

# ***UNIVERGE***<sup>®</sup> ***SV9300***

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## AUDIOCODES MEDIAPACK FXS/FXO CONFIGURATION GUIDE



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# Chapter 1 *Introduction*

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## 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 10.1.10.16.*

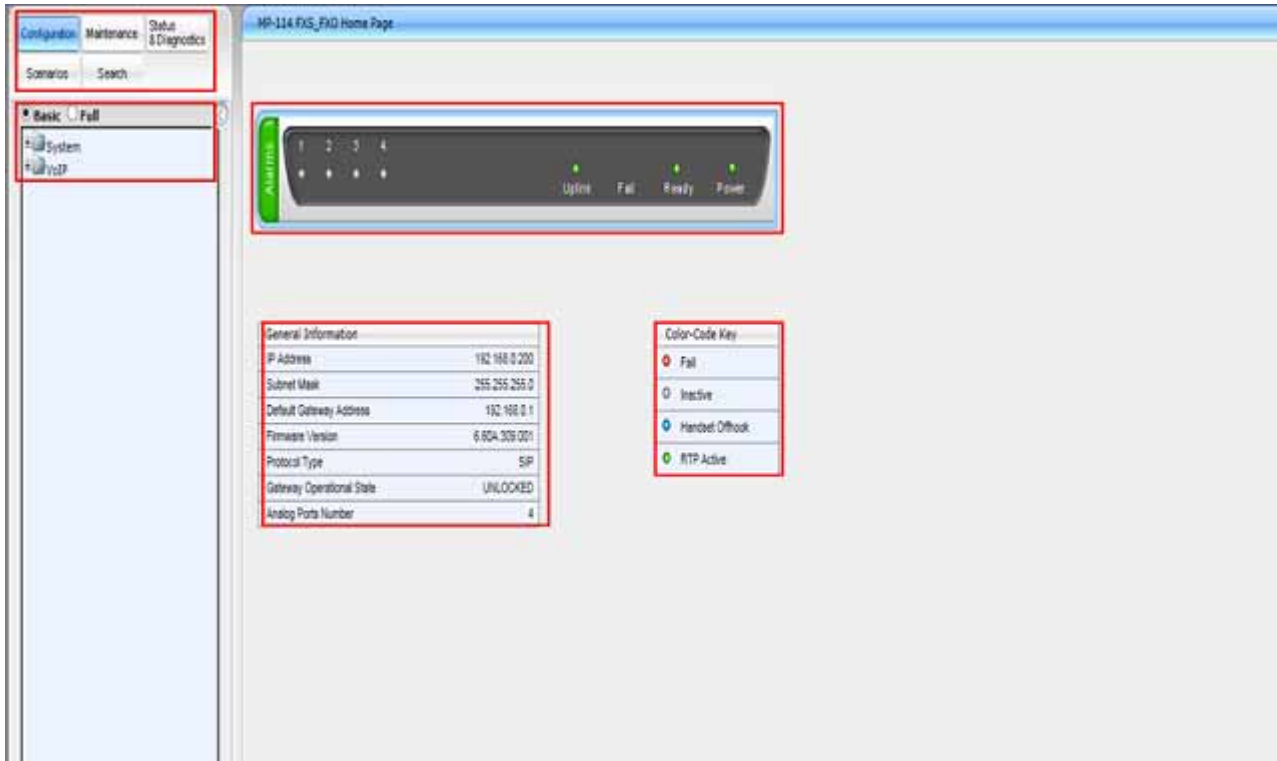
## SECTION 2 LIMITATIONS

- 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.

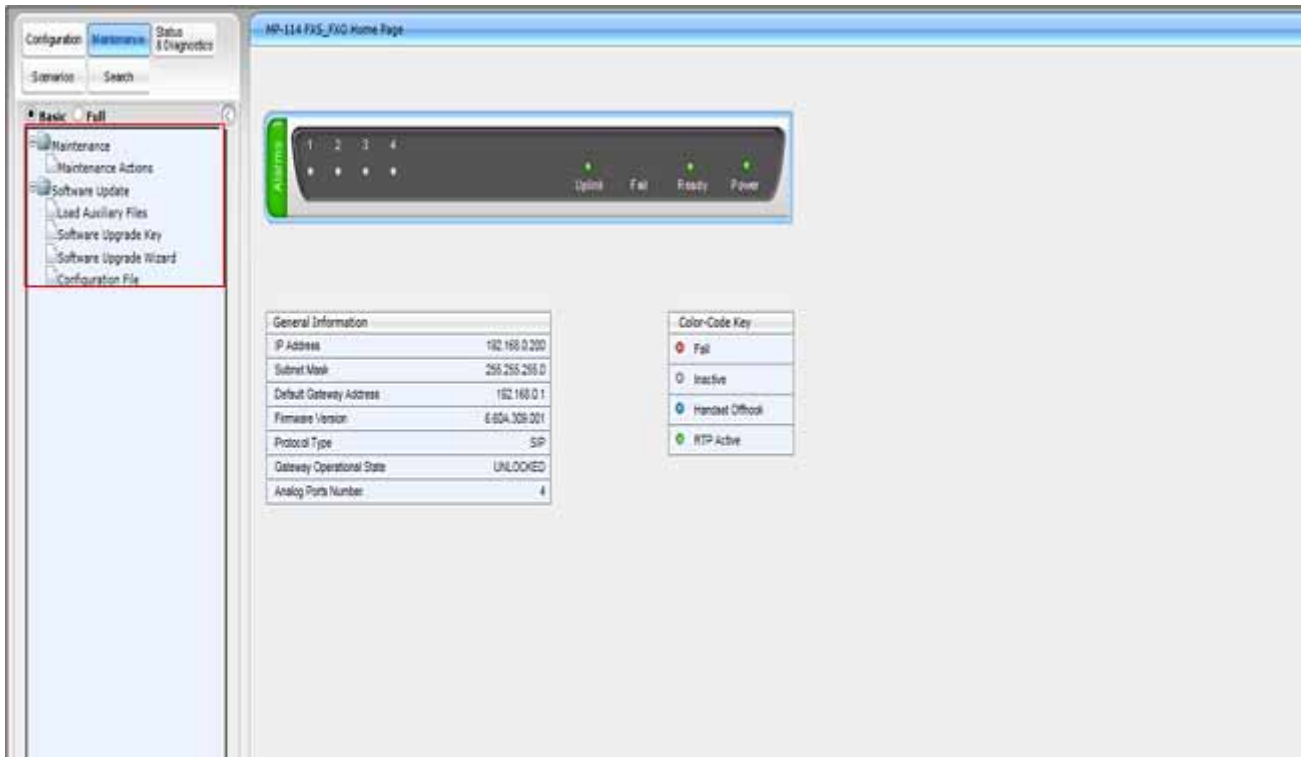
Figure 1-2 Audio Codes Configuration Tab



The following are features of the Configuration tab.

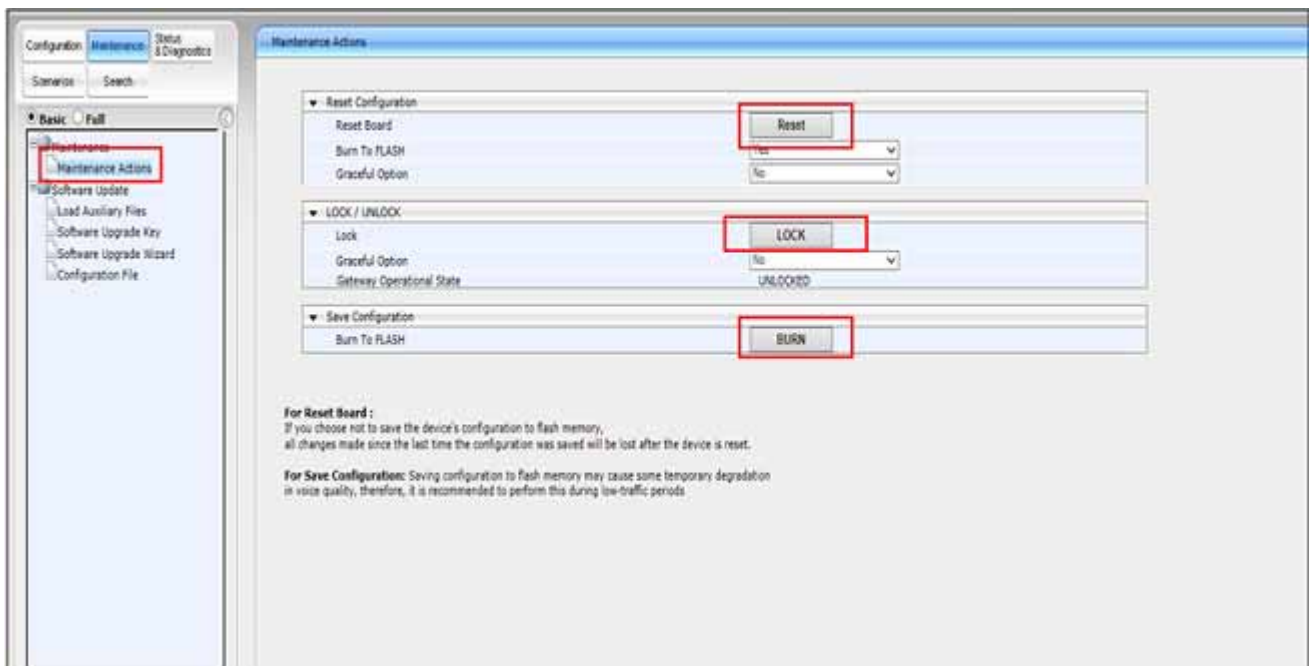
1. The configuration tab will show Basic or Full mode. Within each screen there may be additional settings that will not be available in basic mode. You will need to select the **Full** mode to see additional settings.
2. The configuration mode has a **System** tab and a **VOIP** tab.
3. The System tab has **Application** settings such as 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.

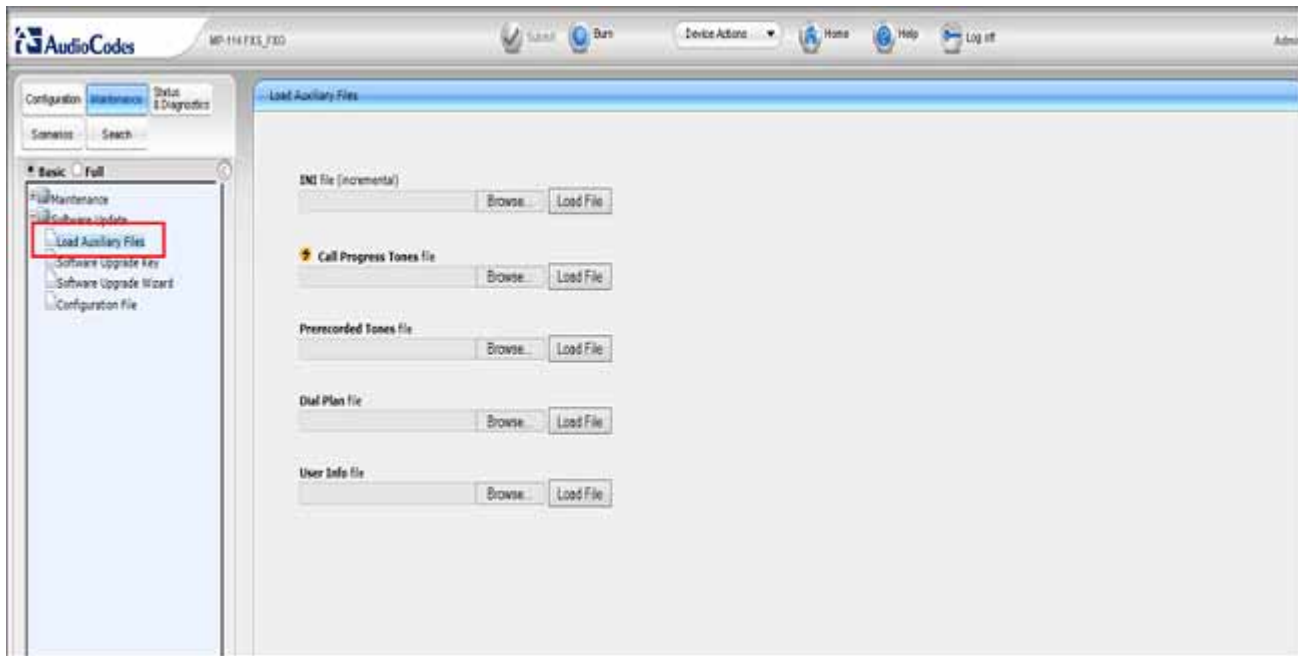
Figure 1-4 Maintenance Actions



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 screenshot displays the 'NP-324 FRS\_FXS Home Page' with the 'Statistics & Diagnostics' tab selected. The left sidebar contains a tree view with 'System Status' and 'Carrier Grade Alarms' highlighted. The main content area features a status bar with four indicators: Uplink (green), Fail (red), Ready (green), and Power (green). Below the status bar are two tables:

General Information	
IP Address	192.168.8.200
Subnet Mask	255.255.255.0
Default Gateway Address	192.168.8.1
Firmware Version	6.604.309-001
Protocol Type	SIP
Gateway Operational State	UNLOCKED
Analog Ports Number	4

Color-Code Key	
<span style="color: red;">●</span>	Fail
<span style="color: gray;">○</span>	Inactive
<span style="color: blue;">●</span>	Handset Offhook
<span style="color: green;">●</span>	RTP Active

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 screenshot shows a web interface for configuring scenarios. On the left side, there is a light blue sidebar containing a form. The form has a header section labeled "Scenario Name -" with a large text input area. Below this, there are two smaller text input fields labeled "Scenario Name:" and "Step Name:". At the bottom of the sidebar, there are three buttons: "Save & Finish", "Cancel Scenario", and "Get/Set Scenario File". The main area of the page is a large, empty light gray rectangle, likely intended for a list of scenarios or a detailed configuration view.

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 screenshot shows the AudioCodes MP-114 FXS\_FXO Home Page. The top navigation bar includes 'Home' and 'Log off' buttons. The left sidebar contains 'Configuration', 'Maintenance', and 'Status & Diagnostics' tabs, with 'Search' selected. A search input field and a 'Search' button are highlighted with a red box. Below the search field is a 'Search History' dropdown menu. The main content area features a device status bar with indicators for 'Uplink', 'Fail', 'Ready', and 'Power'. Below this are two tables: 'General Information' and 'Color-Code Key'.

