4 Port

Analog Gateway

User Manual

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About this manual

This manual describes the Allo product application and explains how to work and use it major features. It serves as a means to describe the user interface and how to use it to accomplish common tasks. This manual also describes the underlying assumptions and users make the underlying data model.

Document Conventions

In this manual, certain words are represented in different fonts, typefaces, sizes, and weights. This highlighting is systematic; different words are represented in the same style to indicate their inclusion in a specific category. Additionally, this document has different strategies to draw User attention to certain pieces of information. In order of how critical the information is to your system, these items are marked as a note, tip, important, caution, or warning.

Icon	Purpose
	**Note**
	**Tip/Best Practice**
	**Important**
	**Caution**
	**Warning**

- **Bold** indicates the name of the menu items, options, dialog boxes, windows and functions.
- The color blue with underline is used to indicate cross-references and hyperlinks.
- Numbered Paragraphs - Numbered paragraphs are used to indicate tasks that need to be carried out. Text in paragraphs without numbering represents ordinary information.
- The Courier font indicates a command sequence, file type, URL, Folder/File name
- e.g. www.allo.com

Support Information

Every effort has been made to ensure the accuracy of the document. If you have comments, questions, or ideas regarding the document contact online support: http://support.allo.com

Table of Contents

Introduction

1. Product introduction

1.1. Overview

The Analog Gateway is a compact system to reach of small businesses and it is standalone, fan less and can convert up to 4 analog ports (FXO or FXS modules) to VOIP. Setting up and configuring the Gateway is a breeze with the user-friendly GUI and this document will show you just how easy it is!

The Model No's are as follows

4 Port FXS Analog Gateway- aGA40

4 Port FXO Analog Gateway- aGA04

2 Port FXS+ 2 Port FXO Analog Gateway- aGA22

A typical network diagram shows the function of Gateway as below.

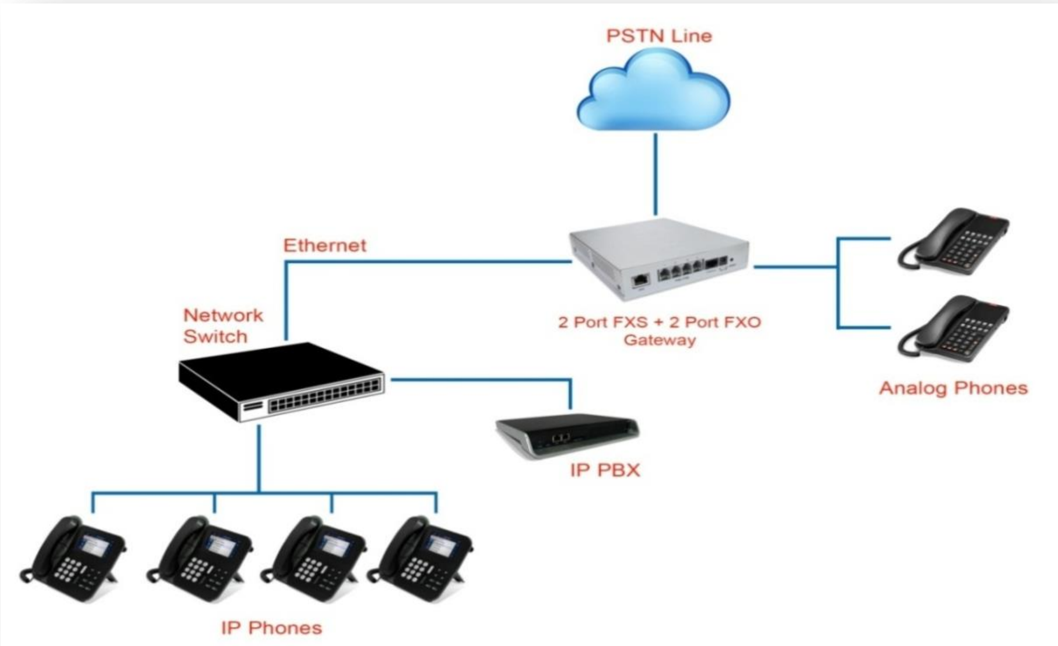

Figure 1: overview

1.2 Equipment Structure

1.2.1 Front View

Figure 2: Front View

Interface	Description
ON/OFF Switch	Gateway Power ON/OFF switch
Power	Power LED is Green if the unit is boot up
FXS/FXO LED Ports	FXO port blinks GREEN FXS port blinks RED

Notification LEDs (On the Front Panel):

Below table shows the Notification LED for 2 port FXS+2 port FXO Gateway.

Port 4	Port 3	Port 2	Port 1
FXO Green Blink	FXO Green Blink	FXS Red Blink	FXS Red Blink

In the case of 4 port FXO gateway all the 4 ports blink GREEN

4 port FXS gateway – all the 4 LED's blink RED

1.2.2 Rear view

Figure 3: Rear View

Interface	Description
WAN	Standard 10/100 BASE-TX Ethernet Interface for WAN
FXS/FXO ports	FXS is used to connect analog phones and FXO is used to connect PSTN lines.
Console	Console cable provided along the box for the trouble shooting
DC +12 V Supply	This is power rating specified for the gateway.
Reset	Reset button for factory default.

1.3 Functions and Features

- Compatible with ElastixTM, Asterisk, Trixbox TM, FreePBX TM, 3CX (Only the 4 port model).
- On board 128ms Line Echo Cancellation DSP On board Transcoding (G711, G729) DSP
- 4, 8, 16 analog ports (FXO or FXS) and 8 analog ports (FXS/FXO) + 8 ISDN BRI model
- T.38 fax relay
- Pre configured for almost plug and play experience.
- Clean and easy to use web interface
- Caller ID
- DTMF Digit Detection and Generation
- SIP User Agent IETF RFC3261 compliant

WARRANTY

Hardware Warranty: 1 year

If the PRI Gateway was purchased from a Distributor/reseller, please contact the company where the device was purchased for replacement, repair or refund. If the device was purchased directly from Allo.com, contact our Technical Support Team for a RMA (Return Materials Authorization) number before the product is returned. Allo.com reserves the right to remedy warranty policy without prior notification.

Use the power adapter provided with the ALLO PRI Gateway. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

Getting Started

2. Getting Started with the Gateway

2.1 Hardware Installation

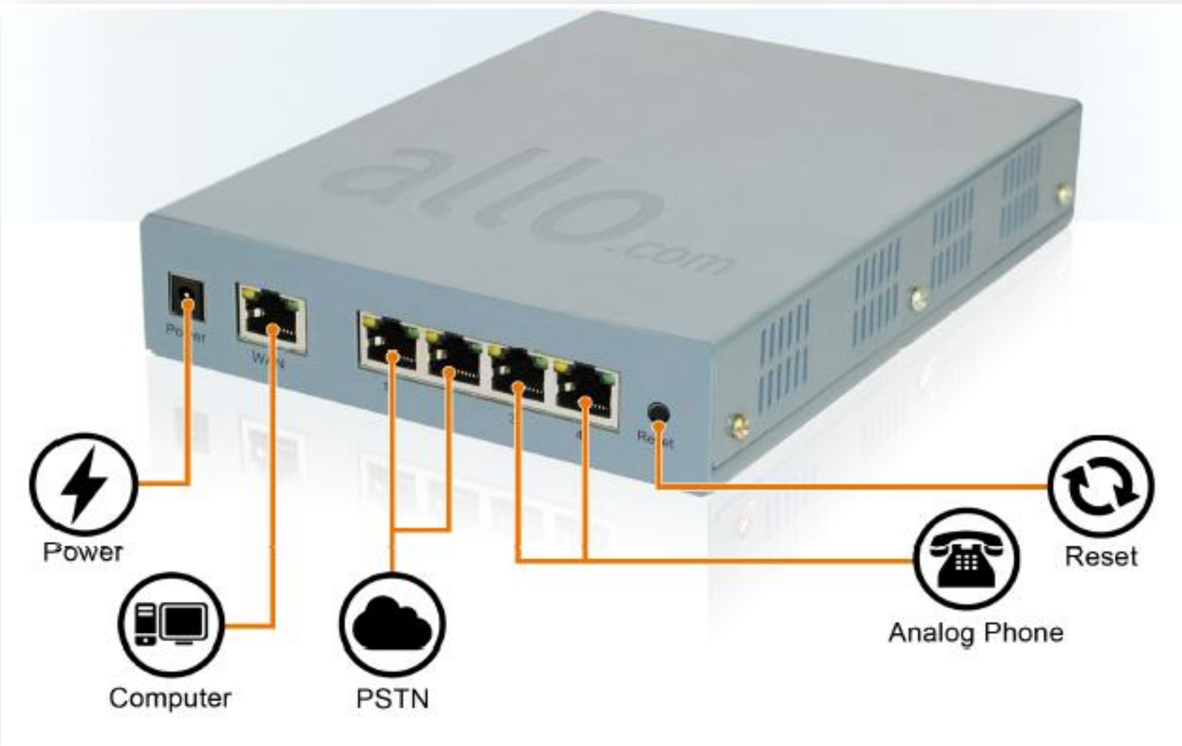

Figure 4: Getting Started with the Gateway

Hardware Installation

1. Unpack the items from the box.

2. Plug one end of the RJ45 Ethernet cable into the WAN port of the FXO/FXS Gateway & other end into the PC.

Use Cross over Ethernet cable to connect between the WAN port of the unit & PC.

3. Insert the Power Adaptor output connector into the "Power" port of the FXO/FXS

Gateway & Plug in the Power Adapter to any available AC power outlet.

2.2 Accessing the Web GUI

FXO/FXS Gateway WEB GUI can be accessed either through WAN

Steps to Access the GUI during the initial setup:

1. Make the setup as described in Hardware setup section and then plug one end of the RJ45 Crossover Ethernet cable to your PC & the other end into the WAN port of the FXO/FXS Gateway.
2. Change the Network setting of the PC is set in manual mode (i.e. Static IP mode). Assign the IP address to the PC in the range of 192.168.113.xxx (E.g:192.168.113.10), net mask as 255.255.255.0 and gateway & DNS as 192.168.113.1.
3. Launch the web browser and enter the URL Access the GUI using http://192.168.113.252:8088which is the default IP address of the WAN of the FXO/FXS Gateway.
4. Login using the default username & password (Default: Username: admin; Password: admin). Successful login takes you to the Dashboard page.

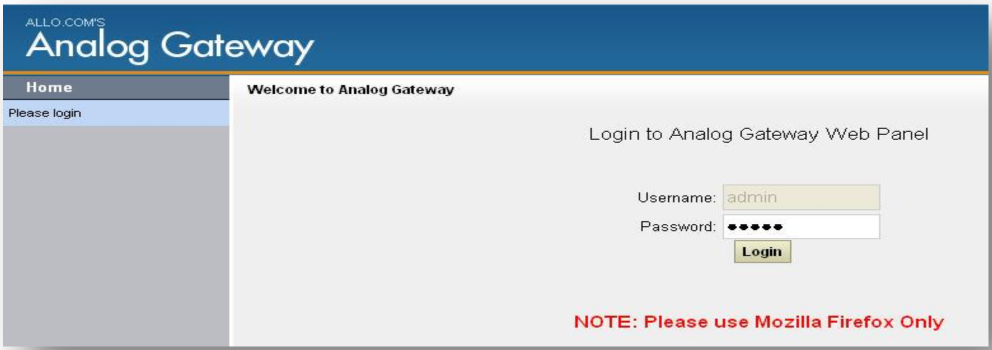

Figure 5: Login Page

5. To change the password, please refer to the **General Settings > Admin Settings → General Settings** section in the navigation.

6. Click on Network Settings, Change the network setting to DHCP/static IP depends upon your network scenario.
7. Now connect the Straight through RJ45 Ethernet cable from WAN port of Analog GW to your Network Switch. Also connect RJ45 Ethernet Cable from PC to Network Switch. Reconfigure your PC network configuration as per your LAN network.

a) We strongly recommend you to change the Gateway Admin password from Factor default to Alpha numeric password to reduce the possibility of a security breach.
b) Use Mozilla Firefox Browser only.
c) Boot up process of the Gateway takes about 40 seconds
d) Make sure to add the port number 8088 at the end of your IP address when logged in. For an Example: 192.168.0.42:8088
e) IP address can be obtained by dialing **# from the Analog phone which is connected to the Gateway.
f) The FXO/FXS VOIP gateway comes with default credentials admin. You have to change these as soon as possible to avoid getting hacked. Allo.com cannot be held responsible for unauthorized access to the FXO/FXS VOIP Gateway. Please make sure secure your Gateway by placing it behind a firewall and changing passwords with frequently. Kindly avoid using multiple login to the GUI.

Status

3. System Status

Navigation to System status, Status of FXO/FXS VOIP Gateway including Memory Status, VOIP Status, Networking Status and Client Status (to see which clients are connected to the system).

