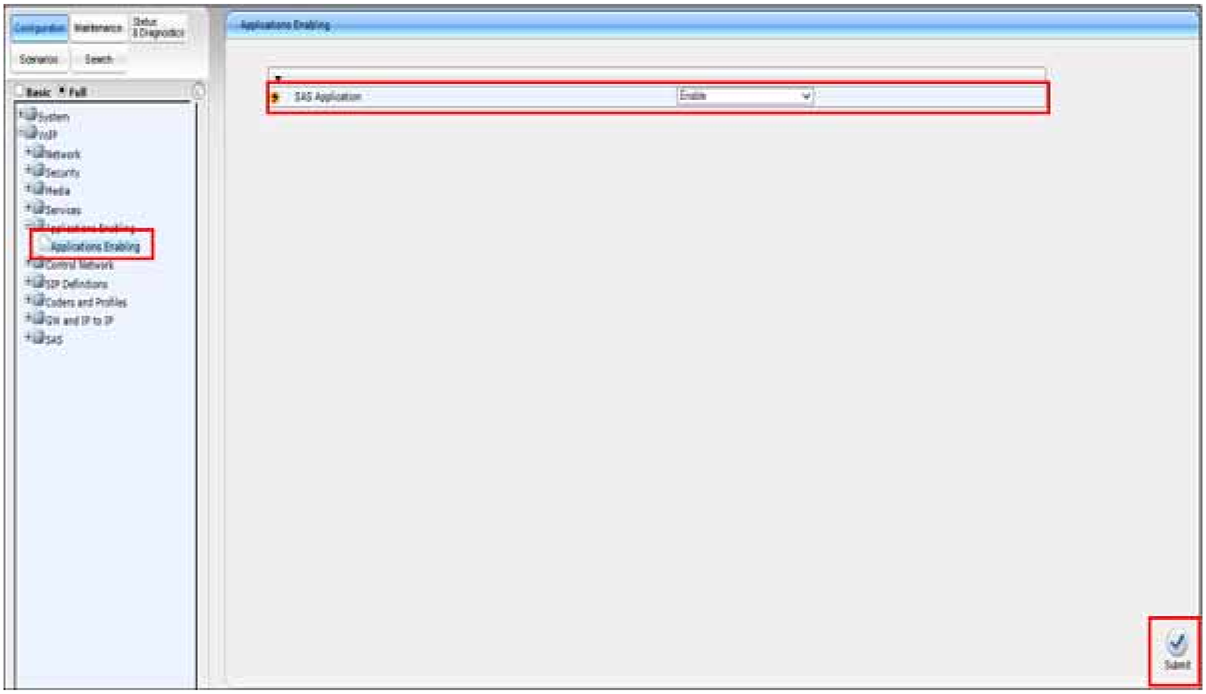
4. To delete a routing rule, select the **Delete Row** check box and then click **Delete**. Refer to the User's Manual for additional information.

**Figure 3-36 Media-Fax Settings**



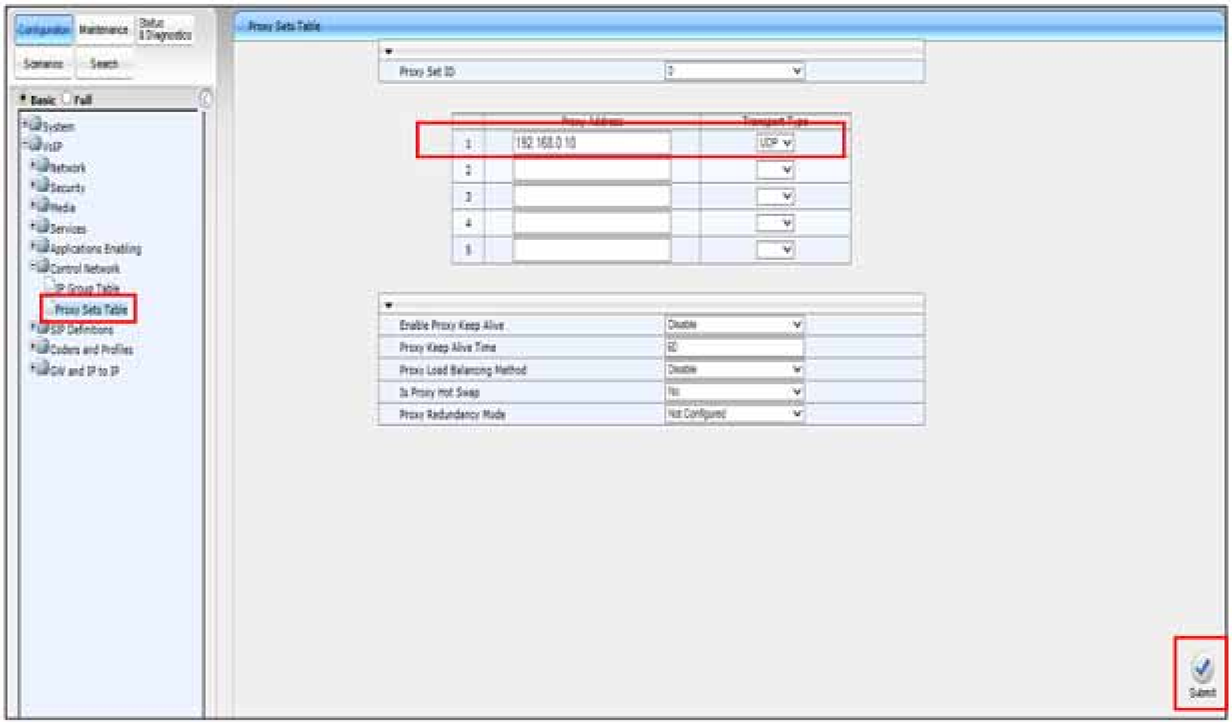

1. Open the **Fax/Modem/CID Settings** page (**Configuration tab- VoIP menu- Media- Fax/Modem/CID settings**).
2. Assign **Fax Transport** Mode to **T.38**.
3. Assign **Fax CNG Mode** to **Sends on CNG tone**.
4. Assign **CNG Detector Mode** to **Events only**.
5. Click **Submit**.