General Information	
IP Address	192.168.0.200
Subnet Mask	255.255.255.0
Default Gateway Address	192.168.0.1
Firmware Version	6.604.309.001
Protocol Type	SIP
Gateway Operational State	UNLOCKED
Analog Ports Number	4

Color-Code Key	
<span style="color: red;">●</span>	Fail
<span style="color: gray;">○</span>	Inactive
<span style="color: blue;">●</span>	Handset Offhook
<span style="color: green;">●</span>	RTP Active

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

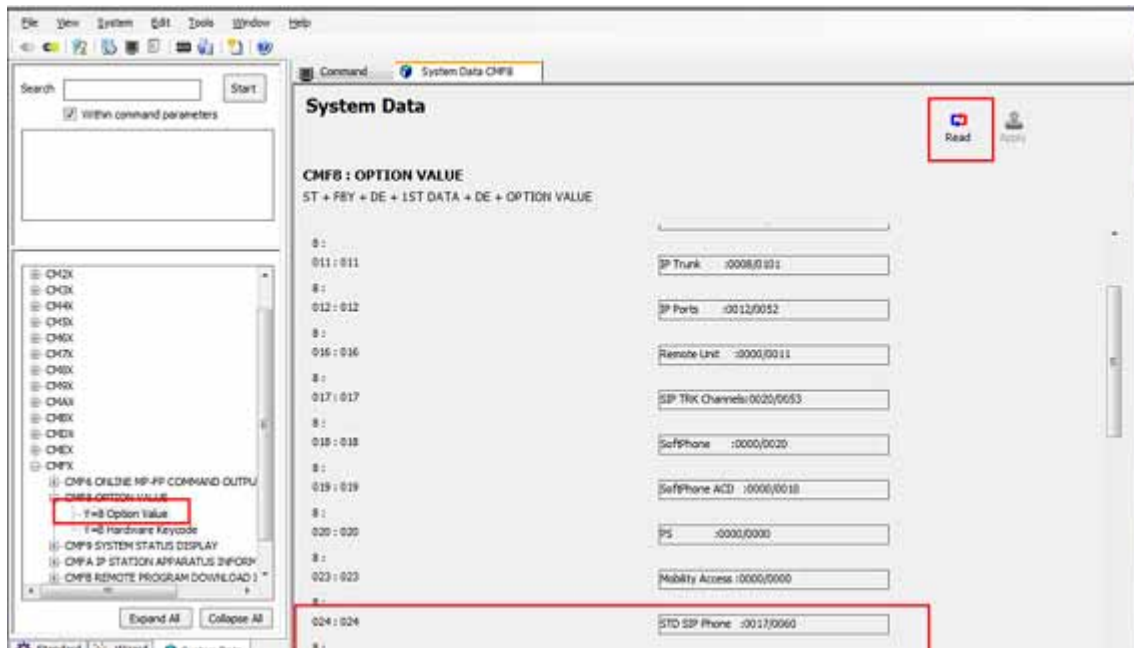
The SV9300 will need to be programmed to use Standard SIP stations for connectivity to the AudioCodes.

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**.

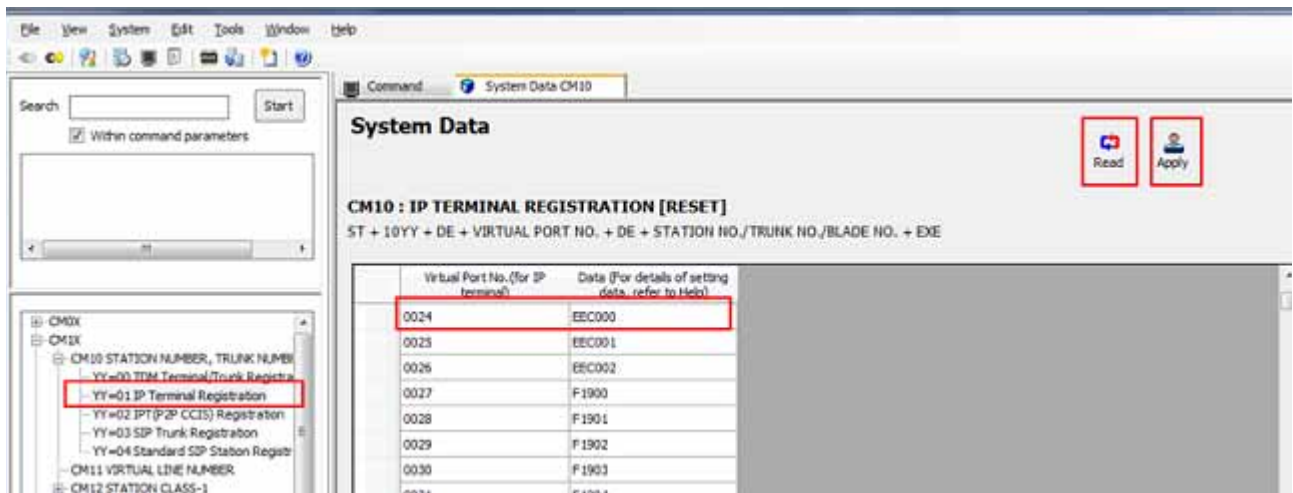
Figure 2-2 CM F88 STD SIP Station Capacity



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.





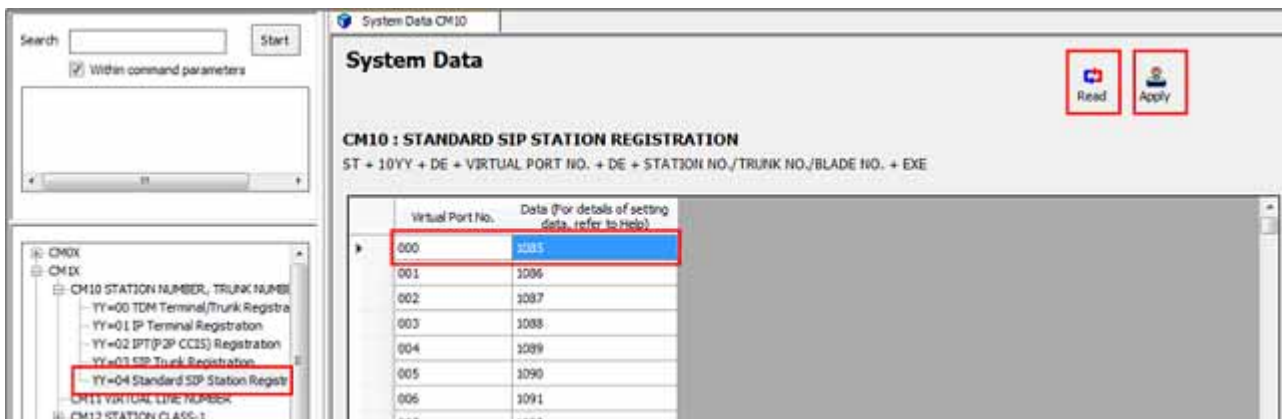
1. Assign the **SIP Converter No.** to the Virtual port number for Standard SIP station.
  2. Click **Read** to get the current settings.  
Example: **CM10 Y=01>0024>EEC000**
  3. Click **Apply**.
-  *SIP converter supports 3 Simultaneous calls by the Standard SIP stations. A system reset is required.*
-  *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

The screenshot shows the 'System Data CM1D' configuration window. The left pane displays a tree view of configuration options, with 'CM1D PS/STANDARD SIP STATION CON' selected. The main pane shows the configuration for 'CM1D : PS/STANDARD SIP STATION CONFIGURATION SETTINGS'. The 'PS/Standard SIP Station No.' is set to '1085'. The 'Terminal type' (14) is set to '0 : Standard SIP/PS (Less than 12 digits)'. The 'Primary/Subline' (16) is set to '1 (Def.) : Primary PS'. The 'Execute' button is visible at the bottom right.

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

The screenshot shows the 'System Data CM1D' configuration window. The left sidebar contains a tree view of configuration categories, with 'CM1D PS/STANDARD SIP STATION CON' highlighted. The main area is titled 'System Data' and contains the following configuration fields:

- System Data** (Title)
- CM1D : PS/STANDARD SIP STATION CONFIGURATION SETTINGS** (Section Header)
- ST + 1DYY + DE + PS/STANDARD SIP STATION No. + DE + SETTING DATA + EXE (Command Line)
- PS/Standard SIP Station No.  (Field)
- 01 : Subline PS No. to each Primary PS station (Label)
- Subline PS Station No.  (Field)
- 14 : Terminal type (Label)
- 0 : Standard SIP/PS (Less than 12 digits)  1(Def.) : Roaming PS (Radio Buttons)
- 05 : Standard SIP station (Label)
- 0 : Subline PS  1(Def.) : Primary PS (Radio Buttons)
- 15 : Terminal type [For PCS] (Label)
- 16 : Primary/Subline (Label)
- 20 : PS Operation Data Download (Only Display) [For PCS] (Label)
- 20 : PS Operation Data Download [For PCS] (Label)
- (Field)
- (Button)
- 21 : PS-ID [For PCS] (Label)
- PS-ID  (Maximum 9 digits, Decimal) (Field)
- 22 : PS Location Search with no ringing (Label)
- 0 : To provide when PS is idle  1(Def.) : Not provided when PS is idle (Radio Buttons)
- 32 : Standard SIP station Authentication (Label)
- (Field)

Buttons for 'Read' and 'Apply' are located in the top right corner of the configuration area.

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

System Data CM42

**System Data**

**CM42 : SYSTEM COUNTER DATA**  
ST + 42 + DE + KIND OF SYSTEM COUNTER + DE + SETTING DATA + EXE

66 : Transmission characteristic of analog LC [New Zealand/China/Brazil/Europe] / Transmission characteristic of analog LC, COT [For EMEA] [RESET] NONE(Def.) : Other countries except for the below / Depends on Nation Code (CM31.Y=0>0)

68 : Volume Control (Side tone level) of Multiline Terminal/DESKCON (Do not change this data normally, incorrect data settings may cause howler of low-level speech.) NONE(Def.) : -30 dB

69 : Call charge per unit for AOC (dollar/euro/integral charge per unit) [Australia/France/Germany/Netherlands/Italy/Greece/Luxembourg/Portugal/Spain/Sweden/ITU-T (JAE)] NONE(Def.) : No data dollars/euro/integral charge per unit

70 : Call charge per unit for AOC (cent/euro cent/two decimals charge per unit) [Australia/France/Germany/Netherlands/Italy/Greece/Luxembourg/Portugal/Spain/Sweden/ITU-T (JAE)] NONE(Def.) : No data cents/euro cents/two decimals charge per unit

72 : Number of times of Multiple Call Forwarding-All Calls/Busy Line/No Answer-CCIS NONE(Def.) : 5 time(s)

73 : Number of digits for Station Authorization Code/IP Station Password 04 : 04 digit(s)

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.

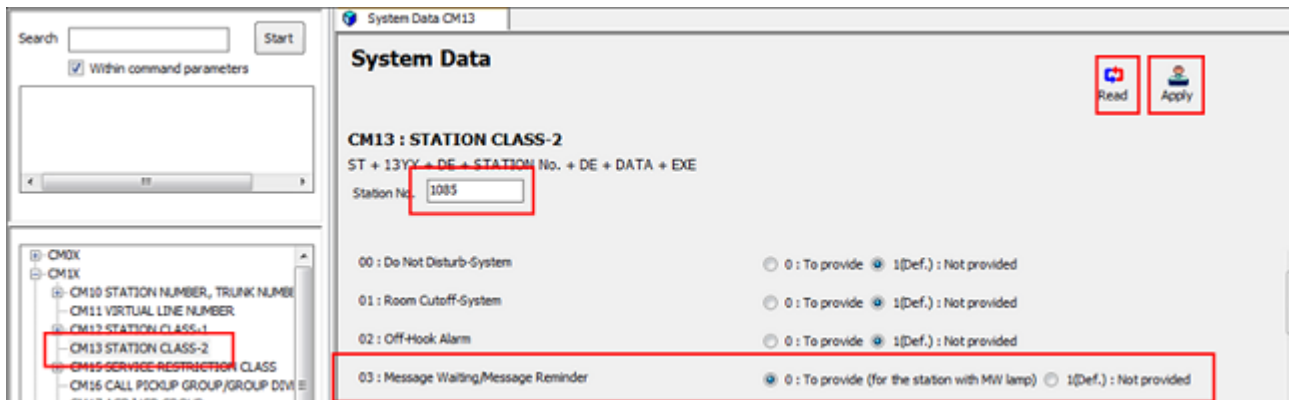
Figure 2-8 CM 2B00 Station Authorization Code to each Standard SIP station

The screenshot shows the 'System Data CM2B' configuration page. The main heading is 'System Data'. Below it, the title is 'CM2B : AUTHORIZATION CODE PER STATION'. The formula for the station number is 'ST + 2BYY + DE + 1ST DATA + DE + 2ND DATA + EXE'. The 'Station No.' is set to '2005'. The 'Authorization Code/Password' is set to '1234' (Maximum 8 digits). The 'Read' and 'Apply' buttons are highlighted with red boxes. The left sidebar shows a tree view of configuration options, with 'CM2B AUTHORIZATION CODE PER STA' selected. The right sidebar shows a list of restriction classes with their default values.

Restriction Class	Default Value
01 : Trunk Restriction Class	1(Def.) : Unrestricted (RCA)
02 : Service Restriction Class A	15(Def.) : 15
03 : Service Restriction Class B	15(Def.) : 15
04 : Service Restriction Class C	15(Def.) : 15
12 : Login password for User Web Portal	(Minimum 1 digit Maximum 16 digits)

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**.