After you login, you are brought to a System Status screen, which offers information about the Gateway, and help files to assist you in learning about all the different features of the system.

1. Under **Trunks** you will see the Registration Status of the VOIP Account(s) configured. If it displays "Registered" then it is successfully configured and connected.
2. Under **Extensions** you will see all the users and the extensions connected to the Gateway.
3. Under **System status > System Info > General** you will have a summary of your General information, such as Hostname, Server Date & time zone, Uptime.
4. Under **System status > System Info > Memory Status** you will see total memory resources, including RAM usage.
5. Under **System status > System Info > Network Status** you will have a summary of your Network information, such as Hostname, WAN IP Address, Subnet Mask, WAN MAC Address and Default Gateway (you may refer to Settings > Network Settings for more info).
6. Under **System status > System Info > Disk** you will have a summary of Disk usage and Disk free space available on the file system.

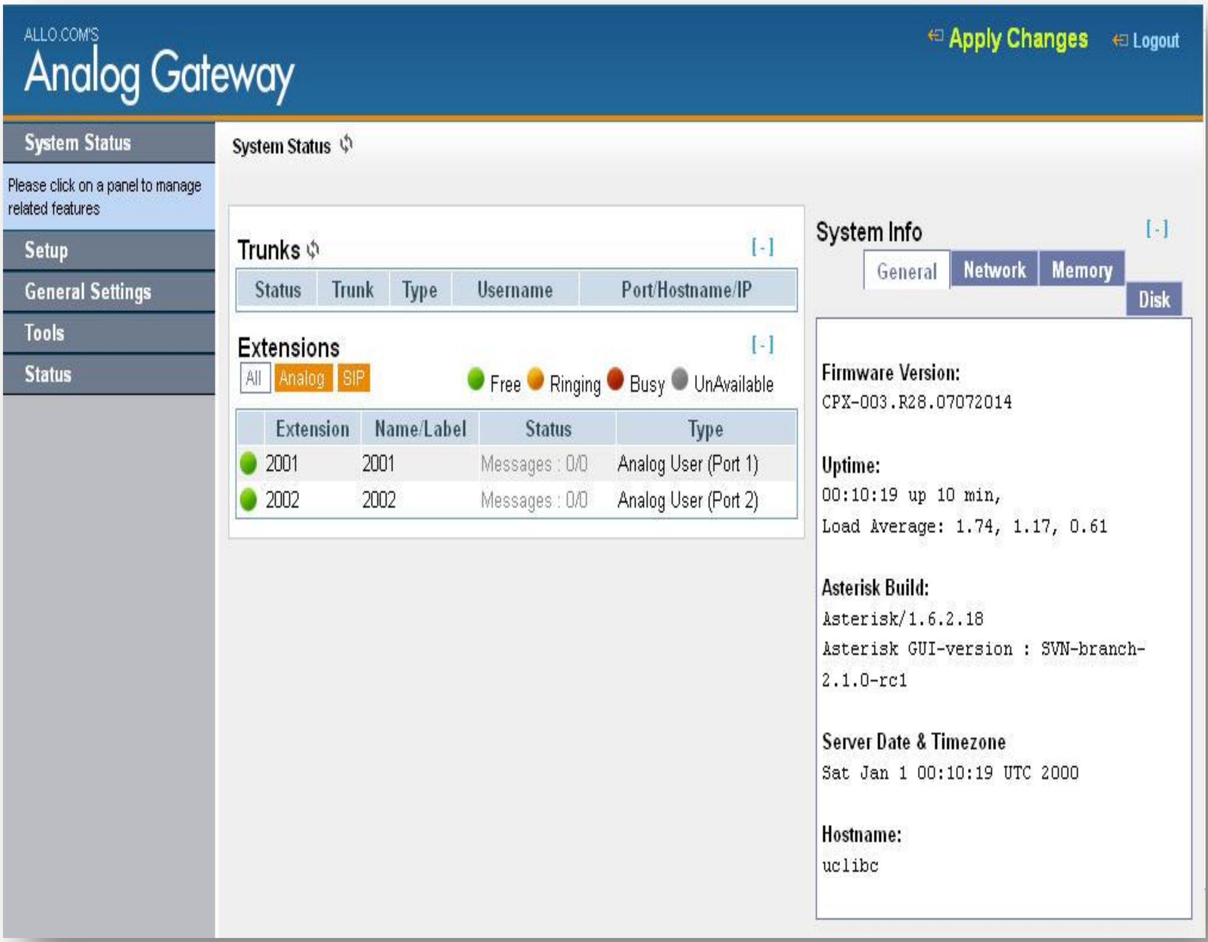

Figure 6: System Status

Setting up Features

4. Set up

4.1 Configure hardware

Configure hardware section lists the Analog port (FXO/FXS) information.
FXS is used to connect analog phones and FXO is used to connect PSTN lines.

Below screenshot list the Analog Gateway port configuration of 2 port FXS+2 port FXO gateway (aGA22)

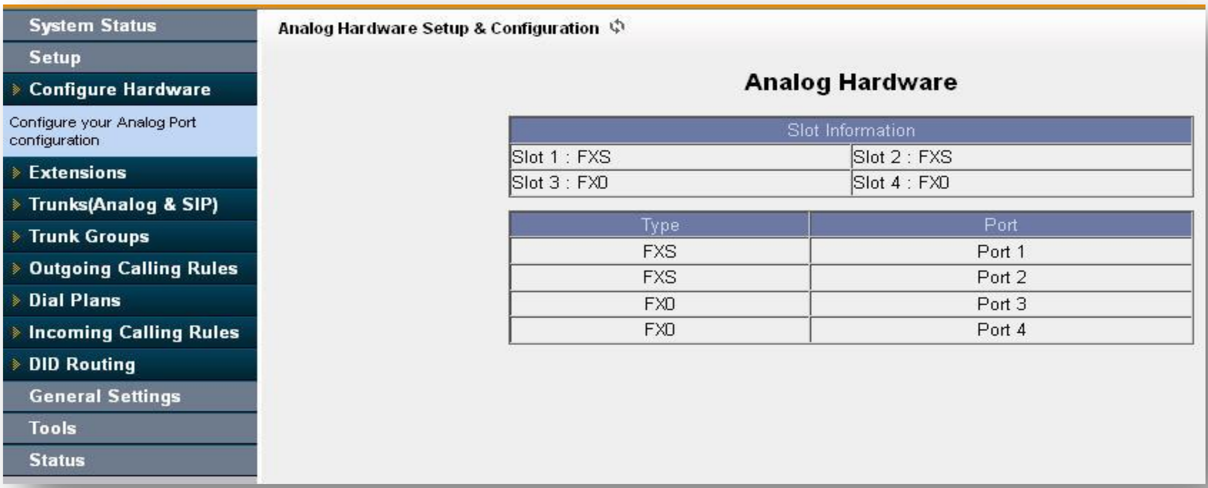

Figure 7: Analog Hardware

4.2 Extensions

Extensions are a shortcut for quickly adding and removing all the necessary configuration components for any new phone. Extensions are the core of the FXO/FXS VOIP. An extension is a number mapped to a person. So basically, every employee that is connected to the Gateway should have their own unique extension number so that he/she can be reached and be able to place calls.

The Gateway supports 2 types of Extensions: IP Extensions and Analog Extensions.

IP Extensions:

IP extension is used to create the "To Gateway Registration" i.e. End point (SIP Phone or PBX) going to register to the Gateways.

Analog Extensions:

An Analog Extension is used with a regular telephone system which can be connected to an available FXS port of the Analog VOIP Gateway.

4.2.1 Create New User

You have to create at least one Dial plan using Dial plan option before trying to create Analog/IP Extensions

Navigation**: Users > Create New user**: This is where you setup your Analog/IP extensions.

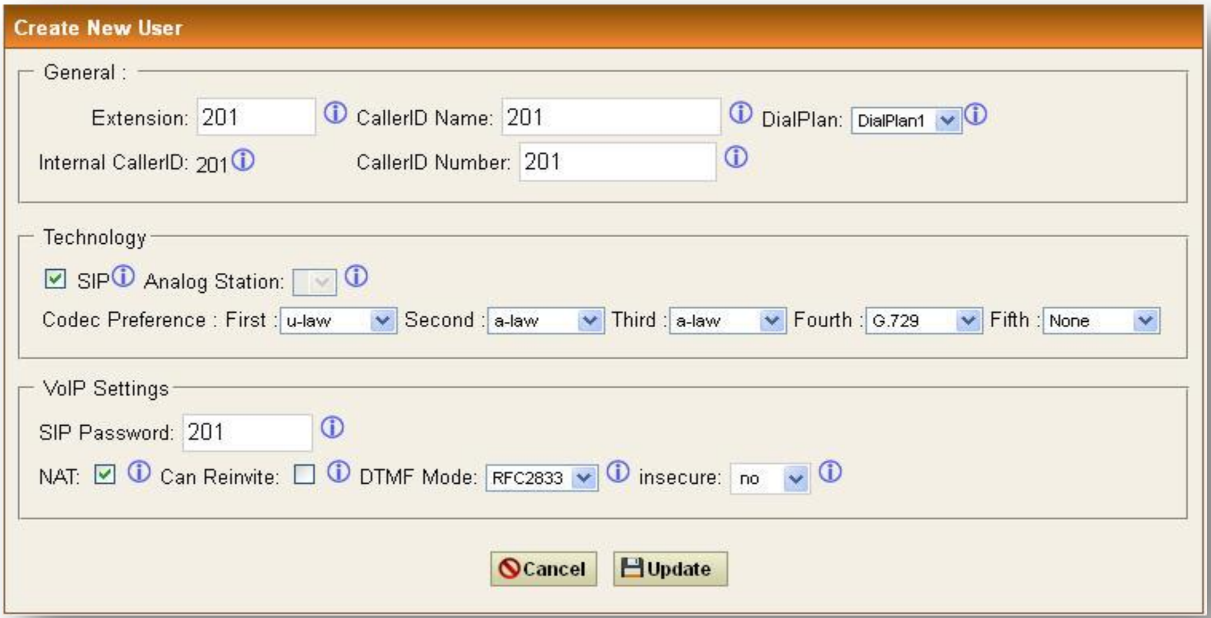

Figure 8: Create new user

Technology	
SIP	Select "SIP" if the user is using SIP or a SIP device
Analog Station	Select the FXS port need to be configured from the drop down menu.

Codec Preference	Select audio and video codec for the extension. The available codec's are: G711U-law, G711A-law, G.726, and G.729.
General	
Extension	The extension number associated with the user.
Caller ID Name	Configure the Caller ID Name associated with the user
Dial Plan	Select one from the dropdown box (appears only if dial plans are created using Dial plans).By Default: Dialplan1
Caller ID Number	Configure the Caller ID Number that would be applied for outbound calls from this user. The ability to manipulate your outbound Caller ID may be limited by your VOIP provider.