**Figure 3-37 Applications Enabling**



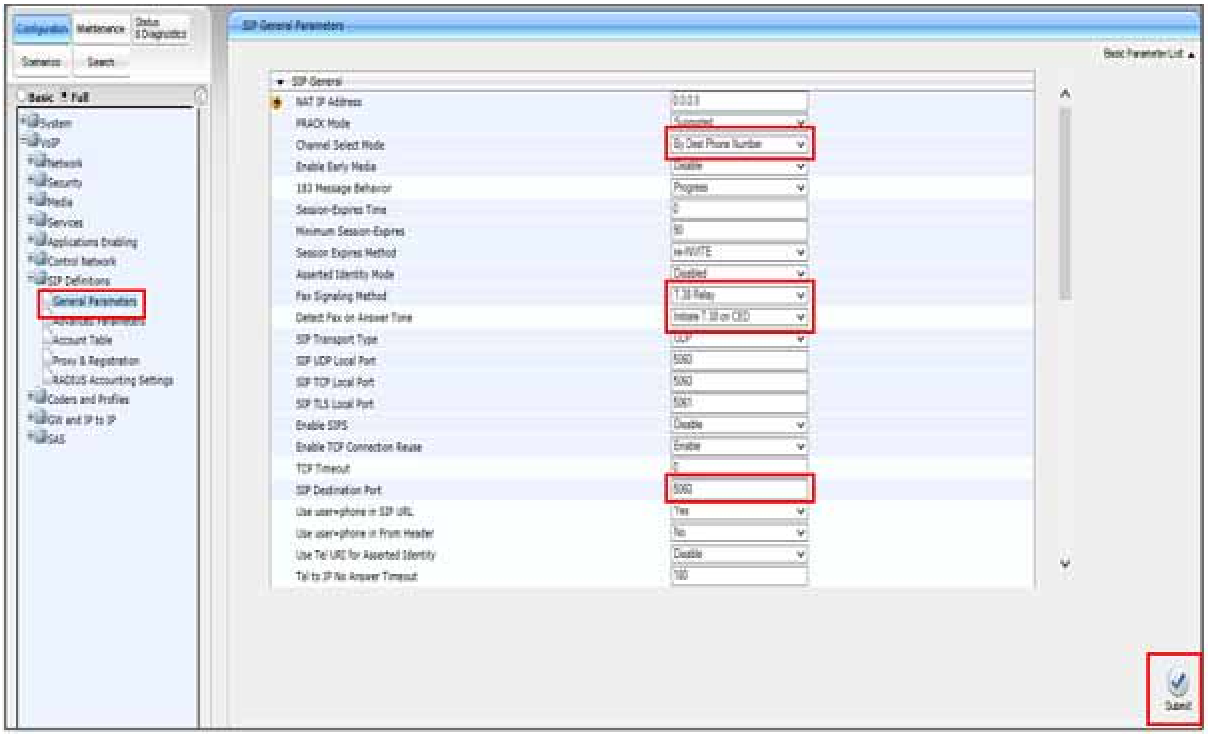

1. Open the **Applications Enabling** page (**Configuration tab- VoIP menu- Applications Enabling submenu- Applications Enabling**).
2. From the relevant application drop-down list, select **Enable**.
3. Click **Submit**.

**Figure 3-38 Proxy Sets Table**



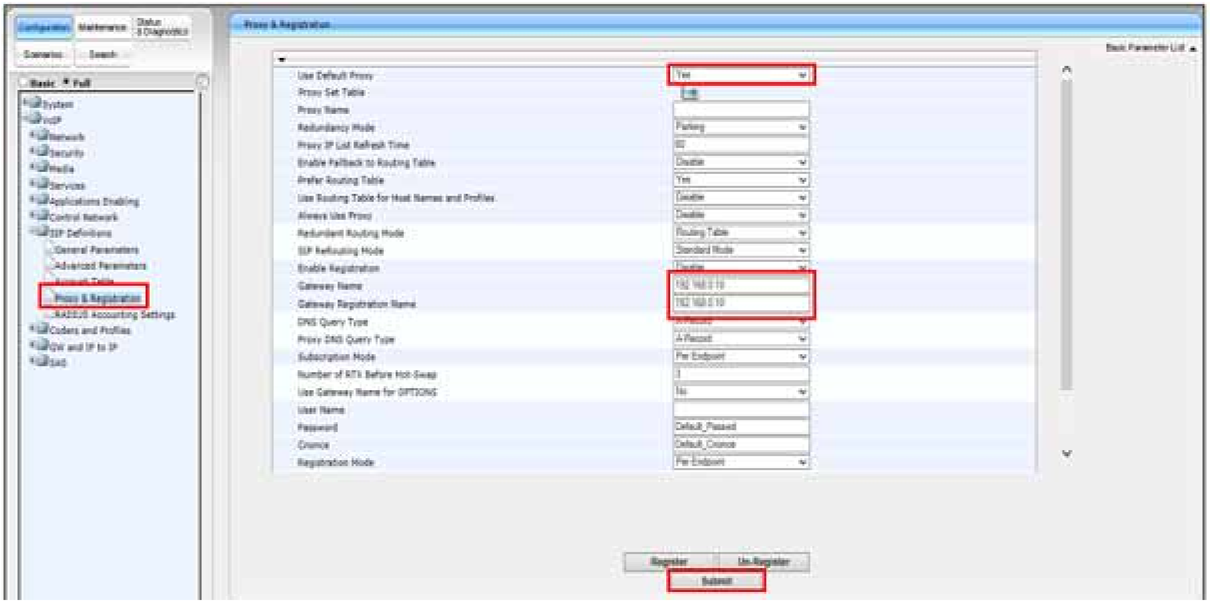

1. Open the **Proxy Sets Table** page (**Configuration- VoIP Menu- Control Network-Proxy Sets Table**).
2. Select a **Proxy Set ID** from the drop down.
3. Configure the **Proxy Address** as required (IP address of the SV9300).
4. Leave all other settings at default.
5. Click **Submit**.

