Mega PBX + PRI

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

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1. Product Introduction

1.1 Overview

The Allo.com's MegaPBX, a SIP-based, affordable, feature-rich converging communication platform designed to meet the communication requirements for small to medium sized enterprises.

The MegaPBX provides the cutting-edge IP-based communications that businesses demand while leveraging existing infrastructure and providing a smooth transition into IP telephony. Based on open standard SIP, the MegaPBX can easily integrate into and interoperate with other components of your existing communications network while providing a rich set of features to reduce costs and increase productivity.

The MegaPBX also includes a Setup wizard and an intuitive user interface that allows users to quickly configure extensions, Voicemail, fax mail, Voicemail& fax to email, conference bridges and other enhanced features can be turned on with minimal effort via the web configuration interface.

Here, 4 models are available and all supporting up to 200 extensions and 50 simultaneous calls, 30 people conference room with full features (invite, schedule, email, kick, mute...). FXS ports for FAX machines only.
MegaPBX-PRI supports ISDN PRI protocol and adopts standard T1/E1 trunk interface to realize docking with traditional PBX /PSTN.

Model No - aMP1

Equipment Packaging

1) One MegaPBX unit

2) One 12 Volt power adapter

3) One Ethernet cable

2. Getting Started With MegaPBX

Initial Setup of IP/PRI/FXS/BRI PBX

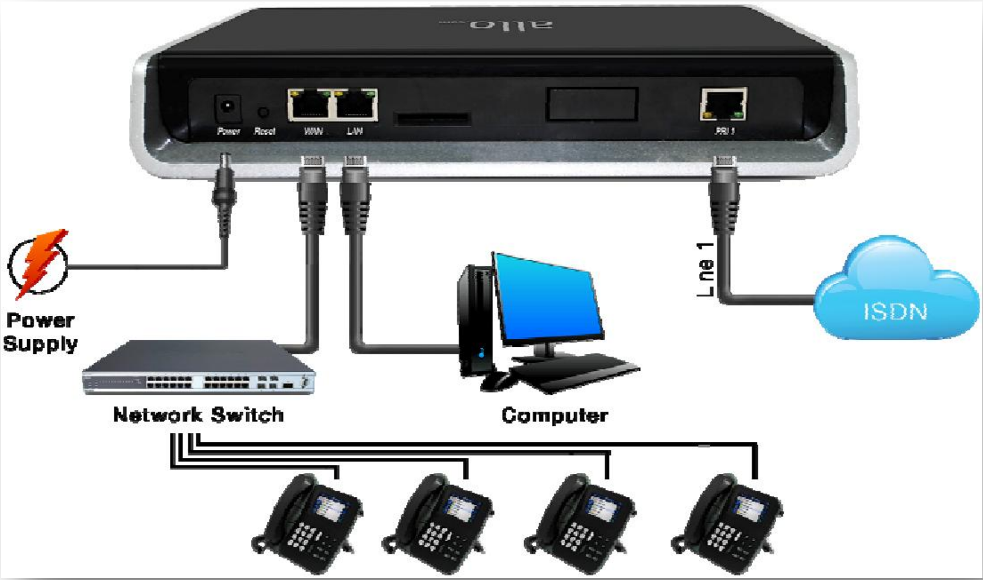


Figure 1: Initial Setup

1. Plug one end of the RJ45 Ethernet cable into your Network Switch
2. Plug the other end of the RJ45 Ethernet cable into the WAN port of the MegaPBX
3. Connect a PC to the LAN port of the MegaPBX; Enable the DHCP option in the Network Settings of the PC
4. Plug the Power Adapter included into an available power outlet
5. Plug the other end of the Power Adapter into the “DC-IN” port of the MegaPBX
6. The MegaPBX will power up (Boot up time takes about 160secs)

 Use Straight-through Ethernet cable to connect between the MegaPBX to Router/Switch/PC

2.1 Hardware Setup

2.2 Equipment Structure

MegaPBX-PRI

Figure 2: MegaPBX-PRI

Interface	Description
Power	Connect the power adapter, 12VDC, 3.5A
Reset	Reset button for factory default.
WAN	Standard 10/100BASE-TX Ethernet Interface for WAN
LAN	Standard 10/100BASE-TX Ethernet Interface for LAN
Memory Card	8GB Storage for voice mails and voice files, IVR
PRI 1	E1/T1 ports with line link LED indicators (Orange and Green)

Notification LEDs (On the Front Panel of the Gateway):

LED 8	LED 2	LED 1	WAN	LAN	Power
Blue On- PRI 1 link is up (No Alarm)	Blue On- System Ready Blue Blink- Factory Reset	Blue ON- If the SD Card Mounted properly	Orange Blinks- WAN link is up.	Orange Blinks- LAN link is up.	Orange On- Power is on.