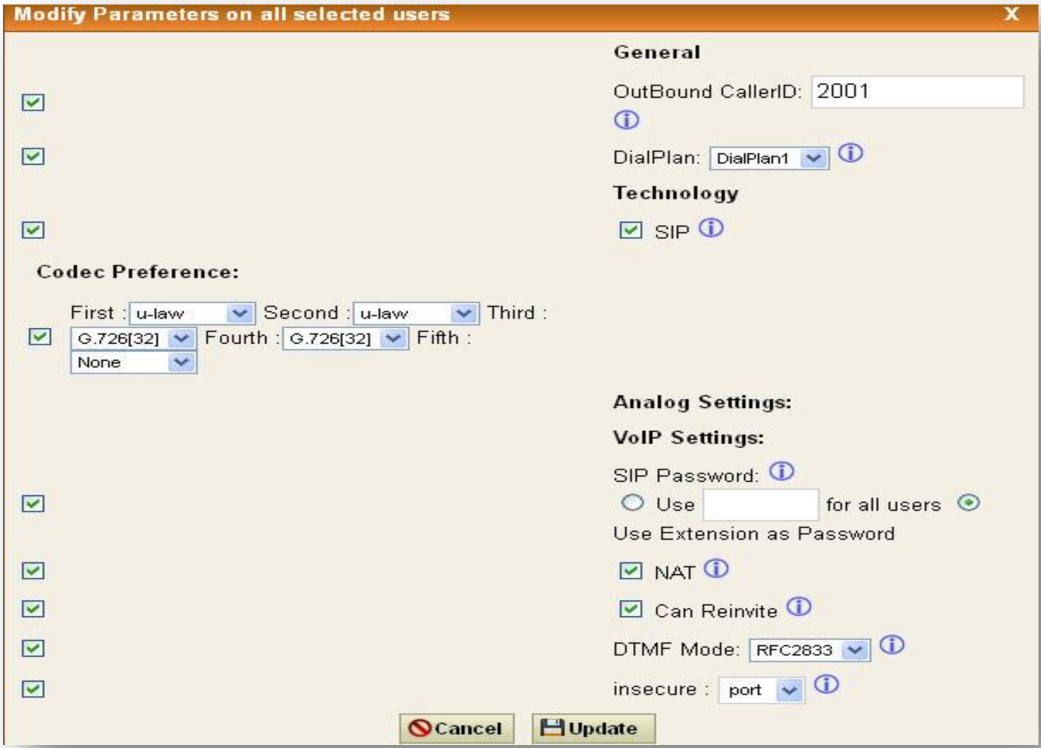

Figure 9: Modify Parameters on all selected users

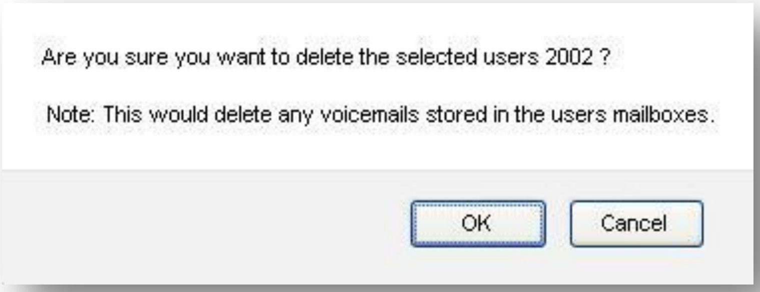

Figure 10: Delete User Name

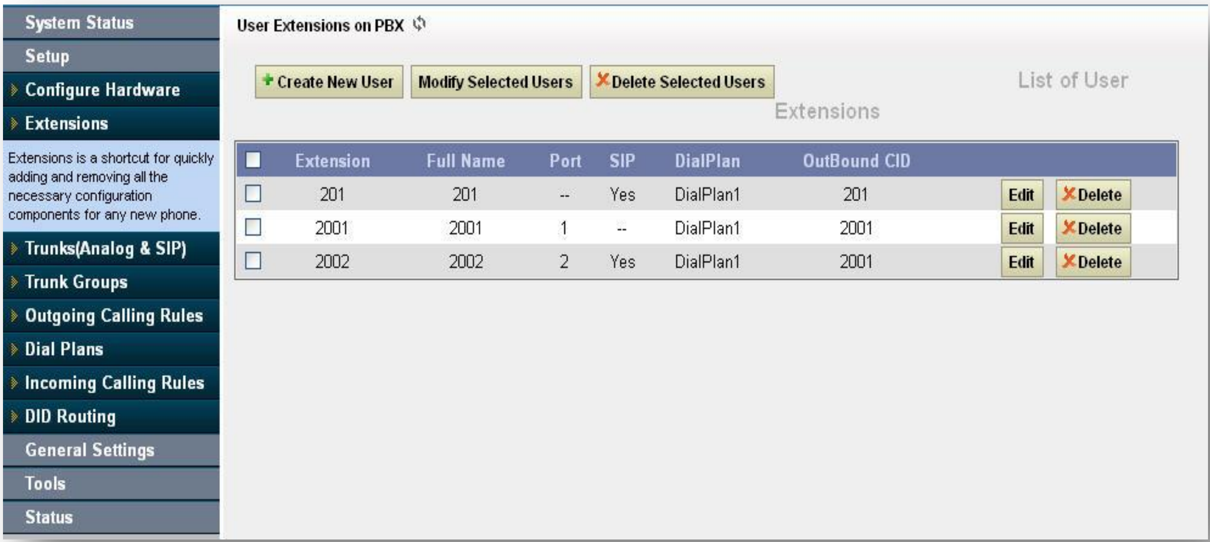

Figure 11: Extension Result

VOIP Settings	
SIP Password	Configure the password for the user.
NAT	Use NAT when the Gateway is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports.
Can Re invite	By default, the Gateway will route the media steam from SIP endpoints through itself. If enabled, the Gateway will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the Gateway to negotiate endpoint-to-endpoint media routing. The default setting is "No".

Insecure	• Port: Allow peers matching by IP address without matching port number. • Very: Allow peers matching by IP address without matching port number. Also, authentication of incoming INVITE messages is not required. • No: Normal IP-based peers matching and authentication of incoming INVITE. The default setting is “No".

4.2.2 Modify/Delete selected users

Navigation: **Users**: This is where you can edit / delete existing Analog/IP Extensions individually.

On the right side of the page, you can see the list of extensions you have setup. To edit or delete any of them simply click the appropriate icon provided to the right of each account/Extension.

Once you click on the Edit button of an Extension then it will display the information about that particular extension. Here you can change the required details and then click on the Update Extension button to save the changes made.

You can delete an extension by clicking on the delete button on the extension from the list of extensions displayed.

You can delete many existing extensions by clicking on the Delete Selected Users button after marking in the check boxes of extensions from the list of extensions displayed. Click ok the popup window to delete the selected users.

Make sure to click the APPLY CHANGES button in the top navigation bar, after adding/editing/deleting any Extension. The APPLY CHANGES tab appears if some changes are made and not saved.

4.3 Trunks

Navigation: **Setup >Trunks > Analog Trunks**: Here you can configure the Analog Trunks.

4.3.1 Analog Trunks

Figure 12: Analog Trunks

The analog trunk options are listed in the table below.

Analog Trunks	
Channels	Select the channel for the analog trunk.
Trunk Name	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules and etc.

4.3.2. VOIP Trunks

Navigate through **Setup > Trunks > VOIP Trunks**: This is where you setup VOIP Trunk or manage existing ones.

In this page, fill in the Provider Name, Host name, Host Port, Username, Password and Proxy information given to you by your VOIP provider (known as SIP Credentials). Apply any Codec Settings required. You can prioritize your active codec's by using the drop down buttons.

After you have entered the details, click the Save button at the bottom.

In Create new sip trunk dropdown option > context naming "Based on provider name" and "Assigned by asterisk GUI "need to be removed from GUI.

4.3.3Adding a New VOIP Account Details

Create New SIP trunk	
Context Naming	You can select the context naming based on username/Provider name/Assigned by asterisk GUI

	By Default: Based on Username
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VOIP provider’s server of the trunk.
Host Port	Enter the port number given by the VOIP provider. 5060 is the default SIP port. If configuring any roaming extension or if connecting gateway to public network then consider changing the SIP port for better security. Once port changes are done, for registering IP phones/sip phone use <domain name ><sip port number>.
Username	Enter the username to register to the trunk from the provider.
Password	Enter the password to register to the trunk from the provider.
Qualify	If you enable Qualify, asterisk will send the SIP Options command regularly to check that the device is still online
DID Routing	DID Routing Routes calls to a single, specific extension
Outbound Proxy	Configure Outbound proxy to send Outbound signaling to that proxy.

4.3.4 Editing / Deleting an Existing VOIP Account

On the right side of the page, you can see the list of VOIP trunk you have setup. To edit, or delete any of them, simply click the appropriate icon provided to the right of each trunk. Once you click on the edit button of a VOIP trunk then it will display the information of that particular VOIP trunk, here you can change the required details and click on the Save button and then click on the apply changes tab to save the changes made.

To ensure successful registration of your VOIP Trunk, you must click the “System Status” tab on the top navigation menu (see Status section for more info)

Make sure to click the APPLY CHANGES tab in the top navigation bar after adding any new VOIP trunk or editing/deleting.

4.4 Trunk Groups

Navigate through Setup > Trunk Groups

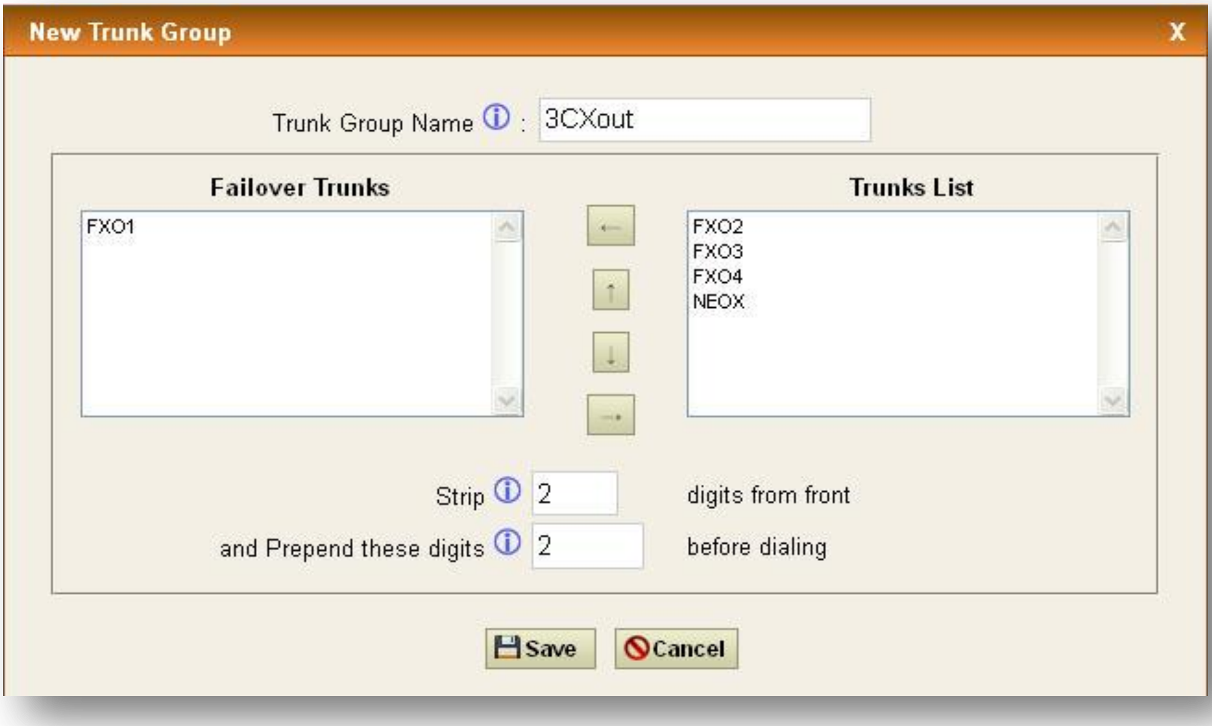

Figure 13: New Trunk Group

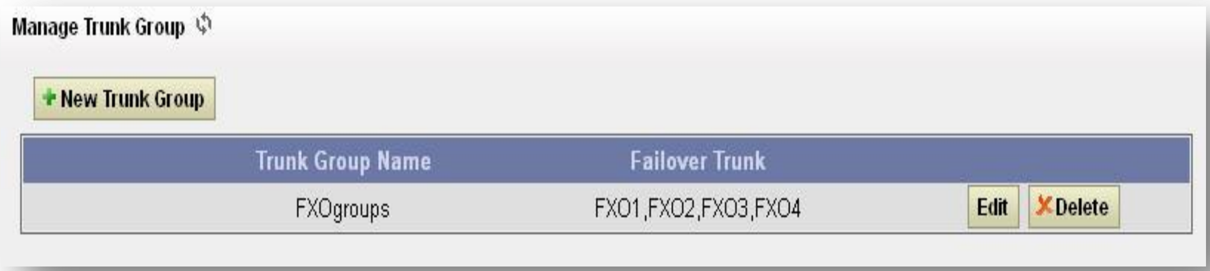

Figure 14: Manage Trunk Group

4.5 Outgoing Calling Rule

4.5.1 New Calling Rules

Navigate through Setup >**Outgoing Calling Rules > New Calling Rule**: This is where you configure Dial out Rules.

Outgoing Calling Rules represent the prefix sequence used to dial when making an outgoing call either through the PSTN (Analog) or VOIP. There are two ways to make outgoing calls for the registered extension users:

- VOIP / SIP trunk via ITSP gateway
- LINE/PSTN trunk via FXO port

Figure 15: New Calling Rule

Calling Rule Name	Configure the name of the calling rule (e.g., local, long-distance, and etc). Letters, digits, etc.

Pattern	All patterns are prefixed with the "_". X: Any Digit from 0-9. Z: Any Digit from 1-9. [12345-9]: Any digit from 1 to 9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately.
Caller ID Name	Configure the CallerID Name associated with the FXO trunk

You can even use the Calling rule to route it to local extensions as a Destination or Route the calls to Trunks created so as to make the calls successful. There as still many options like striping the number of digits from the front, prepend the digits before dialing and filter. There is also another alternative way if the trunk fails to route the call i.e. Failover Trunk an alternative trunk to route calls.

Send this call through trunk	
Use Trunk	Select the trunk for this outbound rule.(FXO/VOIP Trunk)
Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Filter	This option is used to filter out certain characters. The characters listed in the field will be permitted, while all others will be filtered out.

Here we will discuss about how to choose outgoing trunks in two different ways.