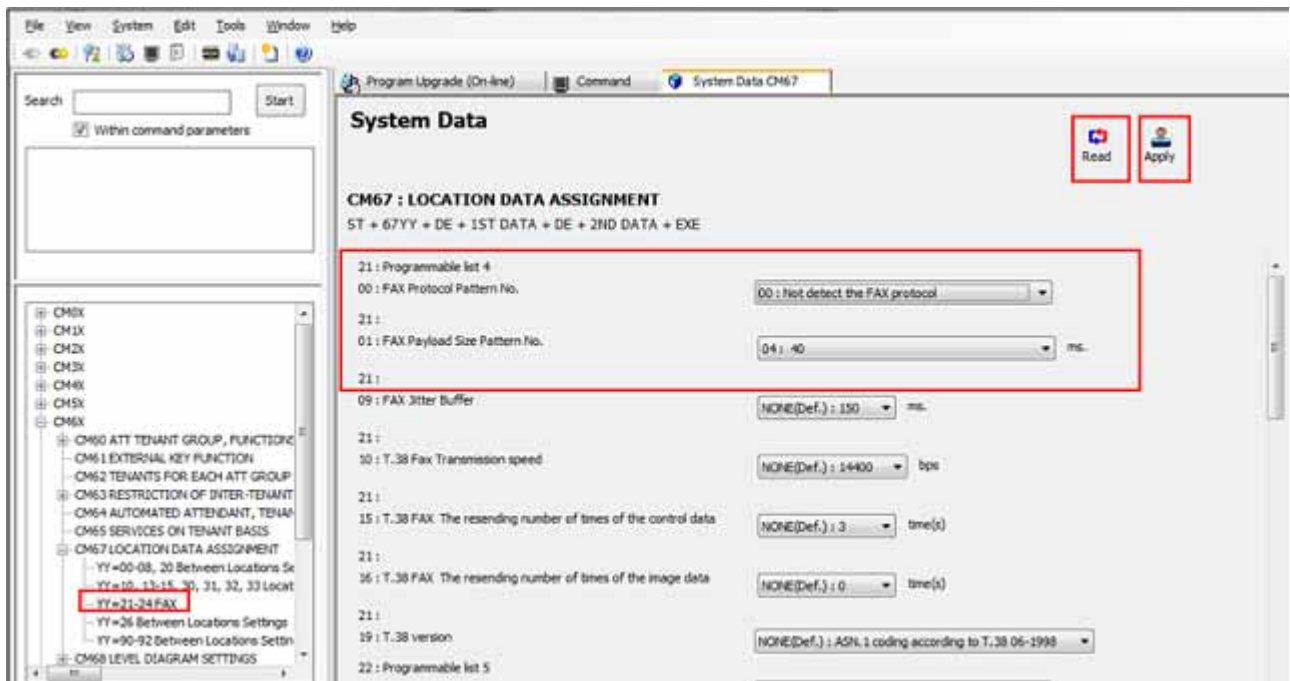
Figure 2-10 CM 67XX Assign Fax control information list

The screenshot shows the 'System Data' configuration page for CM67. The left sidebar contains a tree view of configuration items, with 'CM67 LOCATION DATA ASSIGNMENT' expanded. Under this, 'YY=00-08, 20 Between Locations Settings' is highlighted with a red box. The main area shows various settings for location data assignment, including precedence for control and voice packets, PAD data, jitter buffer, and DSCP settings. At the bottom, setting '20: FAX control information list to each location' is set to '4: Programmable list-4 (depends on the setting CM67 YY=21)', which is also highlighted with a red box. 'Read' and 'Apply' buttons are visible in the top right corner.

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)

The screenshot displays the 'IP Settings' configuration page. On the left is a navigation tree with 'Configuration' selected and 'IP Interfaces Table' highlighted. The main content area is titled 'IP Settings' and contains several sections:

- Single IP Settings:** A table with three rows: 'IP Address' (10.1.10.10), 'Subnet Mask' (255.255.0.0), and 'Default Gateway Address'. Each row has a checkmark icon on the right.
- VoIP DNS Settings:** A section with two rows: 'DNS Primary Server IP' and 'DNS Secondary Server IP'. The 'DNS Primary Server IP' row has a checkmark icon on the right.
- Multiple Interface Settings:** A section with one row: 'Multiple Interface Table'.

A 'Submit' button is located in the bottom right corner of the page.

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

The screenshot shows the 'Fax/Modem/CID Settings' page. The left sidebar contains a navigation tree with 'Fax/Modem/CID Settings' selected. The main content area is a table of settings:

Setting Name	Value
Fax Transport Mode	Events Only
Caller ID Transport Type	Mute
Caller ID Type	Standard Bellcore
V.21 Modem Transport Type	Disable
V.22 Modem Transport Type	Disable
V.23 Modem Transport Type	Disable
V.32 Modem Transport Type	Disable
V.34 Modem Transport Type	Disable
Fax CNG Mode	Sends on CNG tone
CNG Detector Mode	Events Only
<b>Fax Relay Settings</b>	
Fax Relay Redundancy Depth	0
Fax Relay Enhanced Redundancy Depth	4
Fax Relay ECH Enable	Disable
Fax Relay Max Rate (bps)	9600bps
<b>Bypass Settings</b>	
Fax/Modem Bypass Coder Type	G.711Mulaw
Fax/Modem Bypass Packing Factor	1
Fax Bypass Output Gain	0
Modem Bypass Output Gain	0

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

The screenshot shows the 'Proxy Sets Table' configuration page. The left sidebar contains a navigation tree with 'Proxy Sets Table' highlighted. The main content area features a 'Proxy Set ID' dropdown menu. Below it is a table with 5 rows for configuring proxy addresses and transport types. The first row is highlighted with a red box. Below the table is a settings section with several options and their default values. A 'Submit' button is located in the bottom right corner, also highlighted with a red box.

	Proxy Address	Transport Type
1	192.168.0.10	UDP
2		
3		
4		
5		

Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	30
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	No
Proxy Redundancy Mode	Not Configured

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

The screenshot displays the 'SIP General Parameters' configuration page. The left sidebar shows the navigation tree with 'SIP Definitions' expanded and 'General Parameters' selected. The main content area lists various parameters with their current values and dropdown menus. Three parameters are highlighted with red boxes: 'Channel Select Mode' is set to 'Ascending', 'SIP Destination Port' is set to '5070', and the 'Submit' button is located at the bottom right.

Parameter	Value
NAT IP Address	0.0.0.0
REACK Mode	Supported
Channel Select Mode	Ascending
Enable Early Media	Enable
Session Expires Time	0
Minimum Session Expires	90
Session Expires Method	re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	No Fax
SIP Transport Type	UDP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
SIP Destination Port	5070
Enable Multiple Party ID	Disable
Enable History-Info Header	Disable
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Prefer IP
Jax Behavior	Forward
Enable Reason Header	Enable

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 &amp; Definitions

The screenshot displays the 'Proxy & Registration' configuration page. The left sidebar shows the navigation tree with 'Advanced Parameters' selected. The main configuration area contains the following settings:

Parameter	Value
Use Default Proxy	Yes
Proxy Set Table	[+]
Proxy Name	[ ]
Redundancy Mode	Parking
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	Yes
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Enable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Enable
Registrar Name	SV9300
Registrar IP Address	192.168.0.10
Registrar Transport Type	UDP
Registration Time	180
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	5
Re-register On INVITE Failure	Disable
ReRegister On Connection Failure	Disable
Gateway Name	[ ]
Gateway Registration Name	[ ]

Buttons at the bottom: Register, Un-Register, Submit.

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 &amp; Definitions Continued

The screenshot shows the 'Proxy & Registration' configuration page. The left sidebar has 'Proxy & Registration' highlighted. The main area contains a list of parameters with their values:

Parameter	Value
Registrar IP Address	192.168.0.10
Registrar Transport Type	UDP
Registration Time	180
Re-registration Timing (%)	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable
Re-register On Connection Failure	Disable
Gateway Name	
Gateway Registration Name	
DNS Query Type	A Record
Proxy DNS Query Type	A Record
Subscription Mode	Per Endpoint
Number of KIX before Not-on-line	1
Use Gateway Name for OPTIONS	No
User Name	
Password	Default_Password
Challenge	Default_Challenge
Registration Mode	Per Endpoint
Set Out-of-Service on Registration Failure	Disable
Challenge Caching Mode	None

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

Coder Name	Packetization Time	Rate	Packet Type	Silence Suppression
G.729	20	8	10	Disabled
G.711ulaw	20	64	0	Disabled

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

The screenshot displays the 'Tel Profile Settings' configuration page. The left sidebar shows the navigation tree with 'Tel Profile Settings' selected under 'Coders and Profiles'. The main content area contains the following settings:

Parameter	Value
Profile ID	1
Profile Name	
<b>Profile Parameters</b>	
Profile Preference	1
Fax Signaling Method	No Fax
Enable Polarity Reversal	Disable
Enable Current Disconnect	Disable
<b>MWI Analog Lamp</b>	<b>Enable</b>
<b>MWI Display</b>	<b>Enable</b>
Auto Canceller	Enable
Flash Hook Period	700
Enable Early Media	Enable
Progress Indicator to IP	Not Configured
Enable FXO Double Answer	Disable
Dialing Mode	Two Stages
Disconnect Call on Detection of Busy Tone	Enable
Time For Reorder Tone [sec]	255
Enable 911 PSAP	Disable
Swap Tel To IP Phone Numbers	Disable
<b>Coder Group</b>	
Coder Group	Default Coder Group

A red box highlights the 'MWI Analog Lamp' and 'MWI Display' settings, and another red box highlights the 'Submit' button in the bottom right corner.

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

The screenshot displays the 'IP Profile Settings' configuration page. The left sidebar shows a navigation tree with 'IP Profile Settings' selected. The main content area is divided into sections: 'Basic Parameters' and 'Gateway Parameters'. The 'Gateway Parameters' section is expanded, showing a list of settings. The 'First Tx DTMF Option' and 'Second Tx DTMF Option' are highlighted with a red box, and their values are set to 'RFC 2833'. A 'Submit' button is located in the bottom right corner of the page.

Parameter	Value
Input Gain (-32 to 31 dB)(*)	0
Voice Volume (-32 to 31 dB)(*)	0
Symmetric MK2 Negotiation	Disable
MK2 Sca	0
Reset SRTP State Upon Re-key	Disable
<b>Gateway Parameters</b>	
Fax Signaling Method	No Fax
Play Ringback Tone to IP	Conf Play
Early Media	Enable
Copy Destination Number to Redirect Number	Disable
Media Security Behavior	Preferable
CNG Detector Mode	Disable
Modems Transport Type	Disable
NSE Mode	Disable
Number of Calls Limit	-1
Progress Indicator to IP	Not Configured
Profile Preference	1
Coder Group	Default Coder Group
Remote RTP Base UDP Port	0
First Tx DTMF Option	RFC 2833
Second Tx DTMF Option	RFC 2833
Disable RFC 2833 W SOP	True
Call Hold Service	Enable

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

Channel ID	Phone Number	Hunt Group ID	Tel Profile ID
1	1170	1	0
2	1171	1	0
3			
4			

Buttons: Register, Un Register, Submit

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

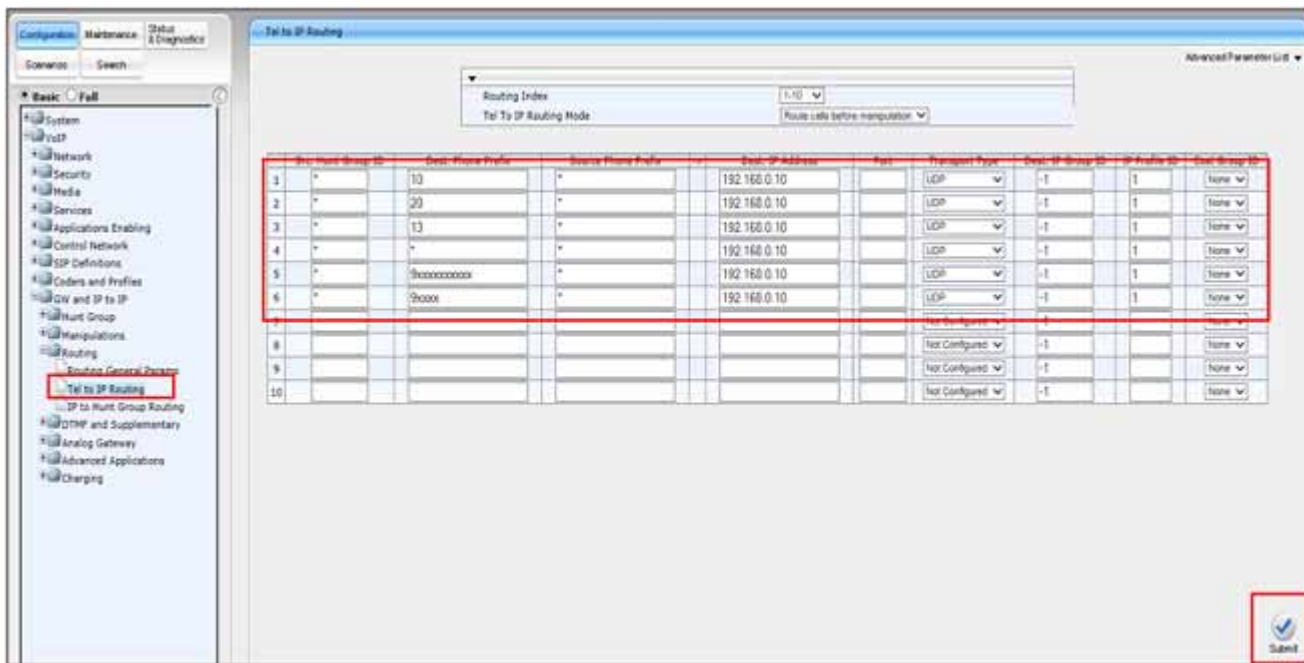
The screenshot displays the 'Hunt Group Settings' page. On the left is a navigation tree with 'Hunt Group Settings' selected. The main area contains a table with the following data:

Hunt Group ID	Channel Select Mode	Registration Mode	Service IP Group ID	Release Time	Contact User
1	By Dest Phone Number	Per Endpoint			
2					
3					
4					
5					
6					
7					
8					
9					
10					
11					
12					

At the bottom of the page, there are 'Register' and 'Un-Register' buttons, and a 'Submit' button in the bottom right corner.

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

The screenshot shows the 'IP To Hunt Group Routing Table' configuration page. The left-hand navigation menu has 'IP to Hunt Group Routing' highlighted with a red box. The main area contains a table with the following columns: Line, Hunt Group, Source IP Address, and Hunt Group ID. The table is currently empty. A red box highlights the 'Submit' button in the bottom right corner.