2.3 Access the web GUI:

MegaPBX-PRI Web GUI can be accessed either through WAN or LAN interface. Steps to Access the GUI during the initial setup through LAN interface:

1. Make the setup as mentioned in the hardware setup
2. Change the Network setting of the PC is set in automatic mode (i.e. DHCP mode). An IP address will be accessed to the PC in manual mode (i.e. Static IP mode). Assign the IP address to the PC in the range of 192.168.113.xxx series (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 http://192.168.113.2 (Default LAN IP address) to open the login page of MegaPBX-PRI Graphical Interface.

Figure 3: Login Page

4. Login using the default username & password (Default: Username: admin; Password: admin). Successful login takes you to the Dashboard page. Observe the WAN IP address on the dashboard, this will be used to access the GUI from the WAN interface.

After successful login we get a MegaPBX-PRI Home Dash Board. To guarantee the system safety, When login for the first time, we need to modify Password. For modifying the password go to TOOLS**->Admin Account Option**

The Recommended web Browser to access GUI is Mozilla Firefox only.

If your network is not enabled with DHCP server, configure the WAN port IP address manually in the SETTINGS > Network Settings section as per your requirement.

WARRANTY

Hardware Warranty: 1 year

If the MegaPBX PRI 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 MegaPBX PRI. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

3. Setting up Features

Setting up your browser for working with MegaPBX-PRI is simple. In order to run this application appropriately the following settings are to be configured.

3.1 System Dashboard

ALLO MegaPBX-PRI Dash Board summarizes the MegaPBX-PRI status with a graphical display. Detailed status of an individual entity is available under the Status Tab or it can be directly accessed by clicking on more.

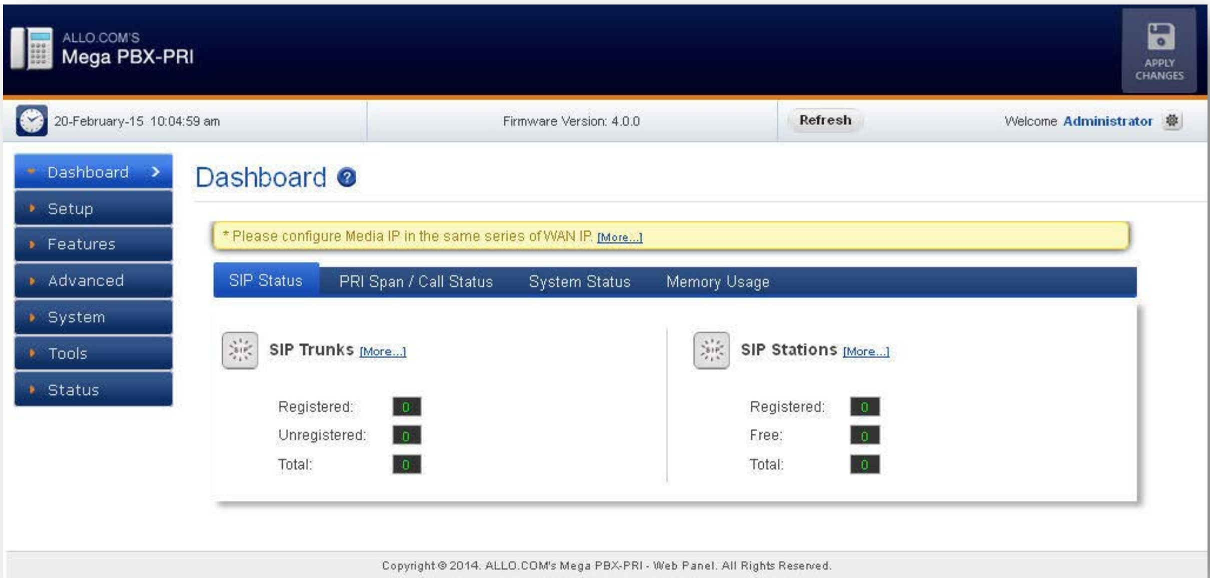


Figure 4: SIP Status

Please change the default administrator password to alphanumeric, to prevent hacking.

4. Setup

4.1Extensions

Here, user can configure Feature (*) Code for different Call Features.

4.1.1 SIP Extensions

Navigate through **Setup** > **Extensions** > **SIP Extensions**

SIP Extensions are the unique number mapped to person that can be reached and be able to place calls.