The First way is, choose a provider or a trunk based on prefix.

This type of rule will allow users to create a prefix for choosing SIP (VOIP) or LINE (PSTN / FXO) trunk to make an outgoing call.

For Eg 1: If you would like to strip out the first digit from the dialed number follow the example: Add **"_8X."** in pattern & configure **"1"** in strip out field to remove the first digit from the dialed number either analog or SIP users which use the same dial plan

The Second way is, choose a provider or trunk based on actual number dialed,

This type of rule will allow the user to choose a suitable provider based on Country code. For e.g. If the user wants to Dial 44-9872837532, then adding **"_X."** in Dial Pattern and configure "0" in Strip Out field and add "44" in Prepend field & selecting VOIP Trunk/ Analog Trunk in trunk sequence, GATEWAY will allow number dialed from **44+** followed by (any digit from 0 to 9) like 449872837532 will route through VOIP Provider 2. If the ITSP / VOIP provider offer cheaper rates for the region where number starts from 44 users can make use of this rule.

In the same way, user can create a prefix and select a LINE (PSTN / FXO) trunk to make an outgoing call.

Use failover trunk	
Failover trunks	Check this option to use Failover trunks where a call goes through an alternate route when the primary trunk is busy or down.
Trunks List	Gives the list of trunks configured.
Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Filter	This option is used to filter out certain characters. The characters listed in the field will be permitted, while all others will be filtered out.

4.5.2 Restore Default Calling Rule

Navigate through **Outgoing Calling Rules > Restore Default Calling Rule**: This is where you configure Default Outgoing Calling Rules.

These are the default calling rules where they show us an example of the patterns and which all fields can be used as default.

4.6 Dial plans

Navigation: **SETUP→Dial plans**: This is where you configure Dial plans for the users.

Figure 16: Create New Dial Plan

A Dial plan is a collection of outgoing rules. Dial plan is assigned to users to specify the dialing permissions they have.

For Example: you might one Dial plan for local calling that permits the users of that Dial Plan to dial local numbers, via the “local” outgoing calling rule. Another user may be permitted to dial long distance numbers, and so would have a Dial Plan that includes both the “local” and “long distance” outgoing calling rules.

You have to create the Dial plan first before you create any user accounts to make your call successful. The Dial plan details are as follows.

Dial plan Name: The name of the user wish to see in that field of Dial plan.

Include Outgoing Calling Rules: when the outgoing Calling rules are created it displays here so that include it in the dial plan.

Include local contexts: Here the user can select the features which he wishes to use.

After all the changes, click on the save button. And don’t forget to click on apply changes button on the top navigation bar immediately after save button.

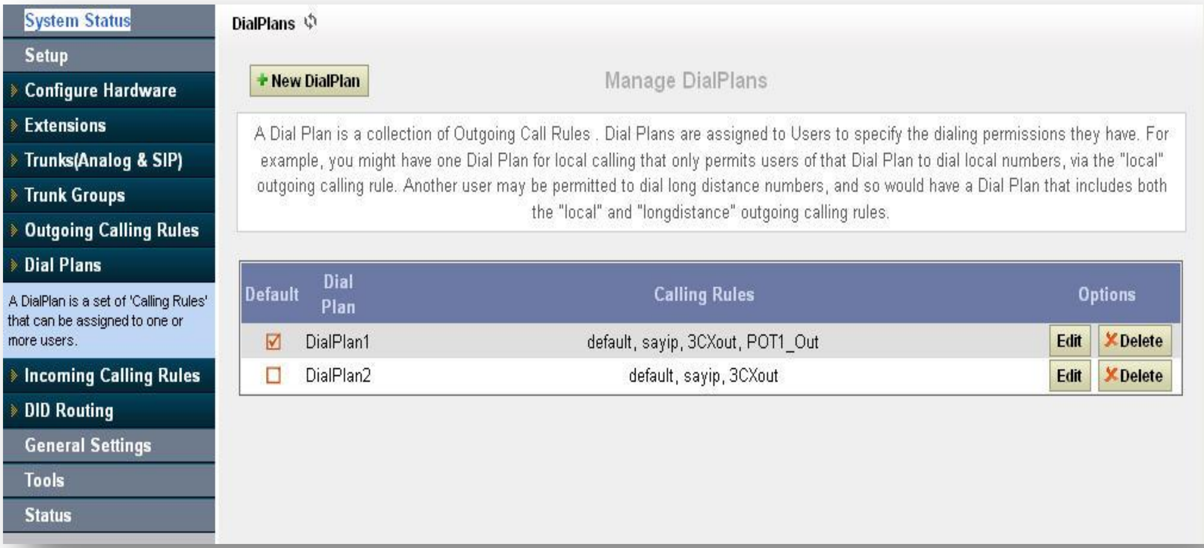

Figure 17: Dial Plan Result

4.7 Incoming Calling Rules

Navigation: **SETUP→Incoming Calling rule**: This is where you can create/edit /delete incoming calling rules.

Figure 18: New Incoming Rule

An Incoming Calling Rule is a rule which routes the incoming call to a phone number. The Incoming Calling Rule is a feature that enables incoming calls to be routed directly to selected stations without attendant assistance.

Incoming Calling Rule Configuration Parameters	
Trunk	Select the trunk to configure the inbound rule.
Time Interval	Select the time interval from the list specify the time for the trunk to use the inbound rule.
Pattern	All patterns are prefixed with the "_". X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately
Destination	Select the default destination for the inbound call • Extension • Trunk • Local Extension by DID

Figure 19: Incoming Calling Rules

4.8 DID Routing

Navigation: **SETUP →DID Routing**

Direct Inward Dial. A specially configured phone line from the telephone company allows for dialing inside a company directly without having to go through an attendant. A DID line cannot be used for out dial operation since there is no dial tone offered. However, it can be configured so an outside caller can reach an inside extension with a 7-digit number throughthe phone company's central office.

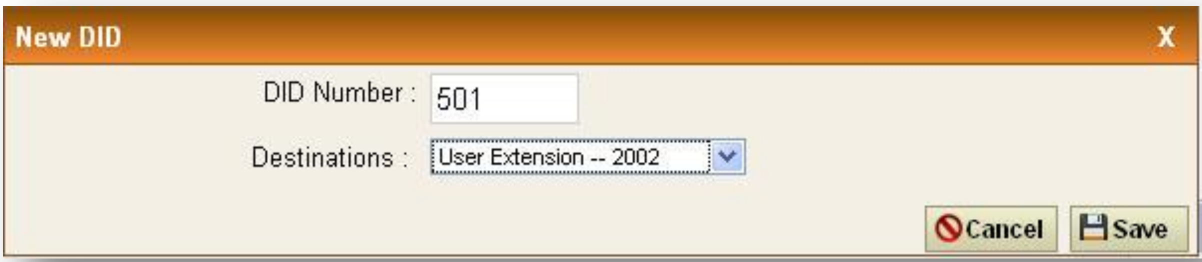

Figure 20: New DID

DID Configuration parameters	
DID Number	Enter the DID numbers provided by the VOIP Service provider
Destinations	Select the DID destination. Only the selected category can be reached by DID. • User Extension. • Trunk.

Figure 21: DID Calling Results

5. General Settings

5.1 Network Settings

Navigation: **Network Settings**: This is where you setup your Networking Configuration

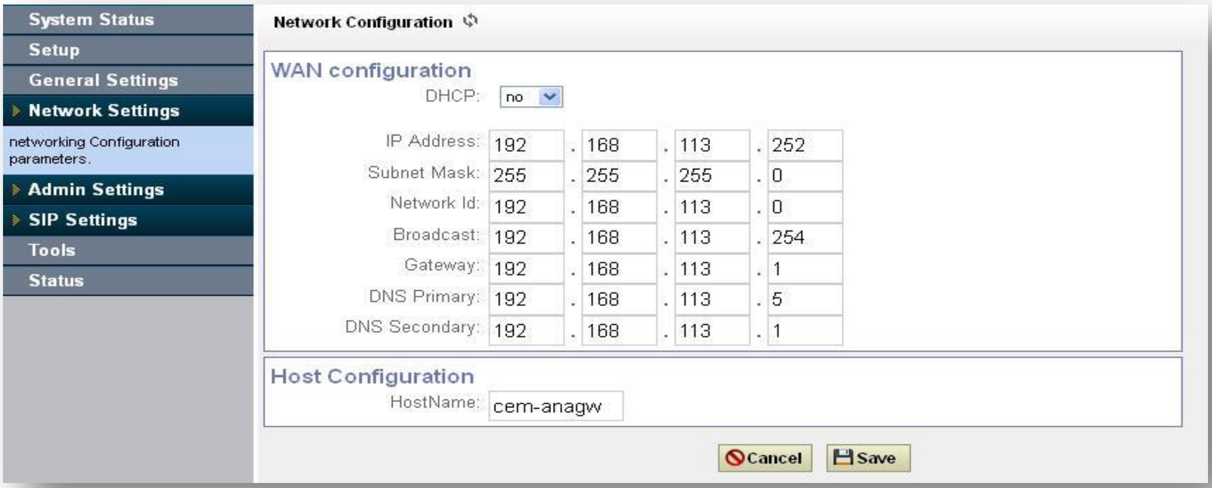

Figure 22: WAN Configuration

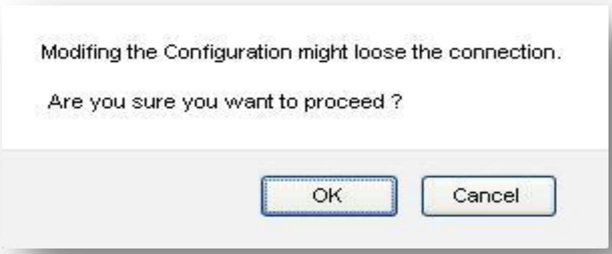

Figure 23: Modifying the confirmation

5.1.1 WAN Configuration

Please refer to the following tables for basic network configuration parameters

WAN Configurations	
IP Address Mode	Select DHCP, Static IP. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings.
Subnet mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.255.0
Network Id	your local network segment (Ex: 192.168.0.0)
Broadcast	Enter the broadcast address
Gateway	Enter the gateway IP address for static IP settings.
DNS Primary	Enter the DNS server 1 address for static IP settings
DNS Secondary	Enter the DNS server 2 address for static IP settings

5.1.2 Host Configuration

Host Configuration is used to manage your FXO/FXS VOIP Gateway Host Name.

Host Name: Used to name the device to identify inside the LAN network. This field is optional but may be required by some Internet Service Providers or system administrators. Click the save button and apply the Network Configuration.

5.2 Admin Settings

5.2.1 General Preferences

Navigation: **Admin Settings> General Preferences**: This is where you can configure the General Settings of the Gateway

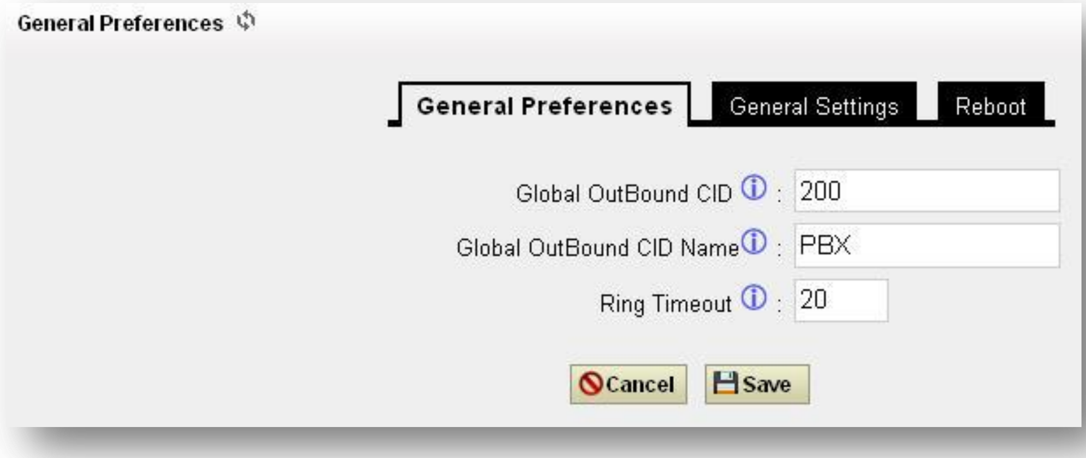

Figure 24: General Preferences

General Preferences	
Global Outbound CID	Configure the global CallerID used for all outbound calls when no other CallerID is defined with higher priority. If no CallerID is defined for extension or trunk, the global outbound CID will be used as CallerID.
Global Outbound CID Na	Configure the global CallerID Name used for all outbound calls. If configured, all outbound calls will have the CallerID Name set to this name. If not, the extension's CallerID Name will be used.
Ring Timeout	Configure the number of seconds to ring an extension, The default setting is 20.