Line	Hunt Group	Source IP Address	Hunt Group ID
1			
2			
3			
4			
5			
6			
7			
8			
9			
10			
11			
12			


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 2-27 DTMF Supplementary

Parameter	Value
Enable Hold	Enable
Hold Format	0.0.0.0
Hold Timeout	1
Call Hold Reminder Ring Timeout	30
Enable Transfer	Enable
Transfer Prefix	
Enable Call Forward	Enable
Enable Call Waiting	Enable
Number of Call Waiting Indications	2
Time Between Call Waiting Indications	10
Time Before Waiting Indications	0
Waiting Beep Duration	300
Enable Caller ID	Enable
Hook-Flash Code	
Flash Keys Sequence Style	Sequence 2
Flash Keys Sequence Timeout	2000
Enable NRT Subscription	Disable
AS Subscribe IPGroupID	1
NRT Subscribe Retry Time	120
Call Forward Ring Tone ID	1
Send All Coders on Retrieve	Disable

Buttons: Submit, Subscribe to MWI, Unsubscribe to MWI

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

The screenshot shows the configuration interface for Supplementary Services. The left sidebar has 'DTMF and Supplementary' expanded, with 'Supplementary Services' selected. The main area shows the following parameters:

Parameter	Value
Flash Keys Sequence Style	Sequence 2
Flash Keys Sequence Timeout	2000
Enable NRT Subscription	Disable
AS Subscribe IPGroupID	-1
NRT Subscribe Retry Time	120
Call Forward Ring Tone ID	1
Send All Coders on Retrieve	Disable
<b>Message Waiting Indication (MWI) Parameters</b>	
Enable MWI	Enable
MWI Analog Lamp	Enable
MWI Display	Enable
Subscribe to MWI	Yes
MWI Server IP Address	192.168.0.10
MWI Server Transport Type	UDP
MWI Subscribe Expiration Time	1200
Stutter Tone Duration	2000
MWI Subscribe Retry Time	120
<b>MLPP</b>	
Call Priority Mode	Disable
Reminder Ring	Enable
MLPP Offserv	00
Precedence Ringing Type	-1

Buttons at the bottom: Submit, Subscribe to MWI, Unsubscribe to MWI.

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

The screenshot displays the 'Supplementary Services' configuration page. The left sidebar shows the navigation menu with 'Supplementary Services' selected. The main content area is divided into sections: 'MPP', 'Conference', and 'Transfer'. The 'Conference' section is expanded, showing the following settings:

Parameter	Value
Enable 3-Way Conference	Enable
Establish Conference Code	1
Conference ID	conf
3-Way Conference Mode	On Board
Max. 3-Way Conference	2
Non Allocatable Ports	0

At the bottom of the page, there are three buttons: 'Submit', 'Subscribe to MWI', and 'Unsubscribe to MWI'. The 'Submit' button is highlighted with a red box.

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

Setting	Value
Dialing Mode	Two Stages
Waiting for Dial Tone	Yes
Time to Wait before Dialing (msec)	1000
Ring Detection Timeout (sec)	8
Reorder Tone Duration (sec)	255
Answer Supervision	No
Rings before Detecting Caller ID	1
Send Metering Message to IP	No
Disconnect Call on Busy Tone Detection (CAS)	Enable
Disconnect on Dial Tone	Disable
Guard Time Between Calls	1
FXO Double Answer	Disable
FXO AutoDial Play BusyTone	Disable
FXO Ring Timeout [100 msec]	0

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

Gateway Port	User Name	Password
Port 1 FXS	1170	*****
Port 2 FXS	1171	*****
Port 3 FXO	1172	*****
Port 4 FXO	1173	*****

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.*

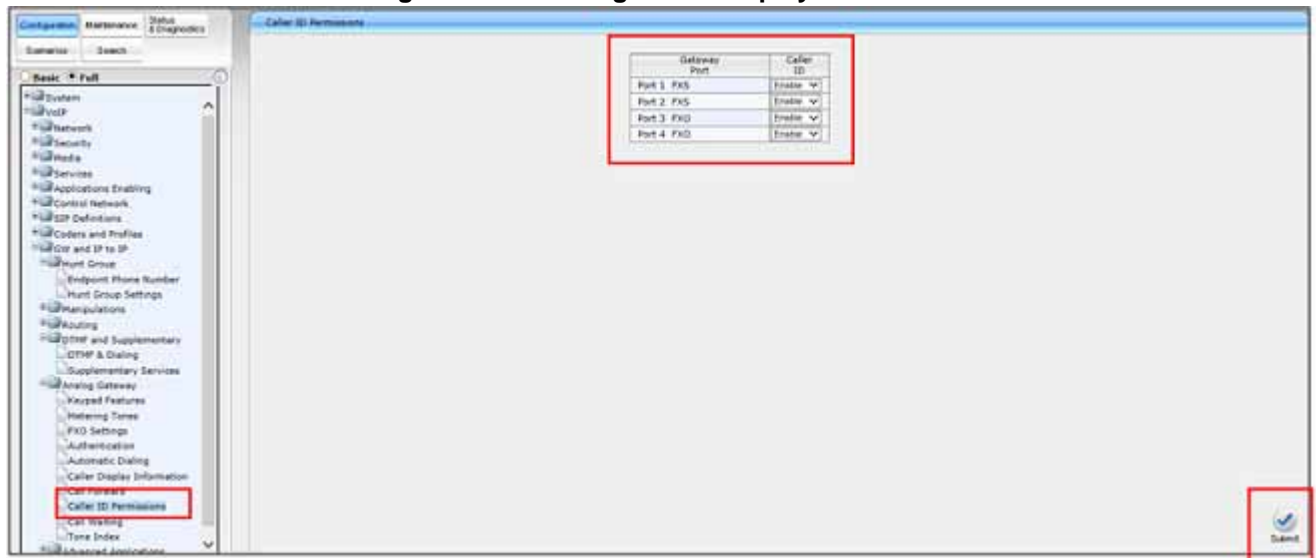
Figure 2-32 Analog Automatic Dialing

Gateway Port	Destination Phone Number	Auto Dial Status	Hotline Dial Tone Duration (sec)
Port 1: FXS		Enable ▼	0
Port 2: FXS		Enable ▼	0
Port 3: FXO	1170	Enable ▼	0
Port 4: FXO		Enable ▼	0

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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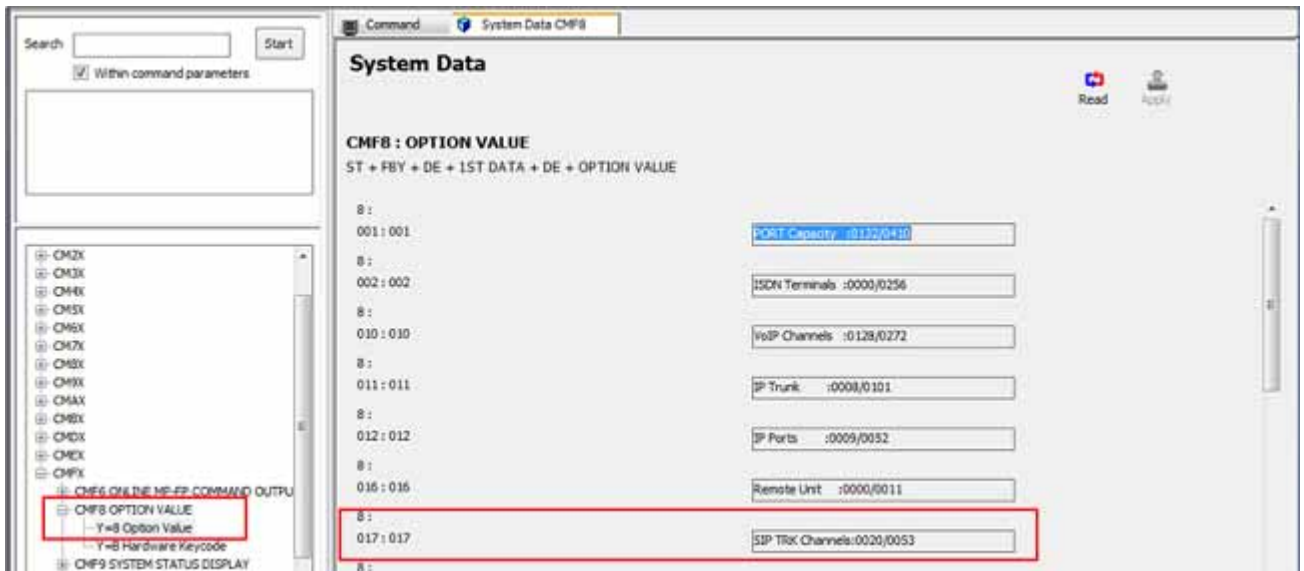


## Chapter 3 *Audio Codes for Trunk Configuration*

### 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.

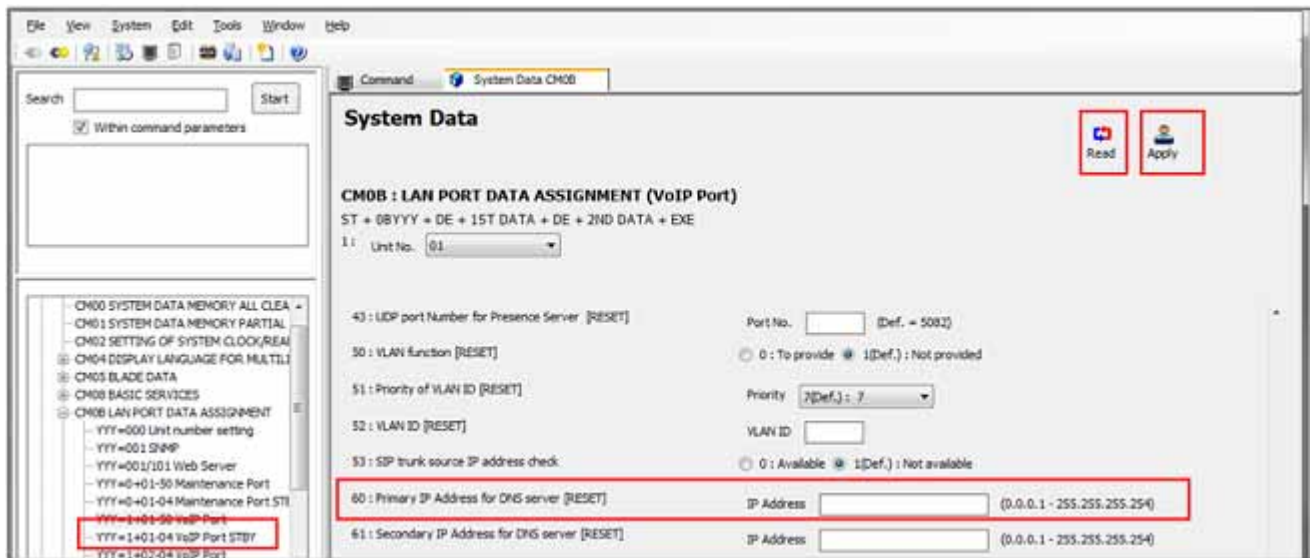
Figure 3-2 CM 0B1XX Network Configuration

The screenshot shows the 'System Data' configuration window for CM0B. The left sidebar lists various system data categories, with 'CM0B LAN PORT DATA ASSIGNMENT' expanded. Under this category, 'YYY=01-50 Maintenance Port' is highlighted. The main window displays the following configuration details:

Code	Description	Current Value	Range
00	IP Address for the system [RESET]	192.168.0.10	(0.0.0.1 - 255.255.255.254)
01	Subnet Mask for the system [RESET]	255.255.255.0	(255.0.0.0 - 255.255.255.252)
02	Default Gateway for the system [RESET]	192.168.0.1	(0.0.0.1 - 255.255.255.254)
09	Speed mode for the LAN Interface [RESET]	15(Def.) : Auto Negotiation (GbE)	
10	Location No. for stations and VoIPDB accommodated in the Unit (Available when location number is not assigned by CM12 YY=39, 50.)	NONE(Def.) : 00	
11	Tenant No. for IP stations accommodated in the Unit	NONE(Def.) : 01	
20	Whether to allow the connection with PCPro [RESET]	<input type="radio"/> 0 : Restricted <input checked="" type="radio"/> 1(Def.) : Allow	

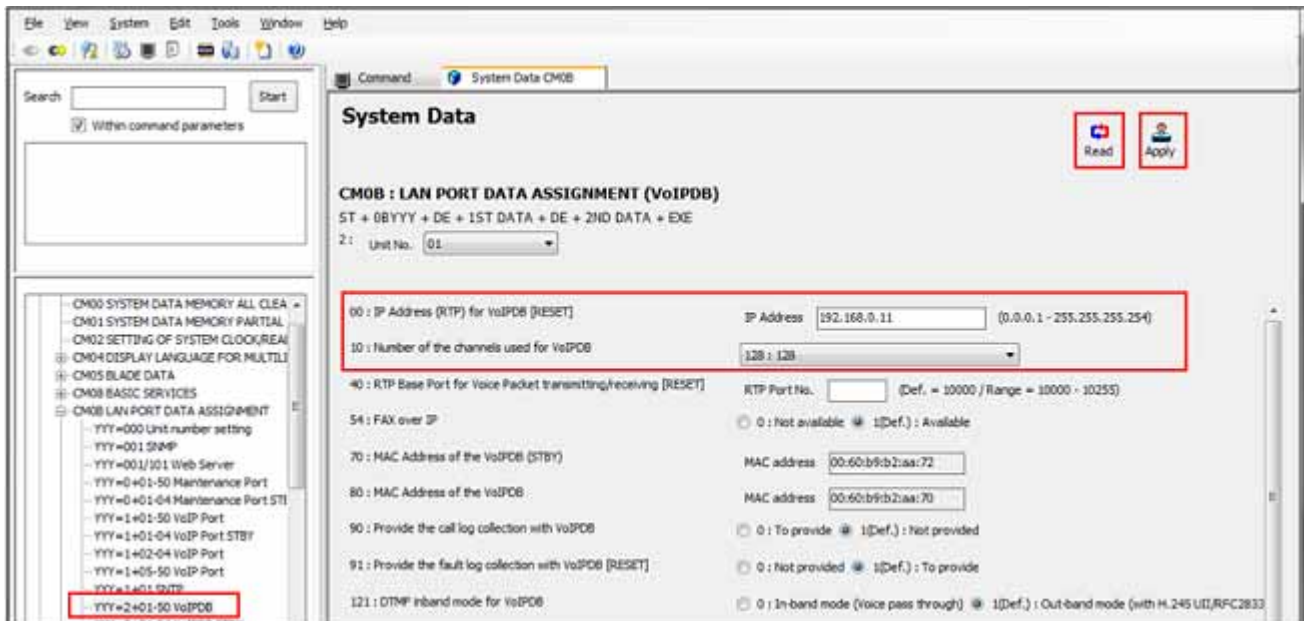
1. Select **SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels**, then click **Read** to get the current data settings.
2. Assign the SV9300 **VoIPDB IP Address**.  
Example: **0B101>00>192.168.0.10**
3. Assign the SV9300 **VoIPDB Subnet Mask**.  
Example: **0B101>01>255.255.255.0**
4. Assign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