**Figure 3-39 SIP Definitions General Parameters**



1. Open **SIP Definitions General Parameters** (**Configuration tab- VoIP menu- SIP Definitions- General Parameters**).
2. Assign **Channel Select Mode** to **By Destination Phone Number**.
3. Assign **Fax Signaling Method** to **T.38 Relay**.
4. Assign **Detect Fax on Answer Tone** to **Initiate T.38 on CED**.
5. Assign **SIP Destination Port** to **5060**.
6. Click **Submit**.

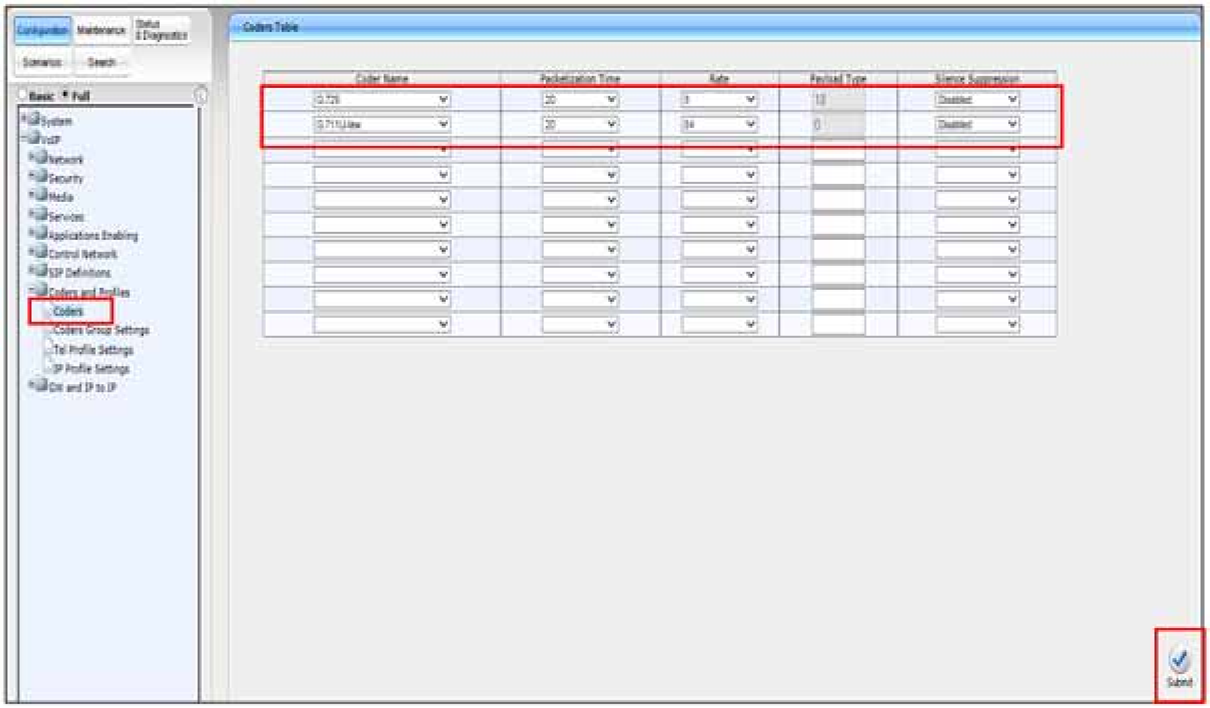
**Figure 3-40 SIP Definitions Proxy and Registration**




1. Open the **Proxy & Registration** page (**Configuration tab- VoIP menu- SIP Definitions- Proxy and Registration**).
2. Assign **Use Default Proxy** to **Yes**.
3. Assign **Prefer Routing Table** to **Yes**.
4. Assign the **Gateway Name** to the SV9300 IP address. (EX: 192.168.0.10)
5. Assign the **Gateway Registration Name** to the SV9300 IP address. (EX: 192.168.0.10)
6. Click **Submit**.