VOIP Phone Digit Map: This option allows the administrator to define a global digit mapping string compatible with RFC 3435. There is no default setting and this option does not sync with the dial plan assigned to an individual user. The following examples should assist in writing an acceptable digit mapping string.

- [2-9]11 - Where calls beginning with digits from 2, 3, 4…. 9 followed by 11 are dialed immediately.
- 0T - Where calls beginning with the digit 0 followed by a pause equal to the "Digit Timeout" option.

- +011xxx.T - Where calls beginning with the + character, followed by 011 digits and then at least three more digits before any arbitrary number is matched, dialed after Digit Timeout is reached.
- 0[2-9]xxxxxxxxx - Where calls beginning with 0, followed by any digit from 2,3,4...9, followed further by 9 more digits are dialed immediately.
- +1[2-9]xxxxxxxx - Where calls beginning with the + character, followed by 1, followed by any digit from 2,3,4.....9, followed by 8 more digits are dialed immediately.
- [2-9]xxxxxxxxx - Where calls beginning with any digit from 2,3,4....9, followed by 9 more digits are dialed immediately.
- [2-9] xxxT - Where calls beginning with any digit from 2, 3, 4....9, followed by three more digits are dialed after Digit Timeout is reached.
- [2-9]11|0T|+011xxx.T|0 [2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT where each entry is separated by the | character. For more information, please refer to RFC 3435.

VOIP Phone Digit Timeout: The timeout variable is the number of seconds the phone will wait for each segment of a digit map expressed as an integer.

5.2.2 Gateway Settings

Navigation: **Admin Settings >General settings > GATEWAY Settings**: This is where you can configure the General Settings of the Gateway

Change Admin Password:

- Enter the new password and retype the new password to confirm. The new password field has to be at least 4 characters
- Click on “Update” and the user will be logged out.
- Once the web page comes back to the login page again, enter the new password to login.

Date &Time Settings: You can set the date and time of the Gateway either through Enabling NTP or by manual entry. If you are enabling NTP, select the time zone according to your country timing and enter the NTP server details.

Enable NTP	Enabling NTP allows you to set the Time zone and Date based on the NTP server provided
Time Zone	Select the proper time zone option so the Gateway can display the correct time

	accordingly.
NTP Server	Specify the URL or IP Address of the NTP server for the Gateway to synchronize the date and Time. For E.g.: pool.ntp.org

5.2.3 Reboot

Navigation: **Admin Settings > Reboot**: This is where you can configure the General Settings of the Gateway

The administrator of the FXO/FXS VOIP Gateway can remotely reboot the FXO/FXS VOIP Gateway by pressing the “Reboot” button at the bottom of the System management. Once done the following screen will be displayed to confirm reboot.

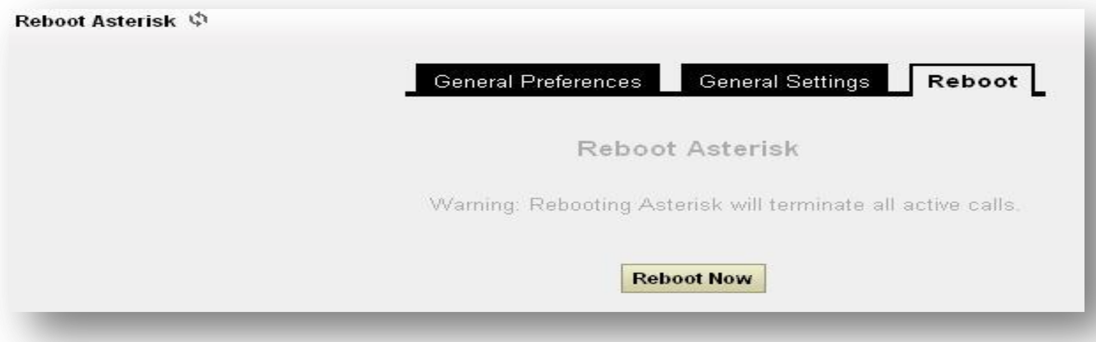

Figure 25: Reboot

FXO/FXS VOIP Gateway will take about 40 seconds to reboot. To reboot the FXO/FXS VOIP Gateway results in termination of active calls.

5.3 SIP Settings

5.3.1 General

Navigation: **SIP settings > general**: This is where you can configure the General sip settings.

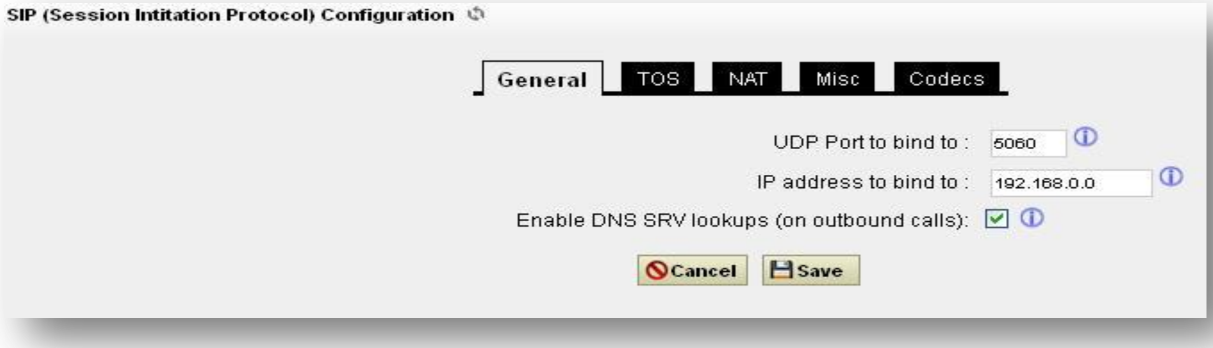

Figure 26: General SIP Settings

General	
Bind UDP Port	Configure the UDP port used for SIP. The default setting is 5060.
Bind IP Address	Configure the IP address to bind to. The default setting is 0.0.0.0, which means binding to all addresses.
Enable DNS SRV lookups	Select to enables DNS SRV lookups on outbound calls from the Gateway

5.3.2 TOS

Navigation: **General Settings**>**SIP Settings > TOS**: This is where you can configure the TOS sip settings.

Figure 27: TOS

The Details to be filled are given as:

TOS	
Generate In band Ringing	Configure whether the FXO/FXS VOIP Gateway should generate in band ringing or not. The default setting is "Never". • Yes: FXO/FXS VOIP Gateway will send 180 Ringing followed by 183 Session Progress and in-band audio. • No: The FXO/FXS VOIP Gateway will send 180 Ringing if 183 Session Progress has not been sent yet. If audio path is established already with 183 then send in-band ringing. • Never: Whenever ringing occurs, the FXO/FXS VOIP Gateway will send 180 Ringing as long as 200OK has not been set yet. In band ringing will not be generated even the end point device is not working properly
Server User Agent	Allows you to configure the user agent string for the FXO/FXS VOIP Gateway
DTMF Mode	Select DTMF mode to send DTMF. The default setting is RFC2833. If "Info" is selected, SIP INFO message will be used. If "In band" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, "RFC2833" will be used if offered, otherwise "In band" will be used. The default setting is "RFC2833".

5.3.3 NAT

Navigation: **General Setting > SIP settings > NAT**: This is where you can configure the NAT SIP settings.

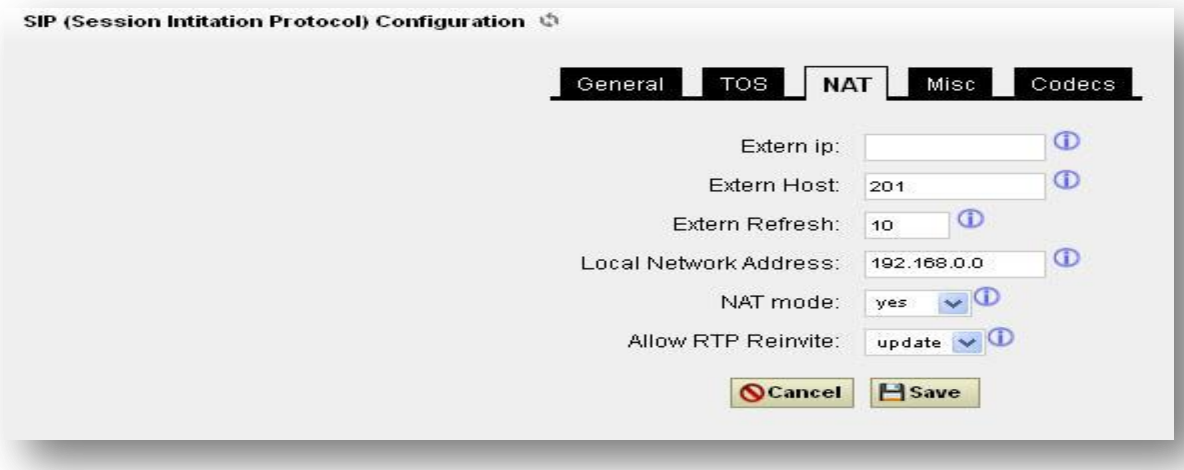

Figure 28: NAT Settings

Email Settings	
Extern IP	Configure a static address and port (optional) that will be used in outbound SIP messages if the Gateway is behind NAT.
Extern Host	you can specify an external host name and Asterisk will perform DNS queries periodically based on the External Refresh Interval
Extern Refresh	Configure the refresh interval for the external host (if used) The default setting is 10.
Local Network Address	Specify a list of network addresses that are considered inside of the NAT network. Multiple entries are allowed. If not configured, the external IP address will not be set correctly. A sample configuration could be as follows: 192.168.0.0/255.255.255.0
NAT Mode	This is a global NAT setting that will affect all peers and users. The default setting is “YES". • YES = Always ignore info and assume NAT • NO = Use NAT mode only according to RFC3581

	• NEVER = Never attempt NAT mode or RFC3581 support • ROUTE = Assume NAT, don't send report.
Allow RTP Reinvite	If enabled, the Gateway will try to redirect the RTP media stream (audio) to go directly from the caller to the Callee. • Yes : Allow RTP media direct • No NAT: Allow media path redirection (Reinvite) but only when the peer is not being behind NAT. The RTP core can detect if the peer is behind NAT or not based on the IP address where the media comes from. • Update: Use UPDATE for media path redirection, instead of INVITE. Some devices do not support this (especially if one of them is behind NAT).

5.3.4 Misc

Navigation: **General Settings > SIP settings > Misc:** This is where you can configure the miscellaneous SIP settings.

Figure 29: Misc

The details to be filled are given as:

FAX Settings	
FAX Format	Select the FAX format either in TIFF or PDF
T.38 fax (UDPTL)	Enables T.38 Fax Mode otherwise pass-through mode is enabled
Out Bound SIP Registration Settings	
Register	Register as a SIP user agent to a SIP proxy (provider)
Register Time-Out	Configure the register retry timeout (in seconds). The default setting is 20.
Register Attempts	Configure the number of registration attempts before the Gateway gives up. The default setting is 0, which means the Gateway will keep trying until the server side accepts the registration request.

5.3.5 Codecs

Navigation: **SIP settings > Codec's**: The following Audio & Video codec's are supported in FXO/FXS VOIP Gateway.

Figure 30: Codecs

- G711 u-law
- G711 a-law
- G.726(32)
- G.729

Select the codec's from the list by enabling it.

Figure 31: Create new config file

Tools

6. Tools

6.1 Back Up:

The Gateway configuration can be backed up locally or via network. The backup file will be used to restore the configuration on Gateway when necessary.