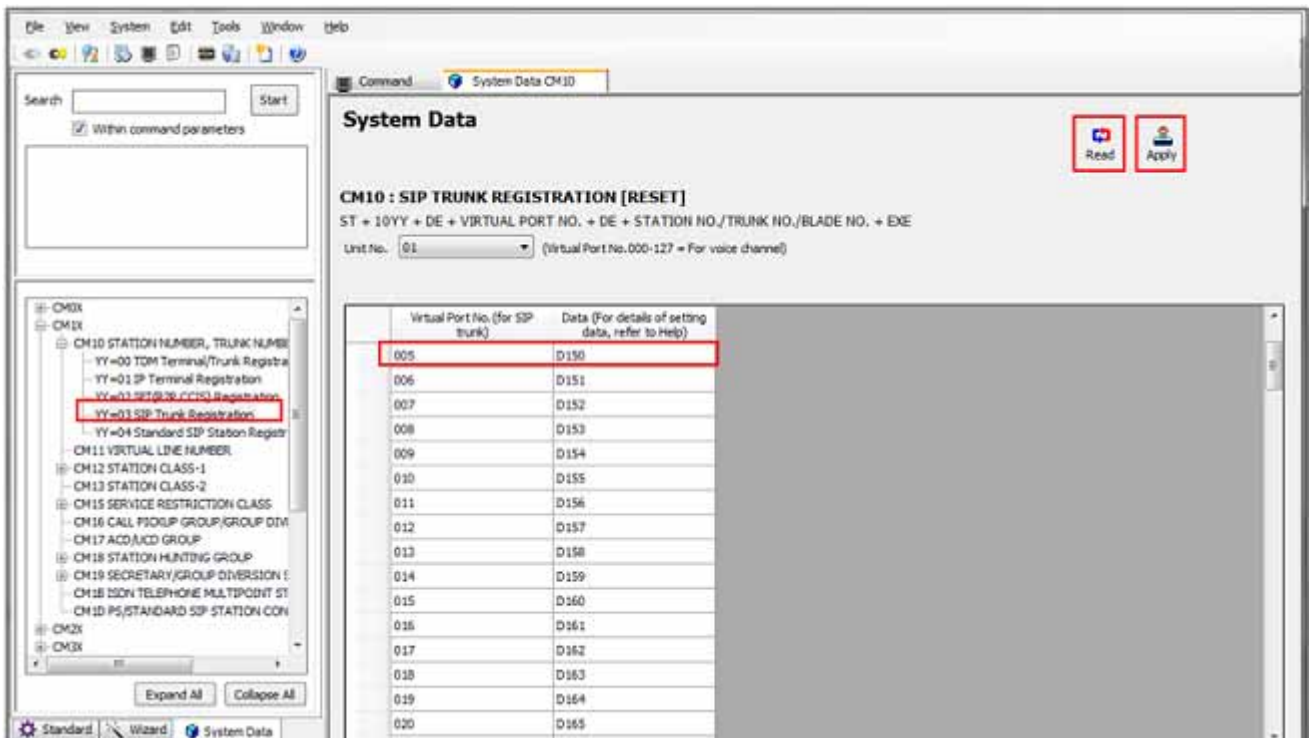
1. Select **SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels**, then click **Read** to get the current data settings.
2. Assign the primary DNS server IP Address (optional).  
Example: **0B101>60>IP address of DNS server**
3. Click **Apply**.

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

Search  Start

Within command parameters

System Data

CM30 : TRUNK DATA

ST + 30XX + DE + Trunk No. + DE DATA + EXE

Trunk No. 150

00 : Trunk route allocation [RESET][BLADE RESET][IP TRUNK RESET] Trunk Route No. 30 : 30

01 : Allocation of tenants to trunks Tenant No. 01(Def.) : 01

02 : Terminating system in Day Mode for incoming C.O. calls 31(Def.) : DID, Tie Line and the call which is not handled by the PBX

Read Apply

CM30 TRUNK DATA

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

The screenshot displays the configuration interface for CM 30XX SIP Trunk Port Settings. On the left, a search bar is present with a 'Start' button and a checkbox for 'Within command parameters'. Below the search bar is a list of voice channels, with 'CM30 TRUNK DATA' highlighted in red. The main configuration area is titled 'System Data CM30' and contains the following settings:

- CM30 : TRUNK DATA**
- ST + 30YY + DE + TRUNK No. + DE + DATA + EXE
- Trunk No. 150
- Access to System (DISA) destination in Day Mode: 15(Def.) : C.O. line release
- 31 : Handling of busy/hot available Automated Attendant/Remote Access to System (DISA) destination in Night Mode: 15(Def.) : C.O. line release
- 32 : Handling of timed-out Automated Attendant call in Day Mode: 15(Def.) : C.O. line release
- 33 : Automated Attendant Handling of all PBR busy when 2nd announcement and DT are connected: 15(Def.) : C.O. line release
- 34 : ISDN Local Office Code Table No. Local Office Code Table No. 15(Def.) : Not assigned
- 35 : CIC (Circuit Identification Code) used for No. 7 CCIS/SIP voice channels [RESET]: CIC 001 : 001
- 37 : Handling of timed-out Automated Attendant call in Night Mode: 15(Def.) : C.O. line release

The 'Read' and 'Apply' buttons are highlighted in red in the top right corner of the configuration panel.

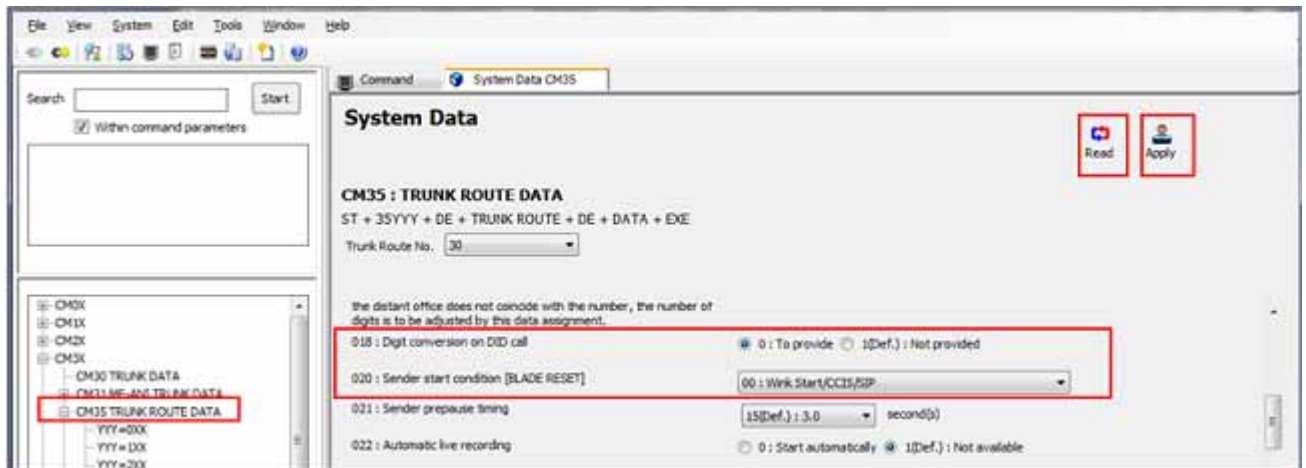
1. Select each **Voice Channel** and click the **Read** button to get the current data settings.
2. Assign a CIC number to each voice trunk.  
Example: **CM3035>150>001**
3. Click **Apply**.

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

The screenshot shows the 'System Data' configuration page for 'System Data CM35'. The main heading is 'System Data' with 'Read' and 'Apply' buttons. Below this is the section 'CM35 : TRUNK ROUTE DATA' with the command 'ST + 35YYY + DE + TRUNK ROUTE + DE + DATA + EXE'. The 'Trunk Route No.' is set to '30'. The configuration area contains several options with radio buttons and dropdown menus:

- 079: Terminal connection form for ISDN Basic Rate Interface [BLADE RESET]
  - 0: Point-to-Point (selected)
  - 1(Def.): Point-to-Multipoint
- 083: Trunk seizure sequence for an outgoing call
  - 0: As per CM08>078
  - 1(Def.): By allotter
- 086: Centrex trunk
  - 0: To provide
  - 1(Def.): Not provided
- 087: Distinctive Ringing by detecting the ringing signal from main PBX or Centrex
  - 0: To provide
  - 1(Def.): Not provided
- 089: Cyclic Redundancy checking for DTI trunk \*NOTE 1: This command is not effective when you use the DTI(2M E1) Trunk(CM05 Y=0 2nd=47), set CMAA YY=01. [BLADE RESET]
  - 0: To provide
  - 1(Def.): Not provided
- 090: Special facilities
  - 0: No. 7 CCIS, SIP trunk (selected)
- 091: Common Channel Handler (CCH) No. used for No. 7 CCIS/SIP
  - CCH: 02: 02 (selected)

The left sidebar shows a tree view with 'CM35 TRUNK ROUTE DATA' highlighted in red. The top left has a search bar and a 'Start' button.

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

The screenshot shows the 'System Data CM36' configuration page. The main title is 'System Data'. Below it, the section is 'CM36 : RESTRICTION DATA/PAD DATA FOR TANDEM CONNECTION'. The formula shown is 'ST + 36Y + DE + INCOMING TRUNK ROUTE + OUTGOING TRUNK ROUTE + DE + DATA + EXE'. There are two dropdown menus for 'INCOMING TRUNK ROUTE' and 'OUTGOING TRUNK ROUTE', both set to '30'. A red box highlights the radio button options: '0 : Allow' (selected) and '1(Def.) : Restricted'. Below this, there are two more dropdown menus for 'PAD data from an incoming trunk route to an outgoing trunk route' and 'PAD data from an outgoing trunk route to an incoming trunk route', both set to 'NONE(Def.) : 0'. On the left side, a tree view shows the configuration hierarchy, with 'CM36 RESTRICTION DATA/PAD DATA FOR' selected and highlighted with a red box. At the top right, there are 'Read' and 'Apply' buttons, also highlighted with red boxes.

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

The screenshot displays the 'System Data CMA7' configuration interface. The left sidebar shows a tree view of configuration categories, with 'CMA7 CCIS Channel/IP Trunk/SIP Trunk' selected and highlighted in red. The main content area shows the configuration for 'CMA7 : CCIS Channel/IP Trunk/SIP Trunk Data'. The 'COMMON CHANNEL SIGNALING HANDLE' is set to '02'. Several parameters are highlighted with red boxes:

- 71: SIP Profile No. for control packet**: Profile No. for control packet is set to '01: 01'.
- 78: Unit of SIP Trunk Number [RESET]**: Unit is set to '01: 01'.

Other visible parameters include:

- 44: TOS field Precedence for IPT (P2P CCIS) control packet, TOS : Type of Service (Ineffective when CMA7 YY=50 is set) [IP TRUNK RESET] PRECEDENCE: 15(Def.): 00
- 45: Release timer for IPT (P2P CCIS) Point-to-Multipoint connection NONE(Def.): Not released minute(s) [ '000' : seconds]
- 46: Connection Method (P-P CCIS/SIP Trunk) [RESET][IP TRUNK RESET] 0: Point-to-Multipoint 1(Def.): Point-to-Point
- 50: DiffServ code setting for IPT (P2P CCIS) control packet [IP TRUNK RESET] DiffServ code point: (HEX, 00-3F)
- 64: Trunk seizure sequence for incoming calls 0: By Allotter 1(Def.): From lowest circuit no. available
- 77: Registration Status/Manual Registration Sending 0(Def.): Not Registered 1: Registration Complete \* Applying 0 in 2nd data, REGISTER

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

Search  Start

Within command parameters

System Data CM A8

**System Data** Read Apply

**CM A8 : CCIS Channel/IP Trunk/SIP Trunk Data2**  
ST+ A8Y + DE + 1ST DATA + DE + 2ND DATA + EXE

Destination Point Code (DPC) sent from distant office assigned by CMA7 YY=02

0 :  CCH/IP trunk/SIP trunk No.

1 :  IP Address  (aaabbbcccddd as input style)

2 : [SV9300 V4] LIN (Location Identification Number) Index  (Def. = 0000)

CM A8 : CCIS Channel/IP Trunk/SIP Trunk

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

**System Data**

**CMBA : SIP PROFILE DATA**  
ST + BAYY/YYYY + DE + PROFILE NO. + DE + 2ND DATA + EXE  
Profile No. for control packet: 01

04 : TOS (Type of Service)/Diffserv Precedence for SIP trunk control packet [RESET] PRECEDENCE: 15(Def.) : 0  
TOS,Diffserv Precedence: (HEX, 00-3F)

10 : TOS (Type of Service)/Diffserv Precedence for SIP trunk control packet [RESET] NONE(Def.) : 150 ms

13 : FAX Jitter buffer 7(Def.) : Standard Mode 1

Data	Mode	HIGH ← SELECTION PRECEDENCE → LOW
1	Standard Mode 2	Q.711 (G.711)
5	Tone Quality Mode	Q.723a (G.723a)
6	Band Mode	Q.723a (G.723a)
7	Standard Mode 1	Q.711 (G.711)

21 : Voice encoding selection precedence for SIP trunk 3(Def.) : 40ms.

22 : Payload size for SIP trunk 3(Def.) : 40ms.

25 : Query a DNS server to get the IP Address [RESET]  0 : Provide  1(Def.) : Not provided

27 : Provide session refreshment when receiving 18X. Provisional response is received again  0 : Not provided  1(Def.) : Provide

29 : Session Timer refresher kind  0 : uas  1(Def.) : uac

30 : SIP server IP Address [RESET] SIP server IP Address: 192168000200 (aaaaabbbccddd as input style)

31 : SIP server Port No. [RESET] SIP server Port number: 05060 (NONE=05060)

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>192168000200**
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)

The screenshot displays the 'IP Settings' configuration page. The left sidebar shows a tree view with 'Network' expanded and 'IP Interfaces Table' selected. The main content area is titled 'IP Settings' and contains several sections:

- Single IP Settings:** A table with three rows:
 

IP Address	10.1.10.10	<input checked="" type="checkbox"/>
Subnet Mask	255.255.0.0	<input checked="" type="checkbox"/>
Default Gateway Address		<input checked="" type="checkbox"/>
- VVoIP DNS Settings:** A section with two rows:
 

DNS Primary Server IP		<input checked="" type="checkbox"/>
DNS Secondary Server IP		<input type="checkbox"/>
- Multiple Interface Settings:** A section with one row:
 

Multiple Interface Table	<input type="checkbox"/>
--------------------------	--------------------------

A 'Submit' button is located in the bottom right corner of the main content area.