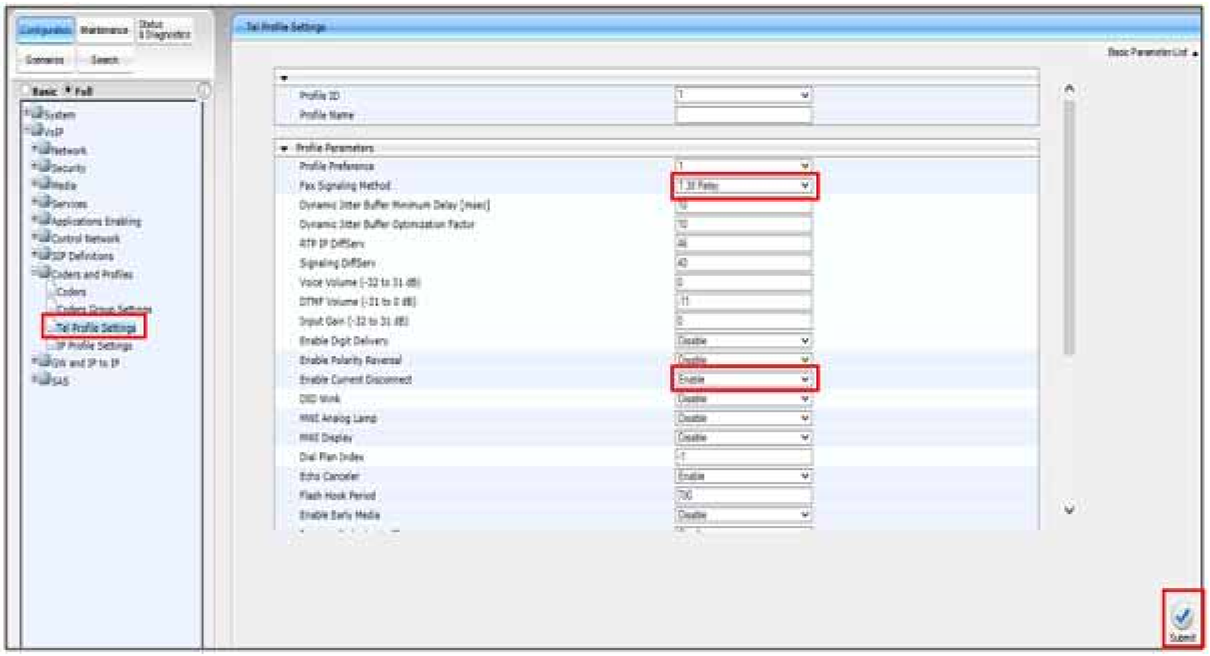
✎ *This is a non registered SIP trunk.*

**Figure 3-41 Coders Table**



1. Open the **Coders** page (**Configuration- VoIP menu- Coder and profiles- Coders**).
2. Under **Coder Name** drop-down list, select the required coder.
3. Under **Packetization Time** drop-down list, select the packet size.
4. Under **Silence Suppression** drop-down list set to **Disabled**.
5. Repeat the steps for the next optional coders.
6. Click **Submit**.

**Figure 3-42 Coders- Tel Profile**



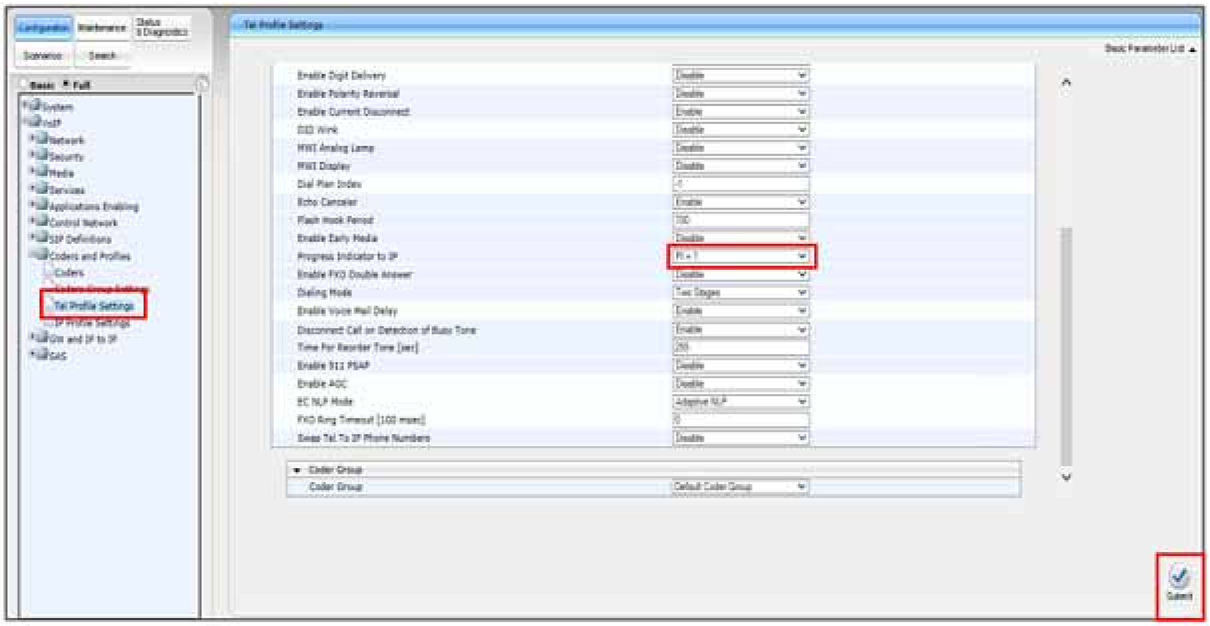
1. Open the **Tel Profile Settings** page (**Configuration tab- VoIP menu- Coders and Profiles submenu- Tel Profile Settings**).