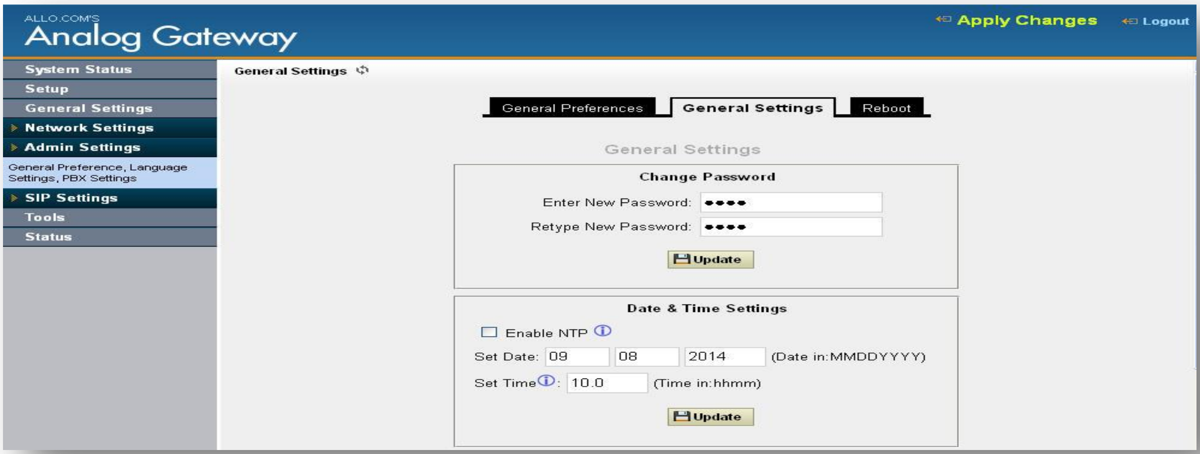

Figure 32: General Settings

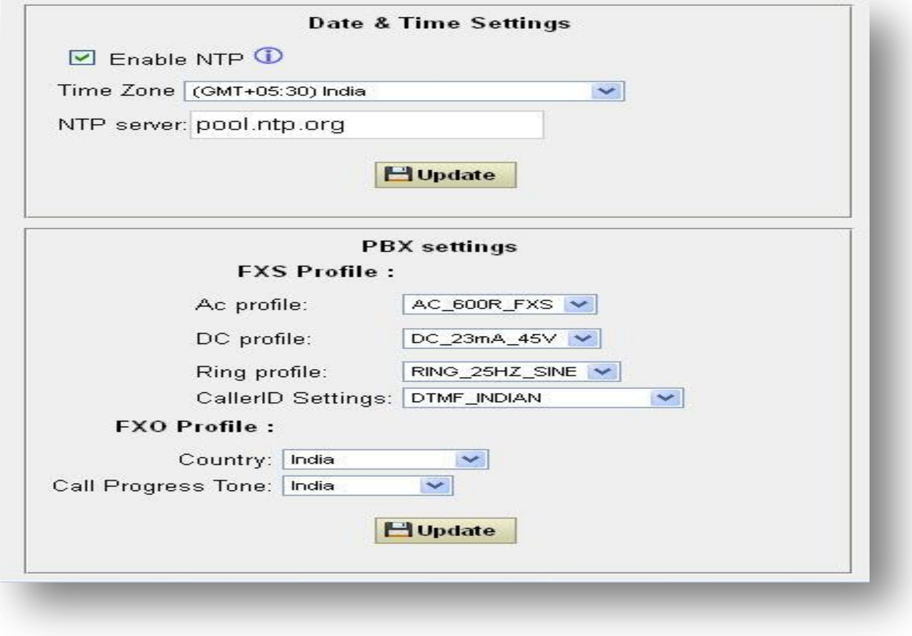

Figure 33: PBX settings

Users could backup the Gateway configurations for restore purpose by clicking on “Create New Backup” under **Tools > Back Up**.

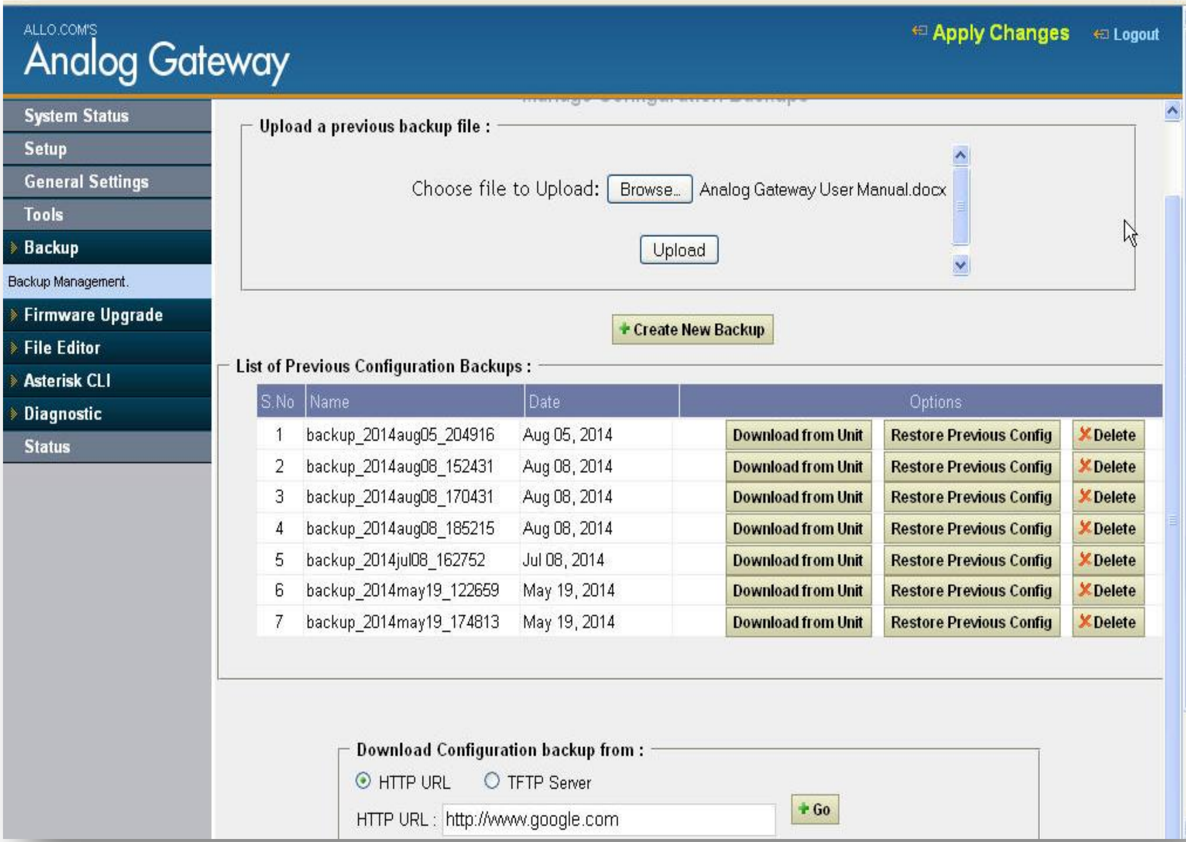

Figure 34: Backup Results

Once the backup is done, the list of the backups will be displayed with the date and time on the web page. Users can download, restore, or delete it from the Gateway internal storage or the external device.

Besides local backup, users could back up the configuration a remote server via TFTP/ HTTP protocol under Web GUI->**Tools > Back Up**.

RESTORE CONFIGURATION FROM BACKUP FILE

To restore the configuration on the Gateway from a backup file, users could go to Web GUI->**Tools** ->**Backup**. A list of previous configuration backups is displayed on the web page. Users could click on "Restore Previous config" of the desired backup file and it will be restored to the Gateway.

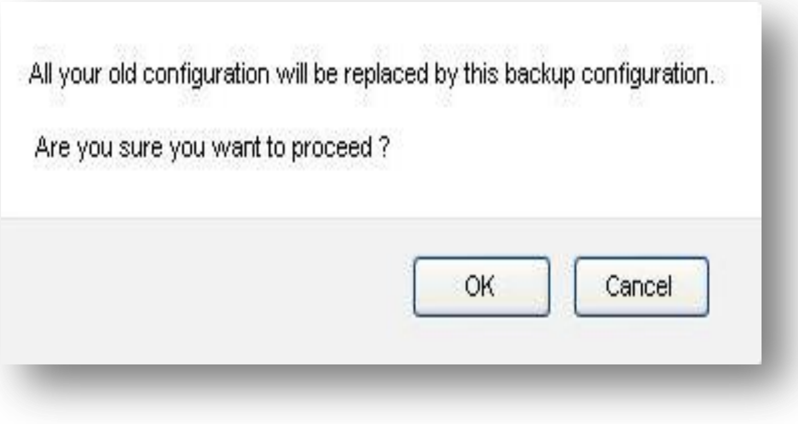

Figure 35: Backup Configuration

If users have other backup files on PC to restore on the Gateway, click on "Browse" first and select it from local PC to upload on the Gateway, and then click “Upload”.

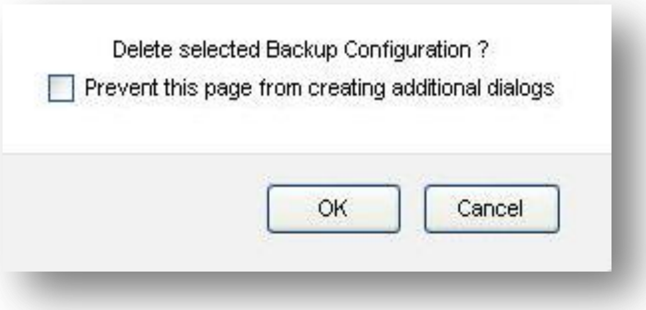

Figure 36: Delete Backup configuration

6.2 Firmware Upgrade

The Firmware Upgrade page allows you to update the Gateway with the latest release available, which can contain key updates, added functionalities and bug fixes. When a new firmware release is available, download it and save to your local PC. Then, browse for the file, and click the Upload button. Please contact ALLO support team for the latest firmware upgrade (http://support.allo.com/)

Figure 37: Firmware up gradation

- While upgrading the firmware, please make sure that there won't be power or network disturbances and also make sure to take back-up of configuration if any.
- During the firmware upgrade All the Front LED's start blinking.

6.3 File Editor

Navigation: **File Editor**: This is where you can edit the configuration files or verify whether the configuration files are updated.

In the File Editor filed mention the configuration file you want to view. It displays the contents of the file. Here you can view modify the contents of the file. If the user wishes to create a new configuration file it can be done using the new file.

6.4 Asterisk CLI

Navigation: **Tools >Asterisk CLI**: This page gives an easy access to the user to execute the commands of the CLI. An Example of the "sip show peers" displays all the Gateway user extensions with the registration status.

Figure 38: Asterisk CLI

6.5 Diagnostics

On the Gateway, users could capture traces, ping remote host and trace route remote host for troubleshooting purpose under Web GUI->**Tools > Diagnostic**.

Ping

Enter the target host in a host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.

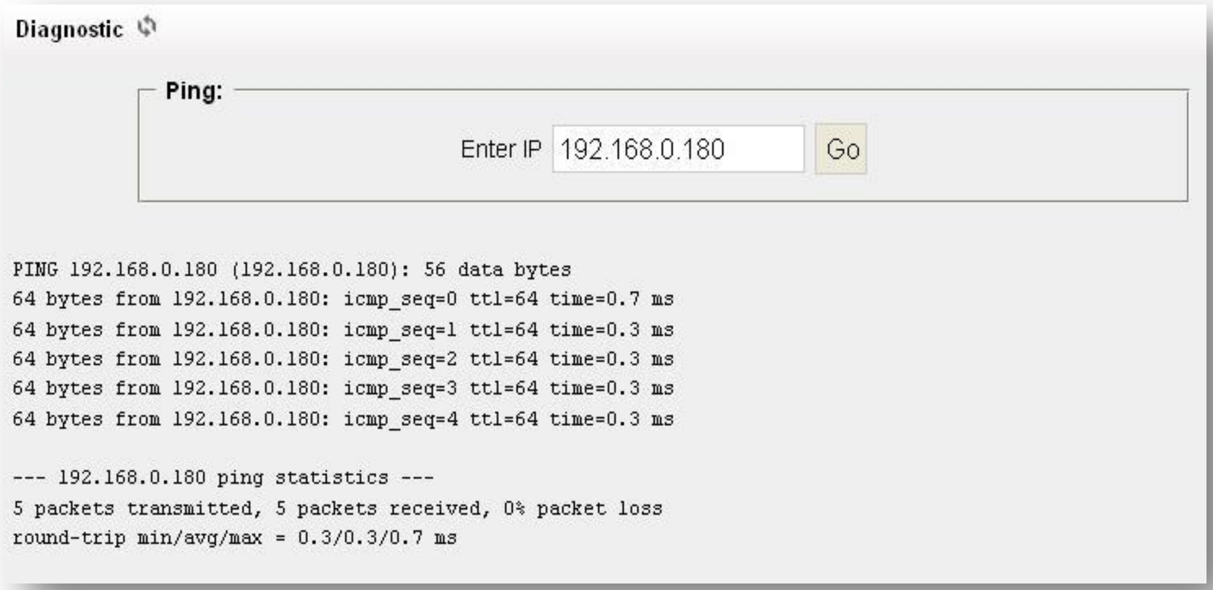

Figure 39: Diagnostic Ping Result

Status

7. Status

7.1 Call Detail Records

Navigation: **Call Detail Records**: This is where you can create Call Reports

	Start time	Duration	Source	Destination	Caller ID	Disposition
26	2000-01-01 00:47:02	0:00:15	1001	9999	1001	ANSWERED
27	2000-01-01 00:46:44	0:00:14	1001	9999	1001	ANSWERED
28	2000-01-01 00:46:17	0:00:14	1002	8999	1002	ANSWERED
29	2000-01-01 00:46:08	0:00:02	10011	2001	"CEM" <10011>	NO ANSWER
30	2000-01-01 00:46:00	0:00:04	10011	2001	"CEM" <10011>	NO ANSWER
31	2000-01-01 00:45:57	0:00:15	1002	8999	1002	ANSWERED
32	2000-01-01 00:45:51	0:00:02	10011	500	"CEM" <10011>	ANSWERED
33	2000-01-01 00:45:45	0:00:04	10011	2001	"CEM" <10011>	NO ANSWER
34	2000-01-01 00:45:39	0:00:15	1002	8999	1002	ANSWERED
35	2000-01-01 00:45:37	0:00:02	10011	500	"CEM" <10011>	NO ANSWER
36	2000-01-01 00:45:36	0:00:05	10011	2001	"CEM" <10011>	NO ANSWER
37	2000-01-01 00:45:29	0:00:02	10011	500	"CEM" <10011>	ANSWERED
38	2000-01-01 00:45:27	0:00:04	10011	2001	"CEM" <10011>	NO ANSWER
39	2000-01-01 00:45:25	0:00:00	1002	8999	1002	NO ANSWER
40	2000-01-01 00:44:55	0:00:25	1003	7999	1003	ANSWERED
41	2000-01-01 00:44:48	0:00:02	10011	2001	"CEM" <10011>	NO ANSWER

Figure 40: Call Detail Report

To create a new Report, select the inbound calls, outbound calls, internal calls, and External calls. A list with call details will display in the Call Reports section.