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

The screenshot displays the IP Routing Table configuration page. The table contains the following data:

Delete Row	Destination IP Address	Prefix Length	Gateway IP Address	Metric	Interface Name	Status
<input type="checkbox"/>	127.0.0.0	0	127.0.0.1	1		Active
<input type="checkbox"/>	127.0.0.1	32	127.0.0.1	0		Active
<input type="checkbox"/>	192.168.0.0	24	192.168.0.200	0		Active

Below the table is a "Delete Selected Entries" button. Below that is a form to "Add a new table entry" with the following fields:

Destination IP Address	Prefix Length	Gateway IP Address	Metric	Interface Name
	16		1	

An "Add New Entry" button is located below the form.

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

The screenshot shows the 'Fax/Modem/CID Settings' page. The left sidebar has 'Fax/Modem/CID Settings' selected. The main content area is divided into three sections:

- General Settings:**
  - Fax Transport Mode: Events Only
  - Caller ID Transport Type: Mute
  - Caller ID Type: Standard Bellcore
  - V.21 Modem Transport Type: Disable
  - V.22 Modem Transport Type: Disable
  - V.23 Modem Transport Type: Disable
  - V.32 Modem Transport Type: Disable
  - V.34 Modem Transport Type: Disable
  - Fax CNG Mode: Sends on CNG tone
  - CNG Detector Mode: Events Only
- Fax Relay Settings:**
  - Fax Relay Redundancy Depth: 0
  - Fax Relay Enhanced Redundancy Depth: 4
  - Fax Relay ECH Enable: Disable
  - Fax Relay Max Rate (bps): 9600bps
- Bypass Settings:**
  - Fax/Modem Bypass Codec Type: G711Mulu
  - Fax/Modem Bypass Packing Factor: 1
  - Fax Bypass Output Gain: 0
  - Modem Bypass Output Gain: 0

A 'Submit' button is located in the bottom right corner of the 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 3-19 Proxy Sets Table

The screenshot shows the 'Proxy Sets Table' configuration page. On the left is a navigation tree with 'Proxy Sets Table' selected. The main area contains a 'Proxy Set ID' dropdown set to '0'. Below it is a table with columns 'Proxy Address' and 'Transport Type'. The first row is highlighted with a red box and contains the value '192.168.0.10' and 'UDP'. Below the table is a settings section with several options and dropdown menus. A red box highlights the 'Submit' button in the bottom right corner.

	Proxy Address	Transport Type
1	192.168.0.10	UDP
2		
3		
4		
5		

Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	50
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	No
Proxy Redundancy Mode	Not Configured

Submit

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. Click **Submit**.

Figure 3-20 SIP Definitions General Parameters

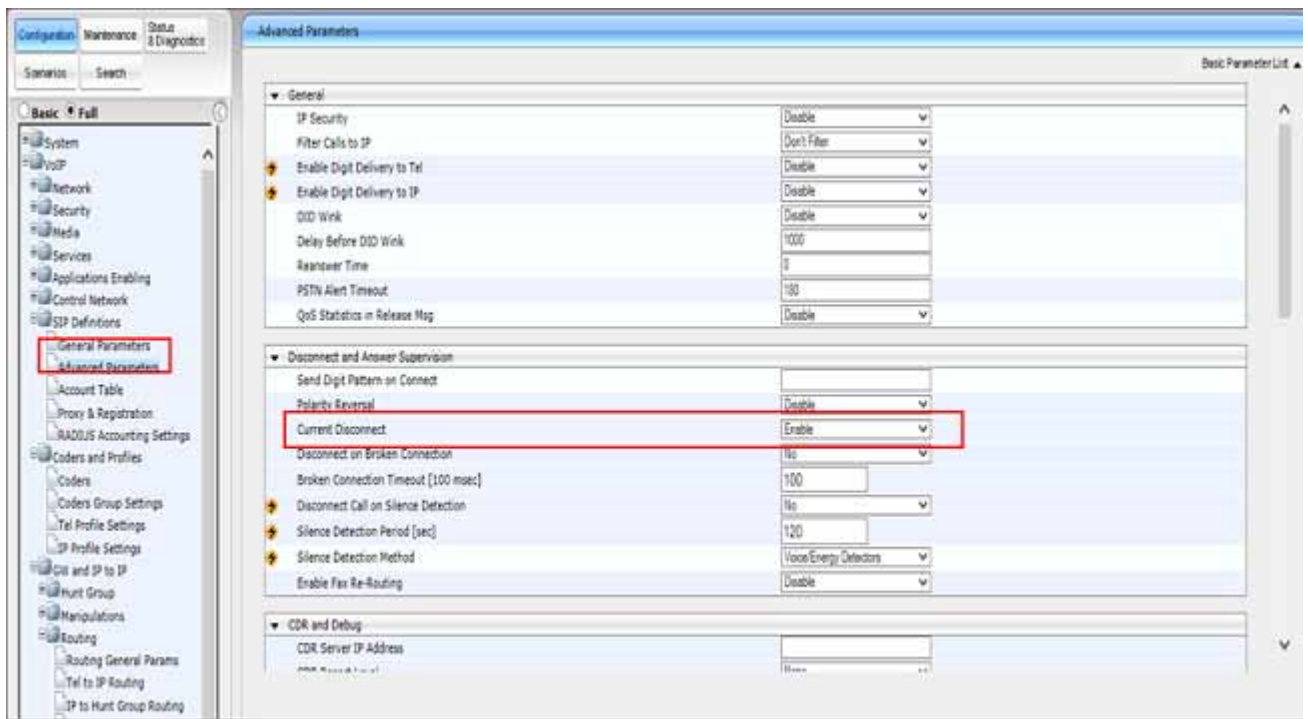
The screenshot shows the 'SIP General Parameters' configuration page. The left sidebar has 'SIP Definitions' expanded to 'General Parameters'. The main area contains a list of parameters with their current values:

NAT IP Address	0.0.0.0
BRACK Mode	Supported
Channel Select Mode	By Dest. Phone Number
Enable Early Media	Enable
Session Expires Time	0
Minimum Session Expires	30
Session Expires Method	re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	No-Fax
SIP Transport Type	UDP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Enable
Enable TCP Connection Reuse	Enable
SIP Destination Port	5060
Enable Remote Party ID	Disable
Enable History-Info Header	Disable
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Play IP
Jax Behavior	Forward
Enable Reason Header	Enable

At the bottom right, there is a 'Submit' button.

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

The screenshot displays the configuration interface for SIP Definitions Advanced Parameters. The left sidebar shows a tree view with 'Advanced Parameters' selected. The main area shows a list of parameters with their current values:

Parameter	Value
Enable Fax Re-Routing	Disable
CDR and Debug	
CDR Server IP Address	
CDR Report Level	None
Media CDR Report Level	None
Misc. Parameters	
Progress Indicator to IP	PI=1
Enable Busy Out	Disable
Graceful Busy Out Timeout [sec]	0
Default Release Cause	3
Max Number of Active Calls	16
Max Call Duration [min]	0
LAN Watchdog	Disable
Enable Calls Out Through	Disable
Enable User-Information Usage	Disable
Out Of Service Behavior	Flourish Tone
Delay After Reset [sec]	7
T.38 Fax Max Buffer	1024
Enable Microsoft Extension	Disable
Reliable Connection Persistent Mode	Disable
First Call Ringback Tone ID	-1
Call Pickup Key	
Enable Delayed Offer	Disable

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

The screenshot shows the 'Advanced Parameters' configuration page. The left sidebar has a tree view with 'Advanced Parameters' highlighted. The main area contains a table of parameters:

Parameter Name	Value
T.38 Fax Max Buffer	1024
Enable Microsoft Extension	Disable
Reliable Connection Persistent Mode	Disable
First Call Ringback Tone ID	-1
Call Pickup Key	
Enable Delayed Offer	Disable
Replace Number Sign With Escape	Disable
AMD Beep Detection Mode	Disabled
Source Header For Called Number	use Request-URI header
Add Empty Authorization Header	Disable
SP2IP Transfer Mode	Disable
ENUM Resolution	enable
T.38 Fax Session	Disable
SP2IP Registration Time	30
Tel2IP Call Forking Mode	Disable
SIP Remote Reset	Disable
Transparent Codec On Data Call	Disable
Enforce Media Order	Disable
Enable Rekey After 181	Disable

Below the table is an 'Emergency Calls' section with the following parameters:

Parameter Name	Value
Emergency Numbers	
[min] Emergency Calls Regret Timeout	10
Emergency Special Release Cause	Disable

A 'Submit' button is located in the bottom right corner of the configuration area.

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

Parameter	Value
Use Default Proxy	No
Proxy Name	
Redundancy Mode	Porting
Proxy IP List Refresh Time	60
Enable fallback to Routing Table	Disable
Prefer Routing Table	No
Always Use Proxy	Enable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Enable
Registrar Name	192.168.0.10
Registrar IP Address	192.168.0.10
Registrar Transport Type	UDP
Registration Time	180
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable
ReRegister On Connection Failure	Disable
Gateway Name	
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record

Buttons: Register, Use Register, Submit

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

Coder Name	Packetization Time	Rate	Packet Type	Silence Suppression
G.729	20	8	18	Disabled
G.711u-ls	20	64	0	Disabled

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

The screenshot displays the 'IP Profile Settings' configuration page. The left sidebar shows the navigation tree with 'IP Profile Settings' selected. The main content area is divided into sections: 'Common Parameters' and 'Gateway Parameters'. The 'Common Parameters' section includes fields for 'Profile ID', 'Profile Name', and various codec-related settings. The 'Gateway Parameters' section includes settings for 'Fax Signaling Method', 'Play Back Time to IP', 'Early Media', 'Copy Destination Number to Redirect Number', and 'Media Security Behavior'. Red boxes highlight the 'Disconnect on Broken Connection' dropdown (set to 'Yes'), the 'Early Media' dropdown (set to 'Enable'), and the 'Submit' button.

Section	Parameter	Value
Common Parameters	Profile ID	1
	Profile Name	
	STP IP DiffServ	45
	Signaling DiffServ	45
	Disconnect on Broken Connection	Yes
	Dynamic Jitter Buffer Minimum Delay (ms)	10
	Dynamic Jitter Buffer Optimization Factor(*)	10
	STP Redundancy Depth(*)	0
	Echo Canceler(*)	Enable
	Input Gain (-32 to 31 dB)(*)	0
	Voice Volume (-32 to 31 dB)(*)	0
	Symmetric MKI Negotiation	Disable
	MKI Size	30
	Reset SRTP State Upon Re-key	Disable
Gateway Parameters	Fax Signaling Method	No Fax
	Play Back Time to IP	Cost Play
	Early Media	Enable
	Copy Destination Number to Redirect Number	Disable
	Media Security Behavior	Preferable

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

The screenshot displays the 'IP Profile Settings' configuration page. The left sidebar shows the navigation tree with 'IP Profile Settings' selected. The main content area contains the following settings:

Input Gain (-32 to 31 dB(*)	0
Voice Volume (-32 to 31 dB(*)	0
Symmetric HQ Negotiation	Disable
MXI Size	0
Reset SRTP State Upon Re-key	Disable
<b>Gateway Parameters</b>	
Fax Signaling Method	No Fax
Play Ringback Tone to IP	Don't Play
Early Media	Enable
Copy Destination Number to Redirect Number	Disable
Media Security Behavior	Preferable
<b>CNG Detector Mode</b>	<b>Events Only</b>
Modems Transport Type	Disable
NSE Mode	Disable
Number of Calls Limit	-1
<b>Progress Indicator to IP</b>	<b>PI = 1</b>
Profile Preference	1
Coder Group	Default Coder Group
Remote RTP Base UDP Port	0
<b>First Tx DTMF Option</b>	<b>RFC 2833</b>
Second Tx DTMF Option	
Declare RFC 2833 in SDP	Yes
Call Hold Service	Enable

A 'Submit' button is located in the bottom right corner of the configuration area.

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

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1				0
2				0
3	3	3	02	0
4	4	4	02	0

Register Use Register Submit

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

The screenshot shows the 'Hunt Group Settings' page. On the left is a navigation tree with 'Hunt Group Settings' selected. The main area contains a table with the following data:

	Hunt Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID	Gateway Name	Contact User
1						
2	2	Ascending	Don't Register		192.168.0.200	192.168.0.200
3						
4						
5						
6						
7						
8						
9						
10						
11						
12						

At the bottom of the page are 'Register' and 'Un-Register' buttons, and a 'Submit' button in the bottom right corner.