2. Assign **Fax Signaling Method** to **T.38**.

3. Assign **Enable Current Disconnect** to **Enable**.

4. Click **Submit**.

**Figure 3-43 Coders- Tel Profile Continued**

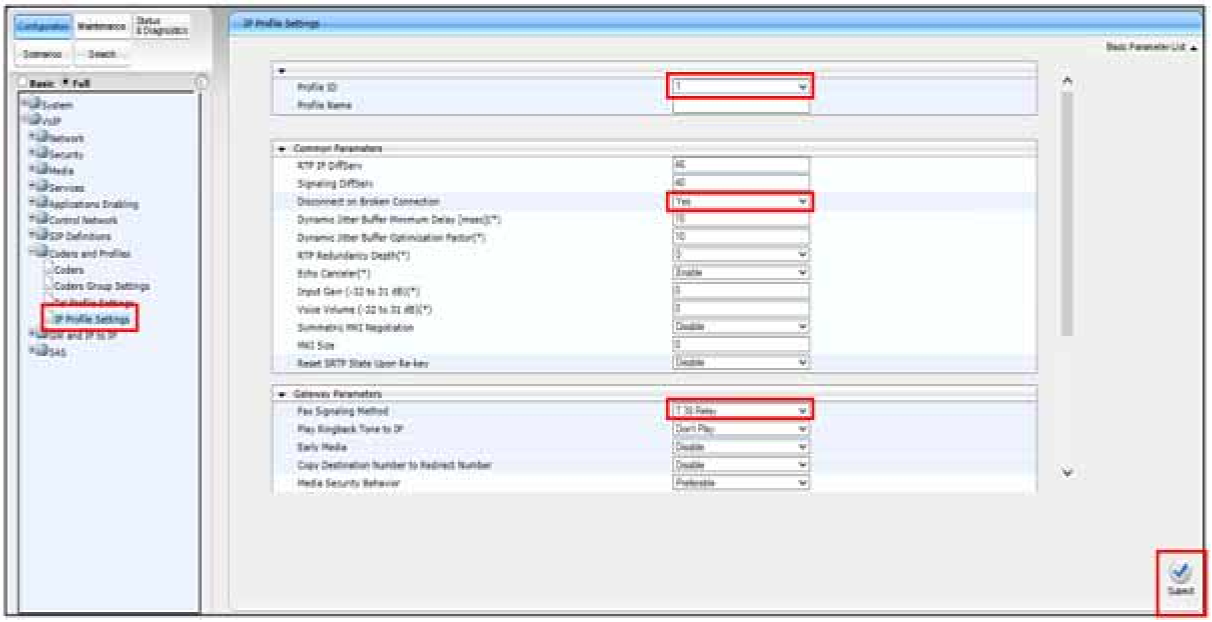


1. Open the **Tel Profile Settings** page (**Configuration tab- VoIP menu- Coders and Profiles submenu- Tel Profile Settings**).

2. Assign Pro**gress Indicator to IP** to **PI=1**.

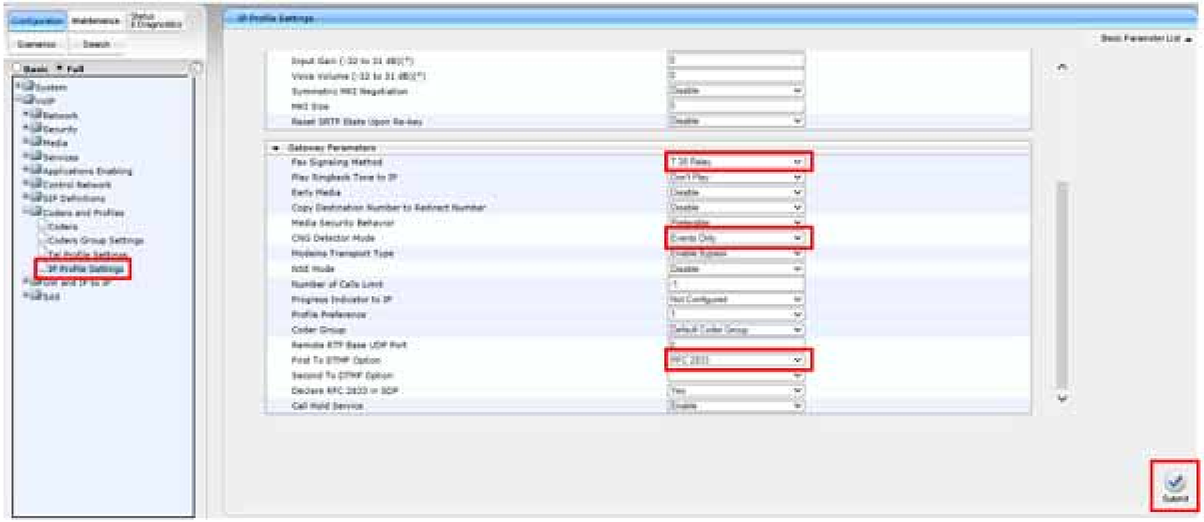
3. Click **Submit**.

**Figure 3-44 Coders IP Profiles Settings**



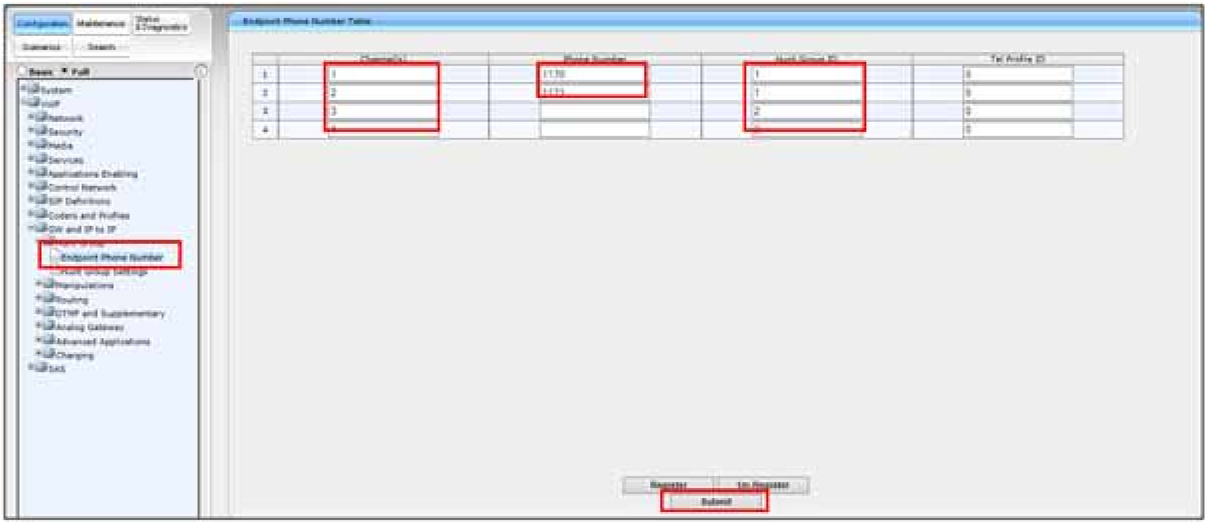
1. Open the **IP Profile Settings** page (**Configuration tab- VoIP menu- Coders and Profile- IP Profile Settings**).
2. Select **Profile ID 1**.
3. Assign **Disconnect on Broken Connection** to **Yes**.
4. Assign **Fax Signaling Method** to **T.38**.
5. Click **Submit**.