By clicking on the delete button at the bottom Entire call details records (CDR) of the Gateway are cleared

CDR can also be filtered by selecting inbound calls, outbound calls, internal calls and External calls.

You can select the number of lists to show in the listings by selecting the right drop down box. By clicking on the Previous and Next button you can see the list pages in the next and the previous pages.

You can even download the CDR in CSV format by clicking "Download" button, which makes the Download file to display the top of the GUI page. Whereby right Click on the "Download File" link and download the CDR using the 'Save Link As'.

7.2 Active Channels

It displays the current Active Channels on the Gateway with the options to Hang-up. When calls are in progress since there is always refreshing Active Channels in 10 seconds the Current Active Channels on the Gateway are displayed.

Figure 41: Active Calls

7.3 System Info

The Gateway status can be accessed via Web GUI->**Status**->**System Info**, which displays the following system information.

- **General**
- **Network**
- **Disk Usage**
- **Memory Usage**

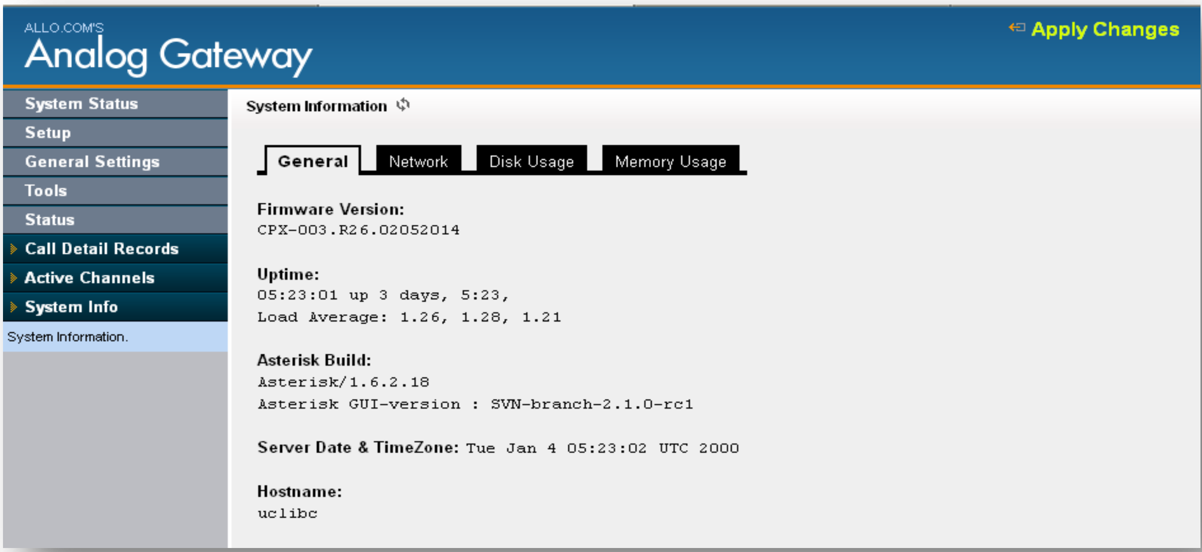

Figure 42: General System Information

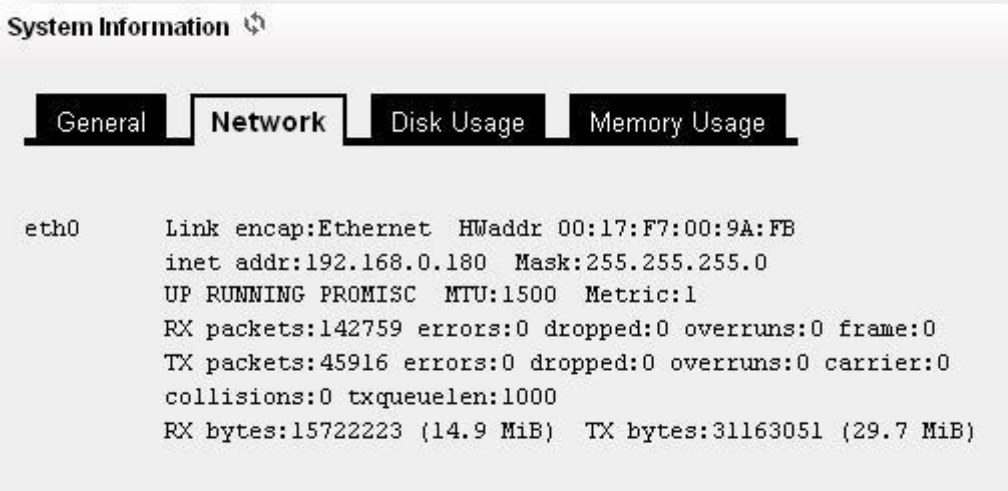

Figure 43: Network Information

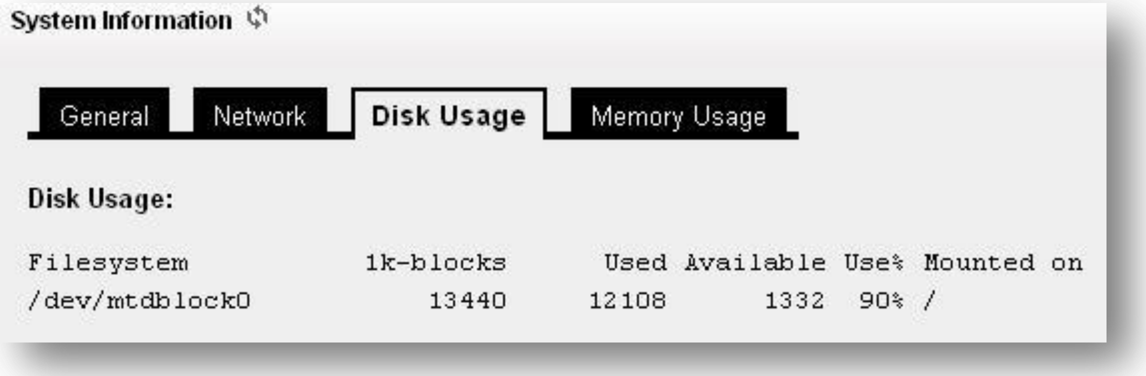

Figure 44: Disk Usage

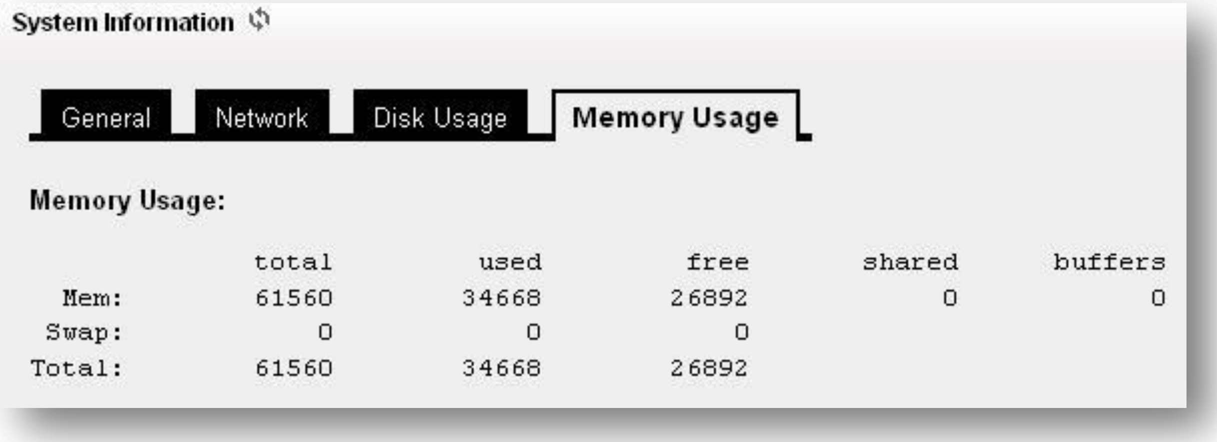

Figure 45: Memory Usage

General	You will have a summary of your general information, such as **Firmware Version, Uptime, Asterisk Build, Server Date & Time Zone, and Hostname.**
Network	Network Status you will have a summary of your Network information, such as Hostname, WAN IP Address, Subnet Mask, WAN MAC Address and Default Gateway (you may refer to Settings > Network Settings for more info).
Disk Usage	Disk you will have a summary of Disk usage and Disk free space available on the file system.
Memory Usage	Memory Status you will see total memory resources, including RAM usage, Compact Flash usage.

8. APPLY Changes

Navigation: **APPLY CHANGES (On the right top of the web GUI)**

This is the button which you must press after adding / editing / deleting such things as Extensions , modifying settings such as General Settings, VOIP Accounts, Network Settings, DID Routing, Firmware Upgrades, and other System Settings.

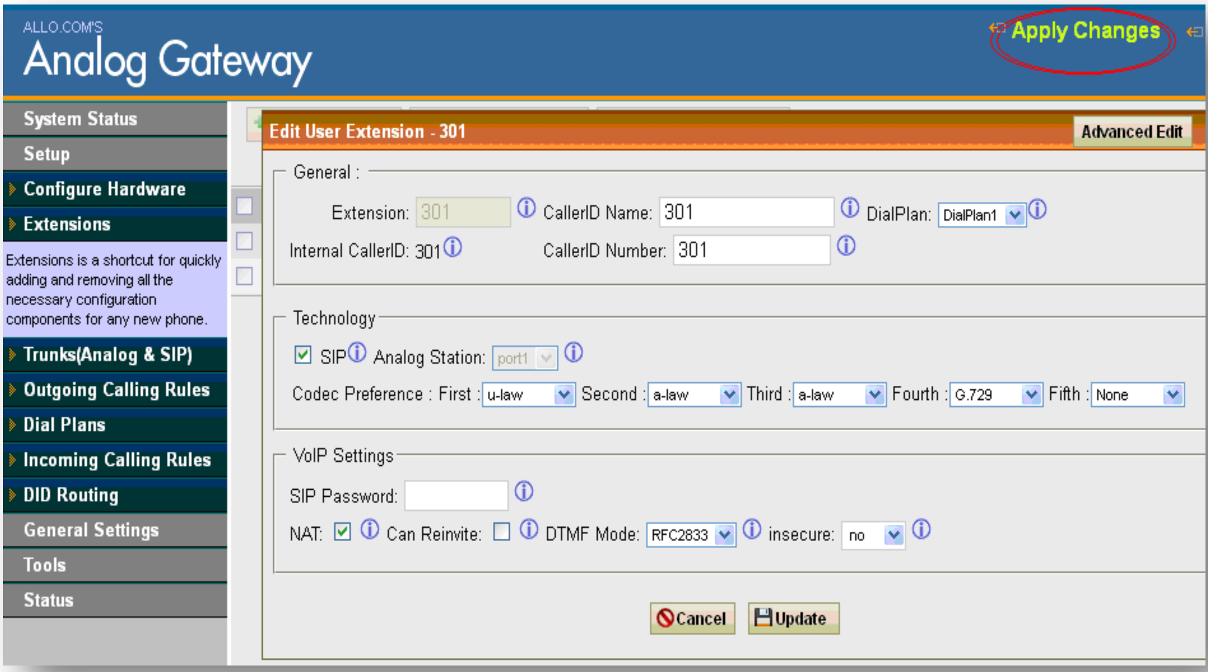

Figure 46: Apply changes

FAQs

9. Frequently Asked Questions (FAQ)

1. How to access GUI of factory default Analog Gateway?

Power on the device and connect RJ45 cross cable on one end to the Ethernet port of the card and the other end to your PC.