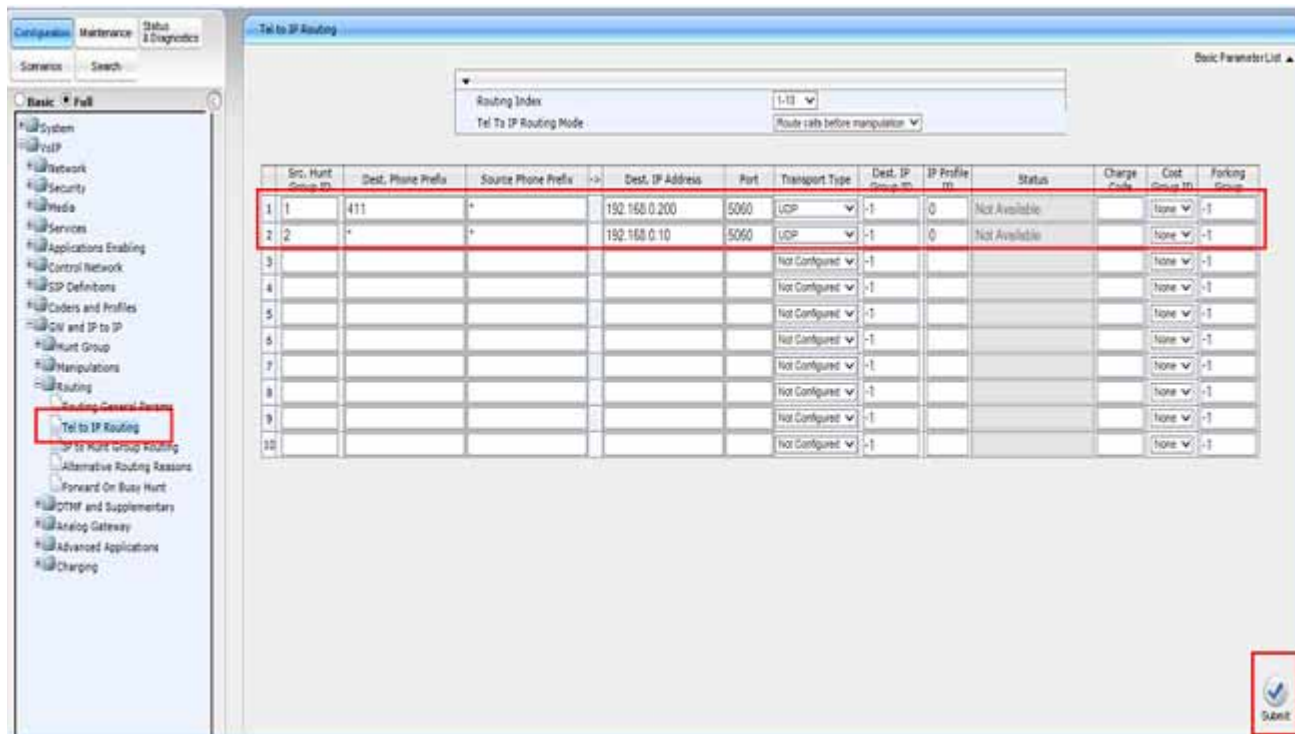
1. Open the **Hunt Group Settings** page (**Configuration-VoIP- GW and IP to IP-Hunt Group-Hunt Group settings**).
  2. From the Index drop-down and select the range.
  3. Assign **Hunt Group ID 2**.
  4. Assign the **Channel Select Mode** to **Ascending**.
  5. Assign **Registration Mode** to **Don't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

The screenshot displays the 'DTMF & Dialing' configuration page. The left sidebar shows the navigation menu with 'DTMF & Dialing' selected. The main content area contains a list of configuration parameters:

Parameter	Value
Max Digits In Phone Num	11
Inter Digit Timeout (sec)	3
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	
RFC 2833 Payload Type	101
Hook-Flash Option	Not Supported
Digit Mapping Rules	411011/xxxxxxxxxx(2-9)xxxxxxxx
Dial Plan Index	-1
Dial Tone Duration [sec]	18
Hotline Dial Tone Duration [sec]	18
Enable Special Digits	Disable
Default Destination Number	1000
Special Digit Representation	Special

The 'Submit' button is located at the bottom right of the page.

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

Setting	Value
Dialing Mode	One Stage
Waiting for Dial Tone	Yes
Time to Wait before Dialing [msec]	1000
Ring Detection Timeout [sec]	8
Reorder Tone Duration [sec]	255
Answer Supervision	No
Rings before Detecting Caller ID	8
Send Metering Message to IP	No
Disconnect Call on Busy Tone Detection (CAS)	Enable
Disconnect On Ura Tone	Disable
Guard Time Between Calls	1
FXO Double Answer	Disable
FXO AutoDial Play BusyTone	Disable
FXO Ring Timeout [100 msec]	0

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

Gateway Port	Destination Phone Number	Auto Dial Status	Hotline Dial Tone Duration (sec)
Port 1 FXS		Enable	0
Port 2 FXS		Enable	0
Port 3 FXO	1301	Enable	0
Port 4 FXO	1301	Enable	0

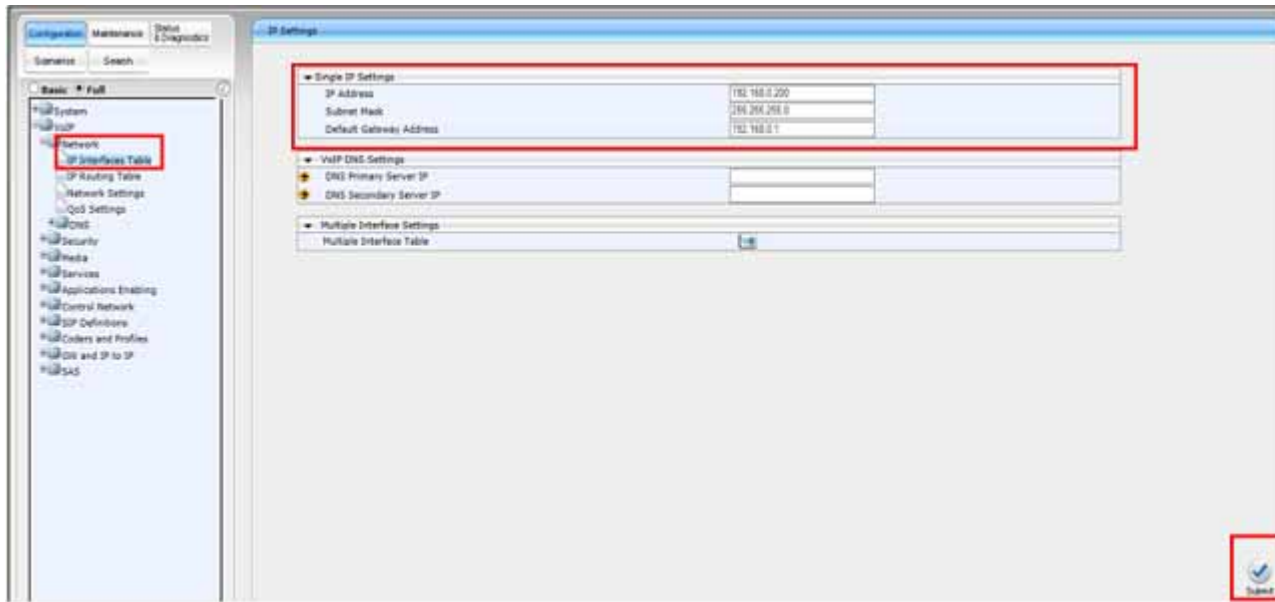
1. Open the **Automatic Dialing** page (**Configuration- VoIP- GW and IP to IP- Analog gateway- Automatic Dialing**).
2. Configure Automatic dialing on a per port basis. This is used for direct termination and Hotline assignments.
3. Click **Submit**.

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

The screenshot displays the 'IP Routing Table' configuration page. On the left is a navigation tree with 'IP Routing Table' selected. The main area shows a table of existing routes and a form to add a new entry.

#	Delete Row	Destination IP Address	Prefix Length	Gateway IP Address	Metric	Interface Name	Status
1	<input type="checkbox"/>	127.0.0.0	8	127.0.0.1	1		Active
2	<input type="checkbox"/>	127.0.0.1	32	127.0.0.1	0		Active
3	<input type="checkbox"/>	192.168.0.0	24	192.168.0.200	0		Active

Below the table is a 'Delete Selected Entries' button. Below that is a form titled 'Add a new table entry' with the following fields:

Destination IP Address	Prefix Length	Gateway IP Address	Metric	Interface Name
	16		1	

Below the form is an 'Add New Entry' button.

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

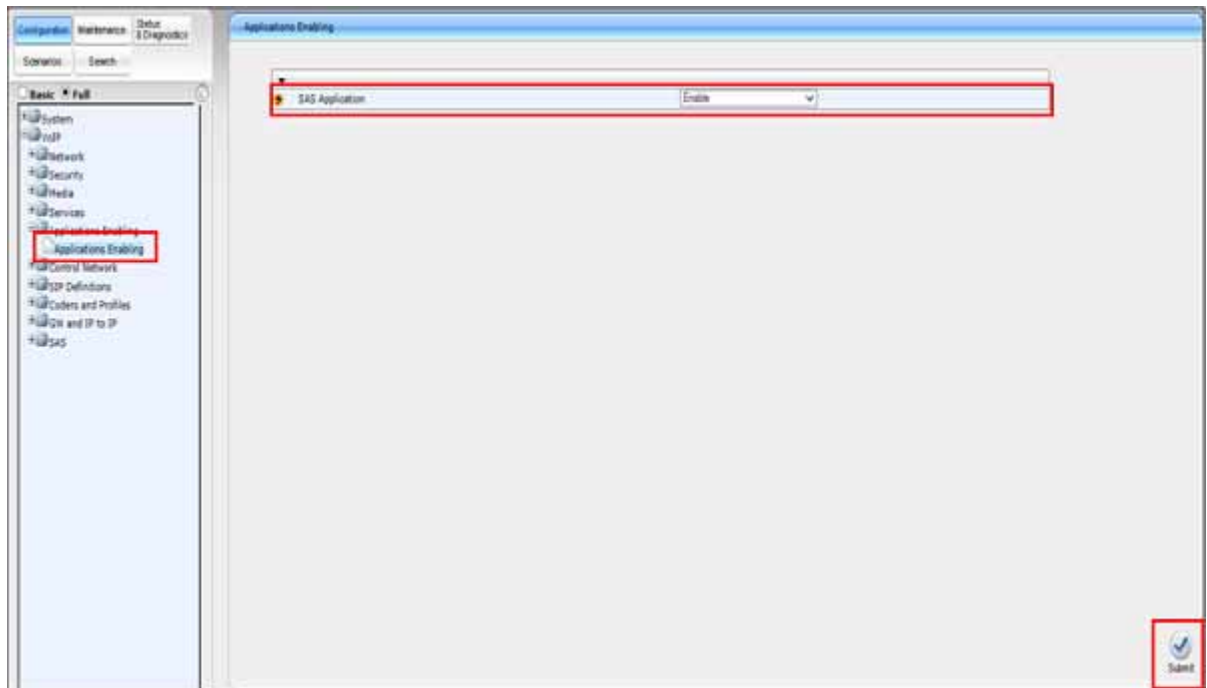
The screenshot shows the 'Fax/Modem/CID Settings' configuration page. The left sidebar contains a navigation tree with 'Fax/Modem/CID Settings' highlighted. The main content area is divided into three sections:

- General Settings:**
  - Fax Transport Mode: T.38 Relay
  - Caller ID Transport Type: None
  - Caller ID Type: Standard Software
  - V.21 Modem Transport Type: Disable
  - V.22 Modem Transport Type: Enable Bypass
  - V.23 Modem Transport Type: Enable Bypass
  - V.34 Modem Transport Type: Enable Bypass
  - Fax CNG Mode: Sends on CNG tone
  - CNG Detector Mode: Events Only
- Fax Relay Settings:**
  - Fax Relay Redundancy Depth: 0
  - Fax Relay Enhanced Redundancy Depth: 4
  - Fax Relay ECH Enable: Disable
  - Fax Relay Max Rate (bps): 14400bps
- Bypass Settings:**
  - Fax/Modem Bypass Codec Type: G.711 (uLaw)
  - Fax/Modem Bypass Packing Factor: 1
  - Fax Bypass Output Gen: 0
  - Modem Bypass Output Gen: 0

A 'Submit' button is located in the bottom right corner of the configuration area.

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

The screenshot shows the 'Proxy Sets Table' configuration page. On the left is a navigation tree with 'Proxy Sets Table' selected. The main area contains a 'Proxy Set ID' dropdown menu. Below it is a table with the following structure:

Proxy Set ID	Proxy Address	Transport Type
1	192.168.0.10	UDP
2		
3		
4		
5		

Below the table are several configuration options:

Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	30
Proxy Load Balancing Method	Disable
In Proxy not Sleep	No
Proxy Redundancy Mode	Not Configured

A 'Submit' button is located in the bottom right corner of the 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.
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

Parameter	Value
NAT IP Address	0.0.0.0
RRACK Mode	Disabled
Channel Select Mode	By Dest Phone Number
Enable Early Media	Disable
183 Message Behavior	Progress
Session Expires Time	0
Minimum Session Expires	30
Session Expires Method	REWRITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on CED
SIP Transport Type	UDP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	5060
Use user-phone in SIP URI	Yes
Use user-phone in From Header	No
Use Tel URI for Asserted Identity	Disable
Tel to IP No Answer Timeout	100

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

The screenshot shows the 'Proxy & Registration' configuration page. The left sidebar has 'Proxy & Registration' selected. The main area contains the following settings:

Use Default Proxy	Yes
Proxy Set Table	Use
Proxy Name	
Redundancy Mode	Priority
Proxy IP List Refresh Time	30
Enable Fallback to Routing Table	Disable
Prefer Routing Table	Yes
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Redundant Routing Mode	Routing Table
SIP Relinquish Mode	Standard Mode
Enable Registration	Disable
Gateway Name	192.168.0.10
Gateway Registration Name	192.168.0.10
DNS Query Type	SRV
Proxy DNS Query Type	A/AAAA
Subscription Mode	Per Endpoint
Number of RTs Before Hot-Swap	3
Use Gateway Name for OPTIONS	No
User Name	
Password	Default_Password
Choice	Default_Choice
Registration Mode	Per Endpoint

Buttons at the bottom: Register, Use Register, Submit.