**Figure 3-45 Coders IP Profiles Settings Continued**



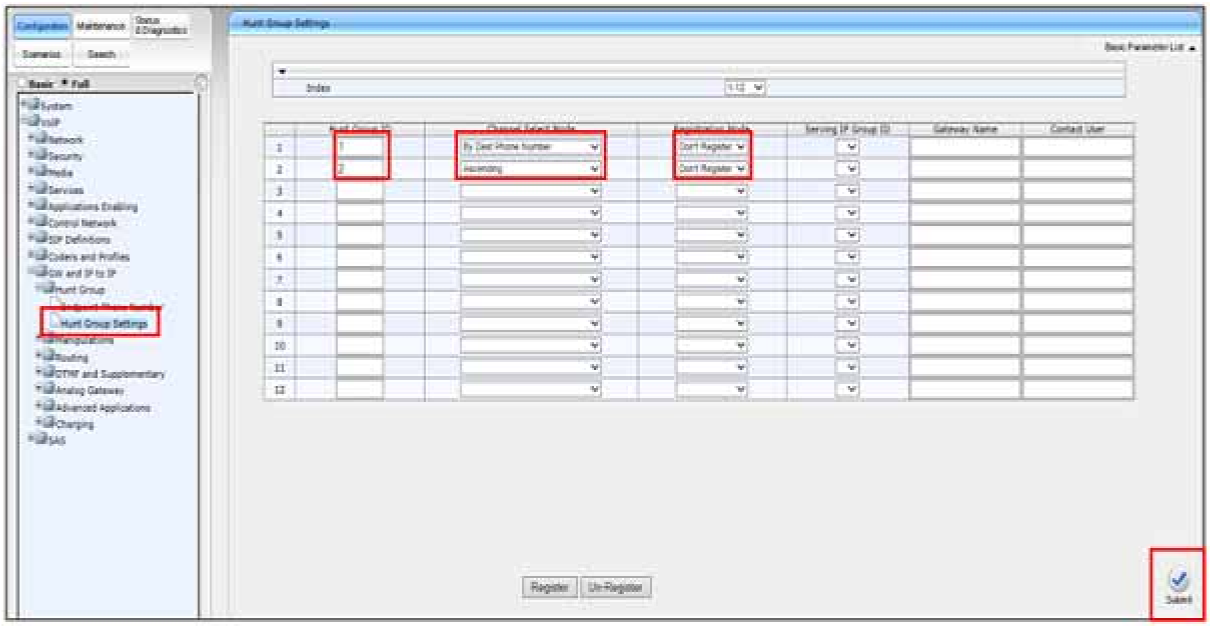
1. Open the **IP Profile Settings** page (**Configuration tab-VoIP menu- Coders and Profile- IP Profile Settings**).
2. Assign **Fax Signaling Method** to **T.38**.
3. Assign **CNG Detector Mode** to **Events Only**.
4. Assign **First TX DTMF Option** to **RFC 2833**.
5. Click **Submit**.

**Figure 3-46 Endpoint Phone Number Table Page**



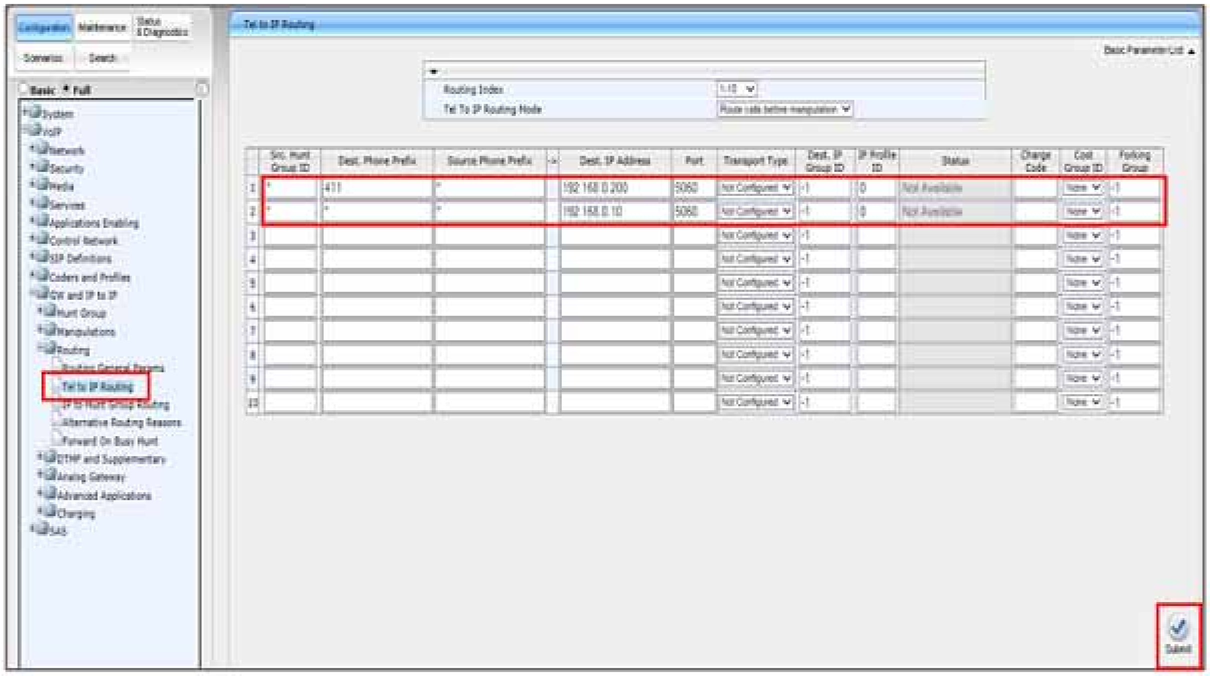
1. Open the **Endpoint Phone Number Table** page (**Configuration tab- VoIP- GW and IP to IP- Hunt Group- Endpoint Phone Number**).
2. Configure the **Endpoint Phone Numbers** to the Channels. (EX: 1170 and 1171)
3. Assign **Channel** numbers to the port numbers.
4. Assign **Phone Number** to an unassigned number in the PBX,
5. Assign **Hunt Group** number to **01** for channels 1 and 2.
6. Assign **Hunt Group** number to **02** for channels 3 and 4.
7. Assign **Tel Profile ID**. (Optional)
8. Click **Submit**.