1) Keep your PC IP network settings in manual mode and assign 192.168.113.xxx with subnet mask 255.255.255.0
2) Open the Mozilla Firefox browser and enter http://192.168.113.252:8088 factory default IP of the card.
3) Login using username: admin and password: admin

2. How to change the network settings of analog gateway?

Connect cross over cable from FXO/FXS card to your PC. Keep your PC network configuration in manual mode and set 192.168.113.xxx.

Open Mozilla Browser and enter http:192.168.113.252:8088.
Username: admin
Password: admin

1. Open the GUI of the FXO/FXS card
2. Go to network settings
3. If your network has enabled DHCP server then set DHCP "yes"
4. If DHCP server is not running in your network then set DHCP "no" and assign a static IP to the device.
5. Click Save and Connect Straight Ethernet cable from FXO/FXS card to your network switch.
6. Connect an analog phone to FXS port, FXO/FXS card and press"**" and device will announce an IP address.
7. Open that IP in Mozilla Firefox browser http://XXX.XXX.XXX.XXX:8088

3. How to create SIP trunk in analog gateway?

1. Go to Trunks (Analog & SIP) → VOIP Trunks.
2. Click New SIP trunk and select type as "SIP"

Figure 47: Create New SIP Trunk

3. Select context naming based on username it will create context based on the username
4. **Provider name:** Give an unique name for this trunk
5. **Hostname:** Add the IP address of the VoIP trunk service provider of IPPBX
6. **Host port:** Enter the default SIP port: 5060
7. **Username:** It would be the user account ID provided by your service provider or IPPBX
8. **Password:** It authenticates the user account provided by VoIP service provider or IPPBX.
9. **DID routing:** Enable DID Routing if you are accepting DID numbers through this SIP trunk. Once you enable this DID Routing, all calls will route to DID Routing table.
10. Once check the Registration box, to send the registration request to a VoIP service provider or IPPBX.
11. Click Save and Apply Changes.

4. How to create outgoing call rule creation in Analog Gateway?

1. Open GUI of the Analog card and go to Outgoing Calling Rules
2. Click “New Ruling Rule”

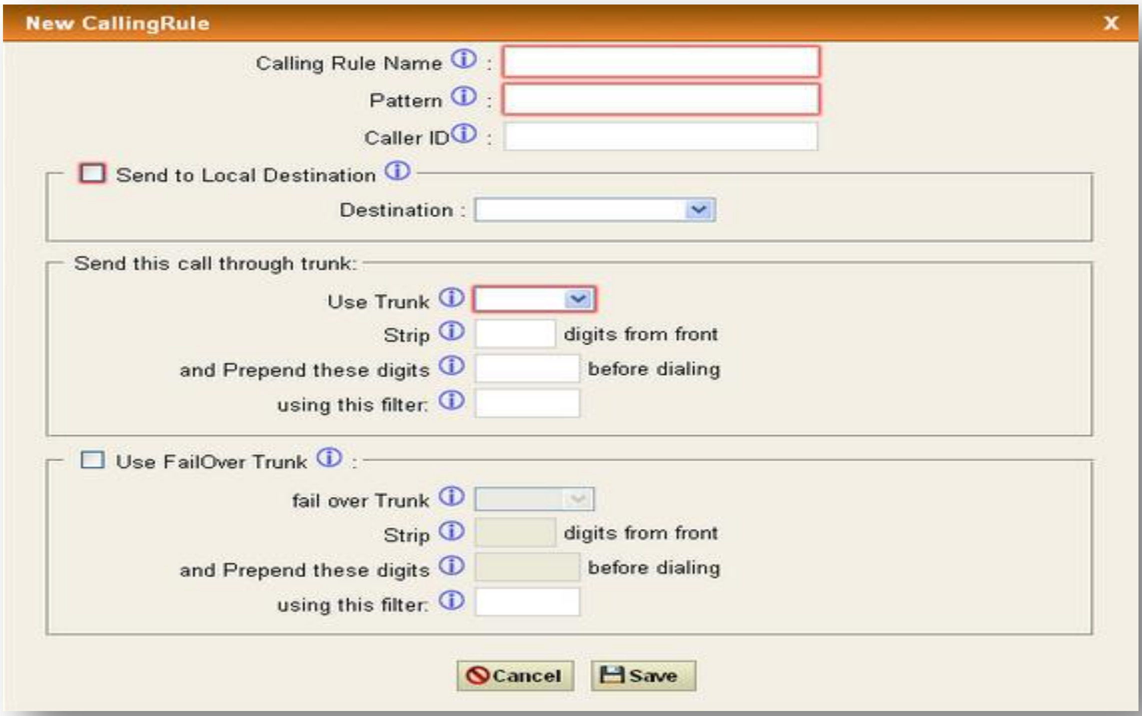

Figure 48: New Calling Rule

3. Provide a name for the rule
4. Pattern should start with followed by "X." and it should look like"_X." And you can make prefix based pattern as well. For an Example: _9X. And Strip: 1 the outgoing calls with prefix 9 will obey this rule and pass through the Trunk which has selected in "Use Trunk" Place.
5. **Use Trunk:** Select the appropriate trunk for which this rule should work.
6. **Strip:** The above example clearly explained how many numbers should be stripped from the called number.
7. **Prepend Digits:** what are the digits should be prefixed for the called number.
   You can leave rest of the fields as default.
   Click save, Go to Dial Plans and add this new outgoing rule to the existed dial plan
   Click Save and Apply Changes.

5. How to create SIP Trunk?

1) Open GUI of the FXO/FXS cards and Go to Trunks (Analog/SIP) and click on VOIP Trunks and create New SIP trunk.

Figure 49: Create New SIP Trunk

2) Select SIP type and enter all the details as shown below

Figure 50: Create New SIP Trunk

If you want DID routing, you can check the Box. All the calls are coming inside the analog card through this 3CX trunk will go to DID routing table.

3) Click SAVE and APPLY CHANGES, you can see the registered status in the System Status page.
4) Create outgoing Calling Rule in FXS/FXO card using this trunk to reach the 3CXPBX and add it to Dial Plan.

6. How to create the new incoming calling rule?

Open the GUI and go to Incoming Calling Rules

Create New Incoming Rule

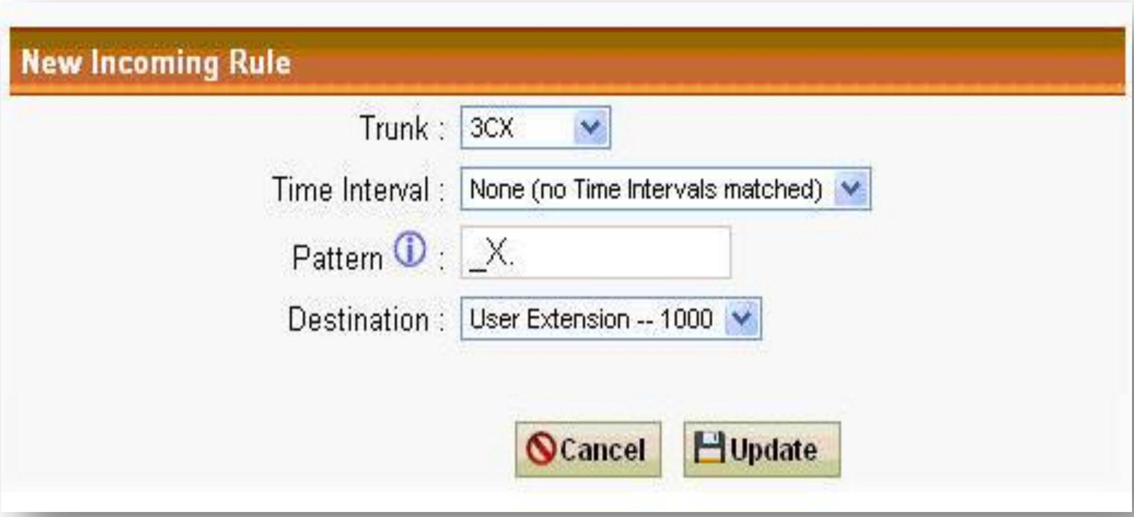

Figure 51: New Incoming Rule

2) Select the Appropriate trunk either FXO trunk or VoIP trunk from the Trunk dropdown box.

3) Select the default time interval

4) Pattern: "_X." will accept the calls

5) Destination: Select the destination for the incoming call.

Click Update and Apply Changes.

7. How to create the FXO trunk?

1) Connect PSTN lines on FXO ports and Go to Trunks (Analog & SIP)
2) Create New Analog Trunk

Figure 52: Create FXO Trunk

3) Select the port and give a name to that trunk and click Add.
4) Create outgoing calling rules for this trunk
5) Add that outgoing rule into existed dial plan.

Click Save and Apply Changes.

8. How to make basic communication between the analog cards with IPPBX?

In order to make basic communication between the analog card with IPPBX, Follow the below steps.

1) Set up the SIP trunk between the analog card and IPPBX.
2) Assign extension numbers for FXS ports
3) Create an outbound rule for the SIP trunk and add it to dial plan
4) Set up the incoming calling rule for that SIP trunk and route the call to one of the FXS ports.

10. Explain Trunks Bridging (Sending call from one Trunk to another Trunk)?

Trunks bridging means, if the call is coming from one trunk then we can send the call directly to another trunk.

Scenario No 1: Call is coming from FXO trunk and sending to SIP trunk

1) Create an incoming calling rule for FXO trunk Pattern as _X.
2) You can set the default option for a time interval
3) Destination will be the SIP trunk.
4) Add the FXO line number in the “Number “field.

Scenario No 2: Call is coming from SIP trunk and sending to SIP trunk

Let us say, if you have two SIP trunks like SIP trunk A and SIP trunk B

1) Create an incoming calling rule for SIP trunk A. Pattern as _X.
2) You can set the default option for a time interval
3) Destination will be the SIP trunk B.
4) **Strip Digits:** Number of the digits will be trimmed from the dialed number.
5) **Prepend Number:** Prefix to add the dialed number.

Appendix

10. APPENDIX-A:

Glossary

Term	Definition
ATA	Analog Telephone Adapter: Used to connect a standard telephone to a high-speed modem to facilitate VOIP or fax calls over the Internet.
DHCP	Short for Dynamic Host Configuration Protocol, a protocol for assigning dynamic IP addresses to devices on a network. With dynamic addressing, a device can have a different IP address every time it connects to the network.
WAN	Wide Area Network. A computer network that spans a relatively large geographical area. Typically, a WAN consists of two or more local-area networks (LANs).
DNS	The Domain Name System is the system that translates Internet domain names into IP numbers. A "DNS Server" is a server that performs this kind of translation.
FXO	In telecommunications, a Foreign Exchange Office, or FXO, is a telephone signaling interface that receives POTS, or "plain old telephone service".
FXS	Foreign Exchange Station is the interface on a VOIP device for connecting directly to phones, faxes, and CO ports on Gateways or key telephone systems.
GATEWAY	A network point that acts as an entrance to another network
IP ADDRESS	Every machine that is on a network (a local network, or the network of the Internet) has a unique IP number [four sets of numbers divided by period with up to three numbers in each set. (i.e. 192.168.0.100)] - If a machine does not have an IP address it cannot be on a network.
LAN	Local Area Network: A LAN is a group of computers and associated devices that share a common communications line or wireless link and typically share the resources of a single processor or server within a small geographic area (for example, within an office building).
NETMASK	Used by the TCP/IP protocol to decide how the network is broken up into sub-networks (ex: 255.255.255.0).

PBX	Private Branch Exchange: An in-house telephone switching system that interconnects telephone extensions to each other, as well as to the outside telephone network.
PROXY	A server that receives requests intended for another server and that acts on the behalf of the client behalf (as the client proxy) to obtain the requested service. A proxy server is often used when the client and the server are incompatible for direct connection. For example, the client is unable to meet the security authentication requirements of the server but should be permitted.
SIP	Session Initiation Protocol: An application-layer control protocol, a Signaling protocol for Internet Telephony. SIP can establish sessions for features such as audio/videoconferencing, interactive gaming, and call forwarding to be deployed over IP networks thus enabling service providers to integrate basic IP telephony services with Web, e-mail, and chat services. In addition to user authentication, redirect and registration services, SIP Server supports traditional telephony features such as personal mobility, time-of-day is routing and call forwarding based on the geographical location of the person being called.
PSTN	Public Switched Telephone Network: This is defined as the regular telephone network services.
VOIP	Voice over Internet Protocol. The technology used to transmit voice conversations over a data network using the Internet Protocol. Such data network may be the Internet or a corporate Intranet.
WAN	Wide Area Network. A computer network that spans a relatively large geographical area. Typically, a WAN consists of two or more local-area networks (LANs).

Thank you for choosing

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