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

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.729	20	8	10	Disabled
G.711Law	20	32	0	Disabled

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

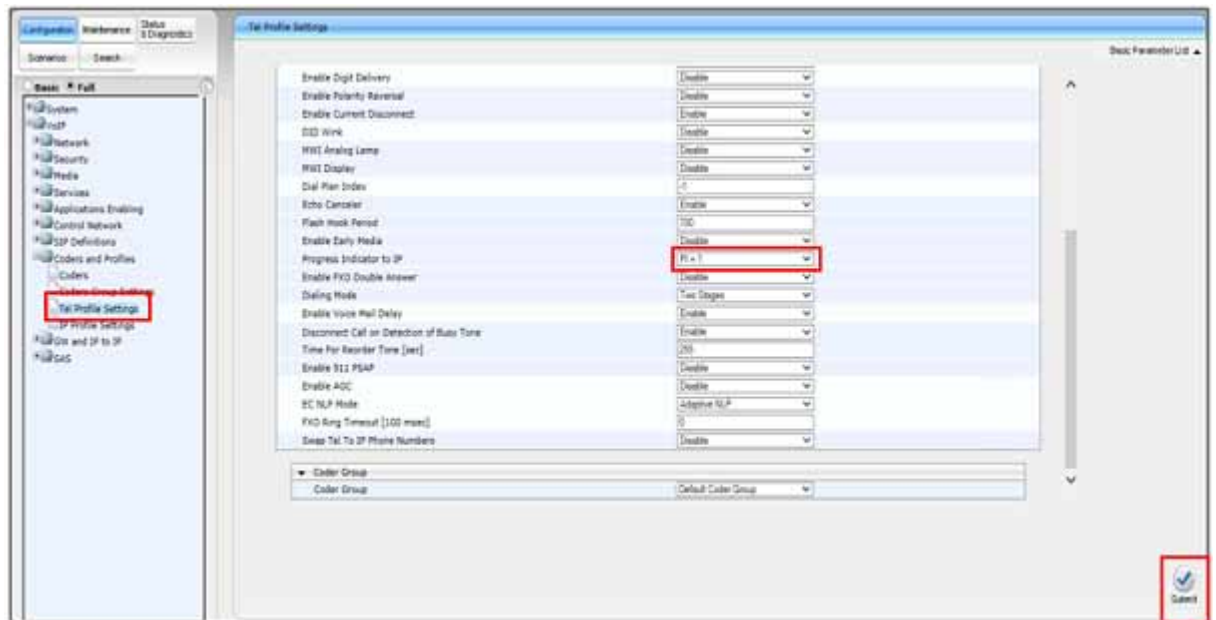
The screenshot displays the 'Tel Profile Settings' configuration page. The left sidebar shows the navigation menu with 'Tel Profile Settings' highlighted. The main content area contains the following parameters:

Parameter	Value
Profile ID	1
Profile Name	
Profile Parameters	1
Fax Signaling Method	T.38 Fax
Dynamic Jitter Buffer Minimum Delay (ms)	10
Dynamic Jitter Buffer Optimization Factor	10
STP DiffServ	40
Signaling DiffServ	40
Voice Volume (-32 to 31 dB)	0
DTMF Volume (-32 to 31 dB)	11
Input Gain (-32 to 31 dB)	0
Enable Digit Delivery	Disable
Enable Polarity Reversal	Disable
Enable Current Disconnect	Enable
DID Work	Disable
MRP Analog Lamp	Disable
MRP Display	Disable
Dial Plan Index	1
Echo Canceler	Enable
Flash Hook Period	750
Enable Early Media	Disable

A 'Submit' button is located at the bottom right of the page.

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 **Progress Indicator to IP** to **PI=1**.
3. Click **Submit**.

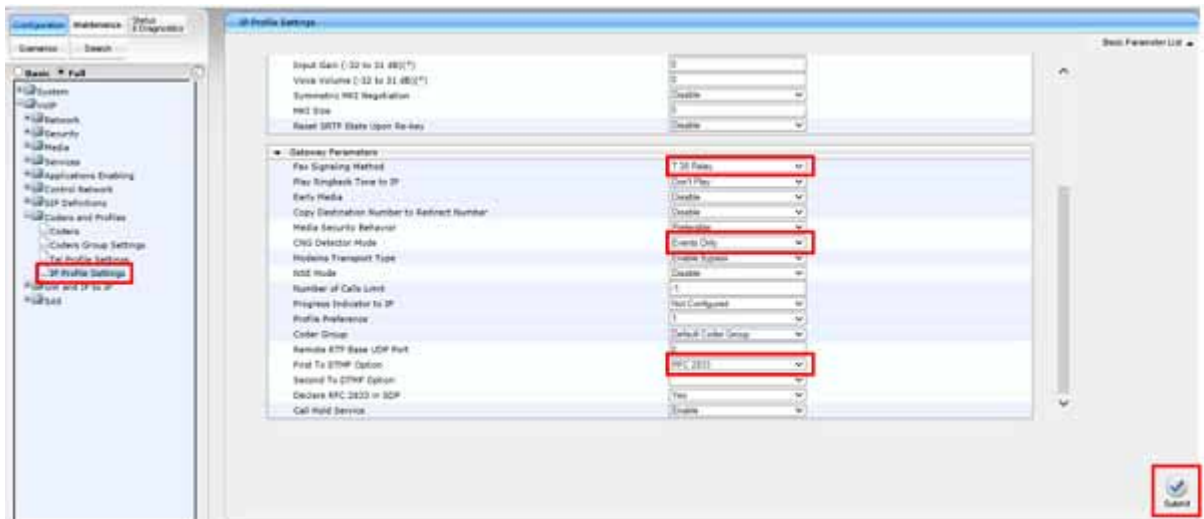
Figure 3-44 Coders IP Profiles Settings

The screenshot displays the 'IP Profile Settings' configuration page. The left sidebar shows a navigation tree with 'IP Profile Settings' selected. The main content area is divided into sections: Profile ID, Profile Name, Common Parameters, and Gateway Parameters. The following table summarizes the settings shown in the image:

Section	Parameter	Value
Profile Information	Profile ID	1
	Profile Name	
Common Parameters	RTP DT Differs	46
	Signaling Differs	46
	Disconnect on Broken Connection	Yes
	Dynamic Jitter Buffer Maximum Delay (msec)(*)	16
	Dynamic Jitter Buffer Optimization Factor(*)	10
	RTP Redundancy Depth(*)	3
	Echo Canceler(*)	Enable
	Drop Gap (-32 to 31 dB)(*)	0
	Voice Volume (-32 to 31 dB)(*)	0
	Summation MUX Negotiation	Disable
Gateway Parameters	Max Size	0
	Reset SIP State Upon Re-Inv	Disable
	Fax Signaling Method	T.38 Fax
	Play Ringback Tone to DP	Start Play
	Early Media	Disable
Copy Destination Number to Redirect Number	Disable	
Media Security Behavior	Enforce	

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

Channel	Phone Number	Hunt Group ID	Tel Profile ID
1	1170	01	0
2	1171	01	0
3		02	0
4		02	0

Submit

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

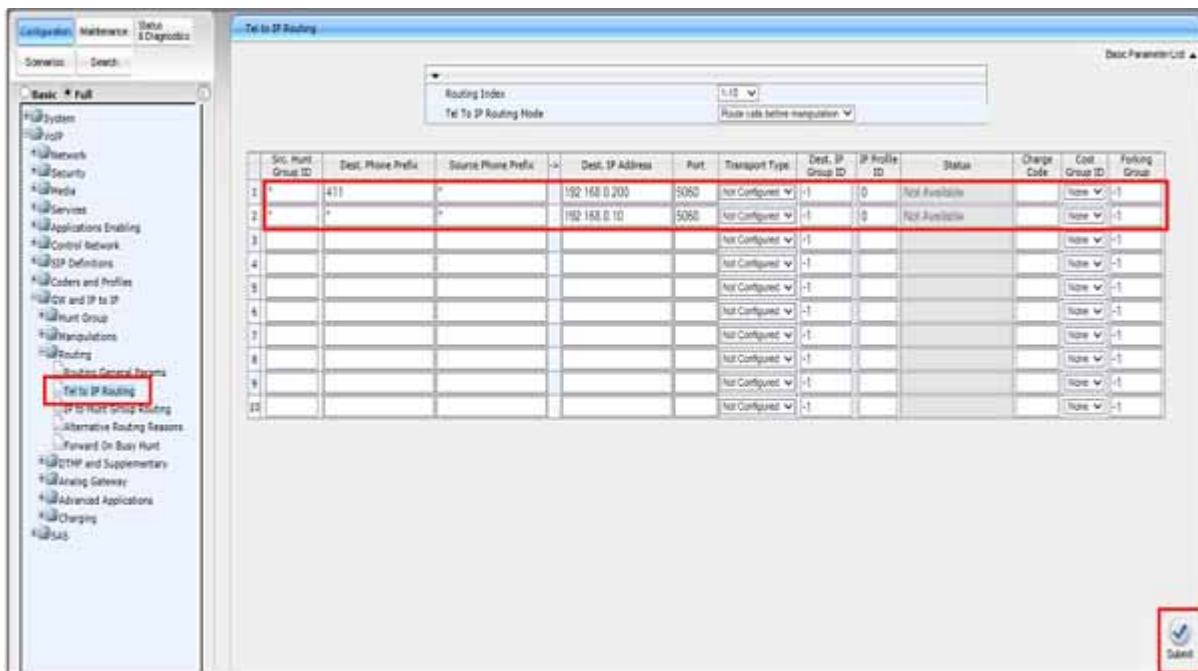
The screenshot shows the 'Hunt Group Settings' page. The left sidebar has a tree view with 'Hunt Group Settings' selected. The main area has a table with the following data:

Index	Hunt Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID	Gateway Name	Contact User
1	1	By Destination Phone Number	Don't Register			
2	2	Ascending	Don't Register			
3						
4						
5						
6						
7						
8						
9						
10						
11						
12						

Buttons at the bottom: Register, Un-Register, and Submit.

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**.


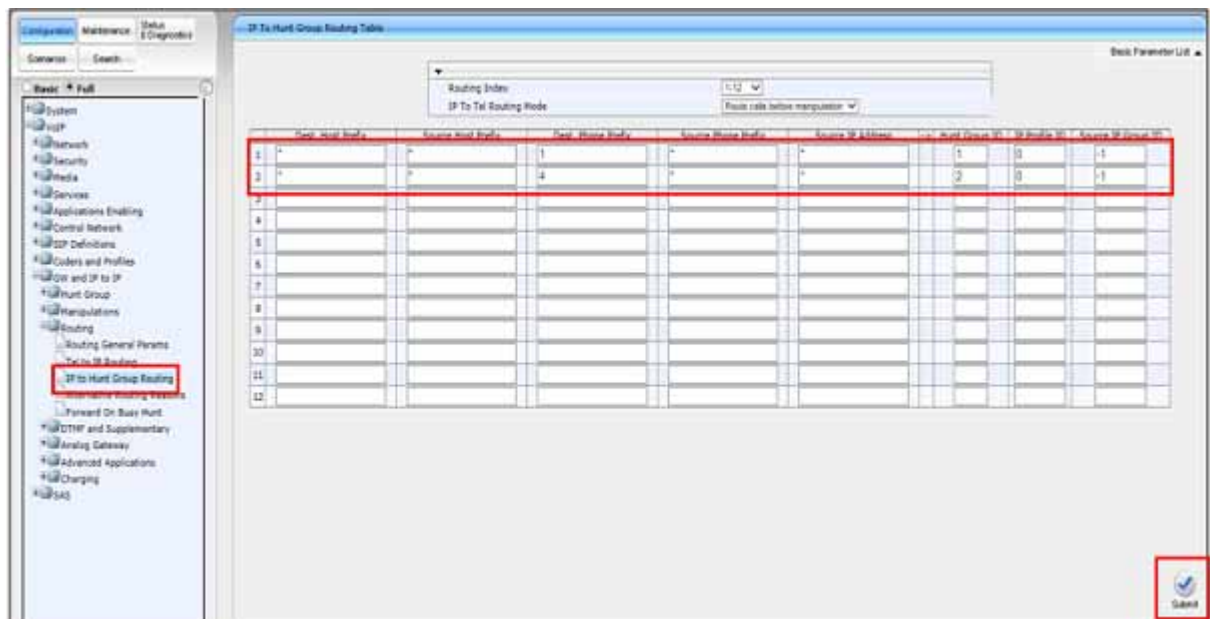
 Refer to the *User's Manual Tel to IP* section for a detailed explanation of the routing parameters.



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**).

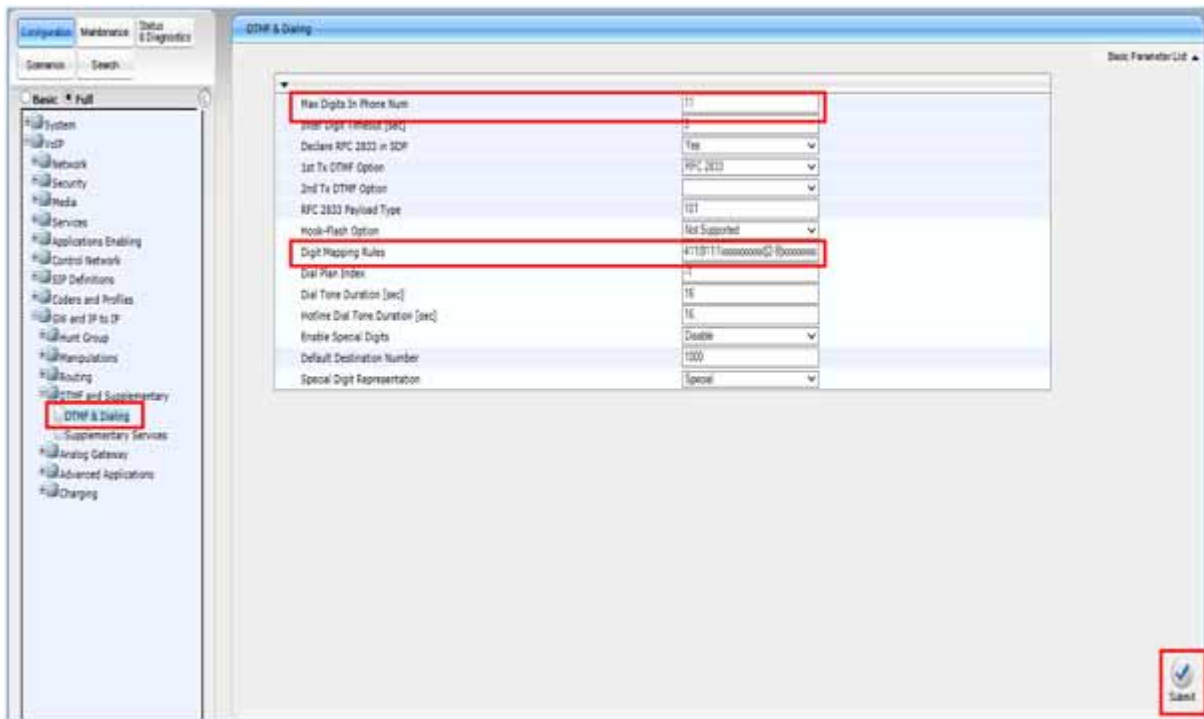
The above example will send an incoming call starting with the digit 1 to hunt group 1 (FXS hunt group).

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

Setting	Value
Dialing Mode	One Stage
Waiting for Dial Tone	Yes
Time to Wait Before Dialing (msec)	1000
Ring Detection Timeout (sec)	5
Reorder Tone Duration (sec)	200
Answer Supervision	No
Rings before Detecting Caller ID	5
Send Waiting Message to IP	No
Disconnect Call on Busy Tone Detection (CAD)	Enable
Disconnect On Use Tone	Disable
Guard Time Between Calls	1
FXO Double Answer	Disable
FXO AutoDial Play Busy Tone	Disable
FXO Ring Timeout (100 msec)	5

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<sup>®</sup> SV9300**

## **AUDIOCODES MEDIAPACK FXS/FXO CONFIGURATION GUIDE**

NEC Corporation of America

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