**Figure 3-47 Hunt Group Settings Page**



1. Open the **Hunt Group Settings** page (**Configuration- VoIP- GW and IP to IP- Hunt Group- Hunt Group settings**).
2. Select the **Index** drop down and select the range.
3. Assign **Hunt group ID** to a **1** to the first entry, this assigns FXS to hunt group 1.
4. Assign **Hunt group** ID to a **2** to the second entry, this assigns FXO to hunt group 2.
5. Assign **Channel Select Mode** to **By Destination Phone Number** to Hunt group 1.
6. Assign **Channel Select Mode** to **Ascending** for hunt group 2.
7. Assign **Registration Mode** to **Don't Register** for both.
8. Click **Submit**.

**Figure 3-48 Configuring Tel to IP Routing**



1. Open **Tel to IP Routing** (**Configuration- VoIP menu- GW and IP to IP- Routing- Tel to IP**).

2. From the **Routing Index** drop-down list, select the range of entries that you want to add.

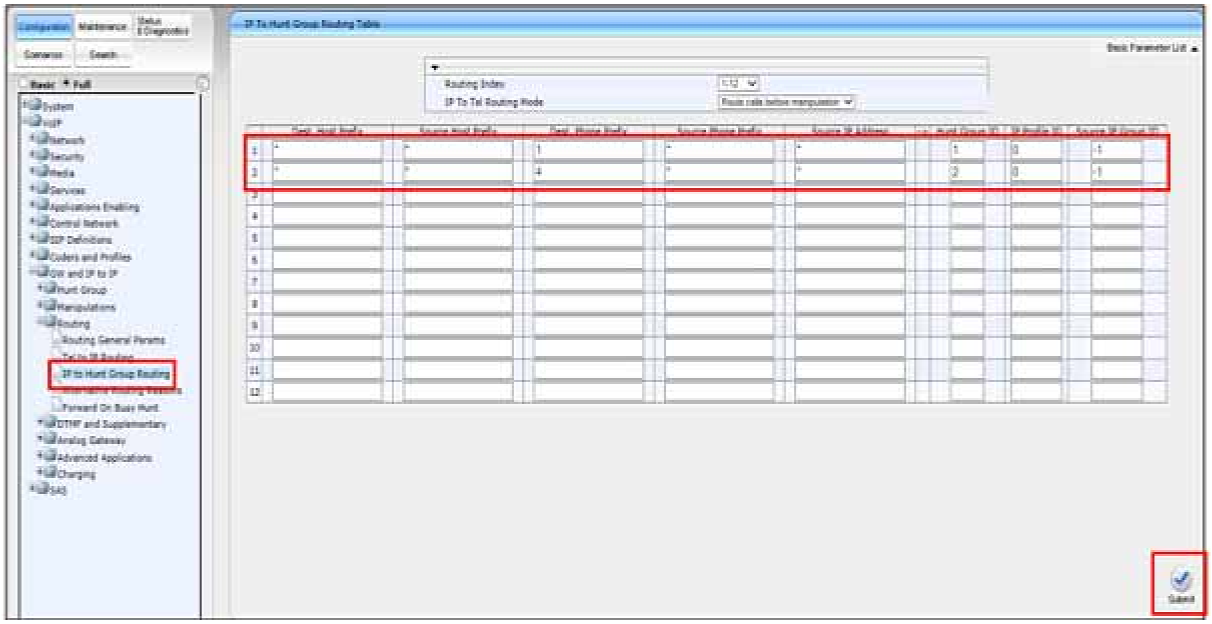
3. Configure the routing rule as required.

   The above example will send an outgoing call to 411 out the FXO port. Any other outgoing call will route out the IP network to the PBX.

4. Click **Submit**.

✎ *Refer to the User's Manual Tel to IP section for a detailed explanation of the routing parameters.*

**Figure 3-49 Configuring IP to Hunt Group Routing**



1. Open the **IP to Hunt Group Routing Table** page (**Configuration tab- VoIP menu- GW and IP to IP- Routing- IP to hunt group routing**).

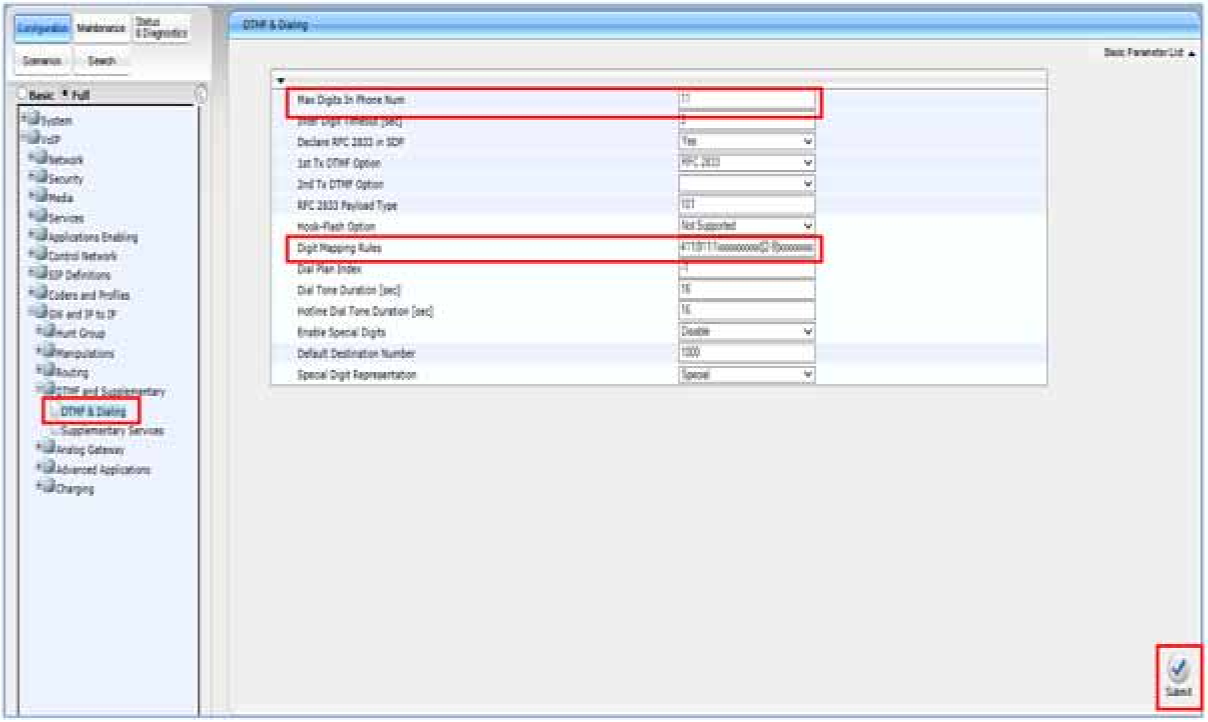
   The above example will send an incoming call starting with the digit 1 to hunt group 1 (FXS hunt group).

   The above example will send an incoming call starting with the digit 4 to hunt group 2 (FXO hunt group).

2. Click **Submit**.

✎ *Refer to the User's Manual IP to Hunt group section for a detailed explanation of the routing parameters.*

**Figure 3-50 DTMF and Dialing**



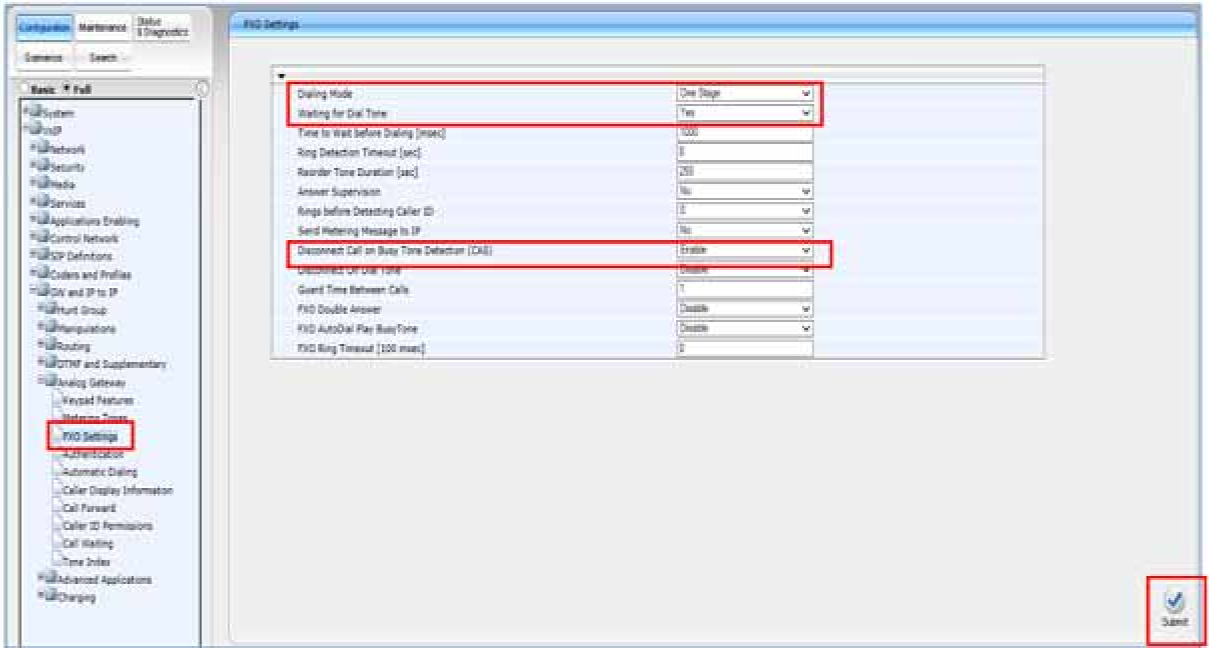

1. Open the **DTMF & Dialing** page (**Configuration- VoIP menu- GW and IP to IP- DTMF & Supplementary- DTMF & Dialing**).

2. Assign **MAX Digits in Phone Num** to 11.

   Digit mapping rules can be added to make dialing more efficient.

3. Click **Submit**.

**Figure 3-51 FXO Settings**



1. Open **FXO Settings** (**Configuration- VoIP- GW and IP to IP- Analog gateway- FXO settings**).
2. Assign **Dialing Mode** to **One Stage**.
3. Assign **Waiting for Dial Tone** to **Yes**.
4. Assign **Disconnect Call on Busy Tone Detection** to **Enable**.
5. Click **Submit**.

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UNIVERGE® SV9300

# AUDIOCODES MEDIAPACK FXS/FXO CONFIGURATION GUIDE

NEC Corporation of America

Issue 1.0