Figure 5: SIP Extensions

Click Add New, to create an Extension

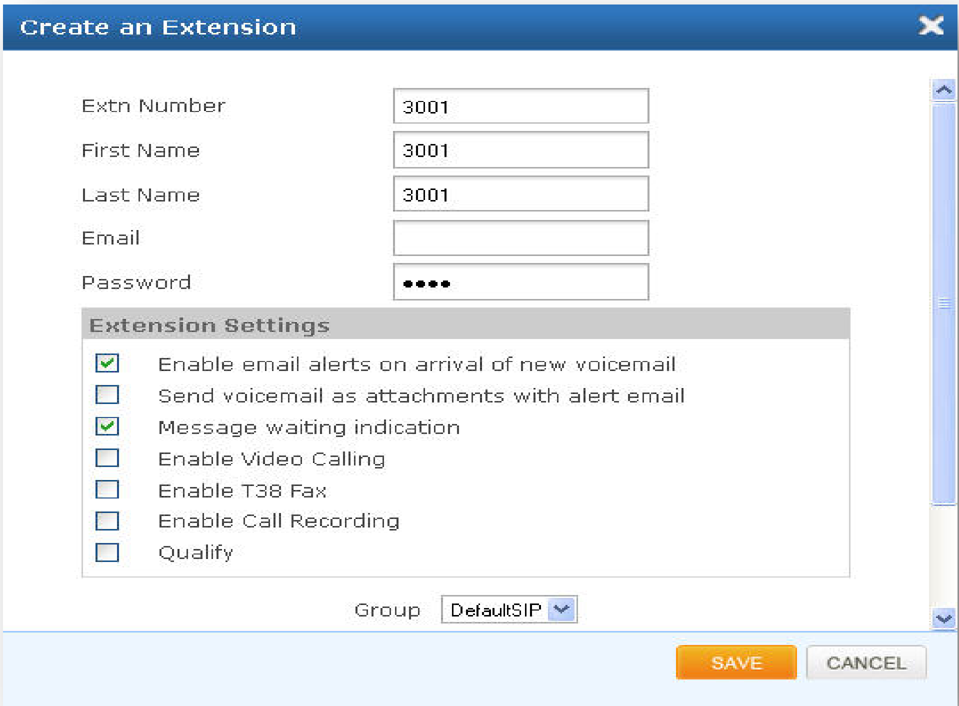


Figure 6: Create an Extension

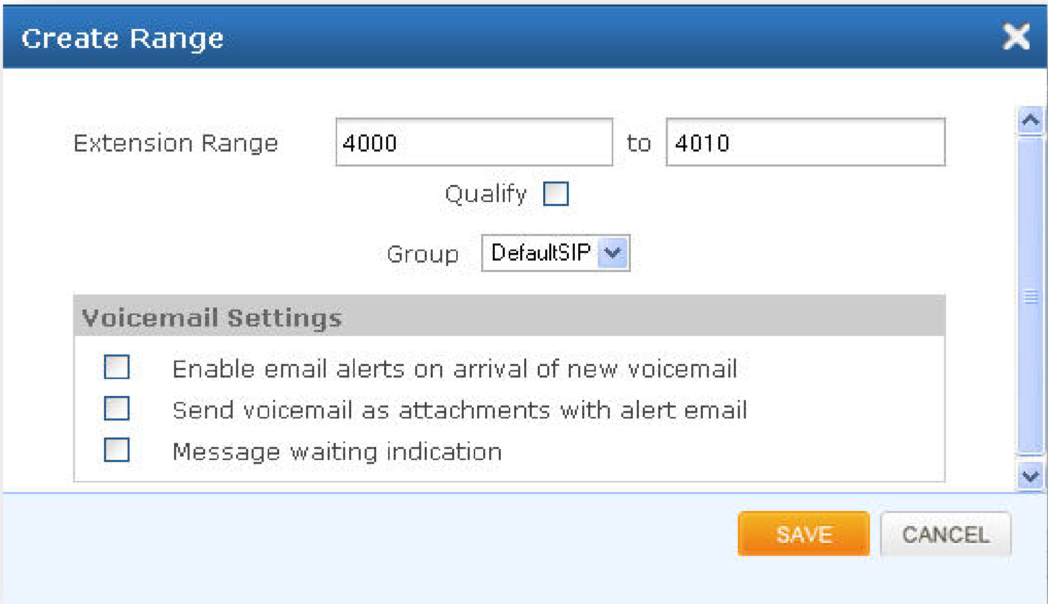


Figure 7: Create Range

Extn Number	Extension Number of endpoint (e.g.: IP phone) will use to authenticate with the MegaPBX, eg: 4000.
First Name	A character based first name for the user eg:"Bob".

Last Name	A character based last name for the user eg:"Jones".
Email	E-mail address for the user eg:"Bobjones@xyz.com".
Password	Password used to authenticate your phone and voicemail pin.
Extension Settings	This allows the user to either enable or disable email alerts or message waiting indication.
Add Extension Range	Range of Extensions can be added at once with required voicemail settings, Eg: from 4000 to 4010.

Show/ Hide Advance Options

Advance options, allows enabling the Features, Codec Configuration and Dialout Settings for a specific user. Default SIP group is disabled when Advanced Options is enabled.

Advanced Options	
DTMF	Set default DTMF mode for sending DTMF digits. Options: • INBAND – sent along with audio (requires 64 kbit codec - alaw, ulaw) • INFO – sent as SIP INFO messages • RFC2833 – sent as RTP packets • AUTO – System automatic selects the mode. Uses RFC2833 if offered, inband otherwise. Default: **AUTO**
Features	Enable or Disable the desired features like voicemail, call queue, conference, and call back, call pickup, call park, etc…
Codec Configuration	Choose the available Codecs and set priority in the order in which Mega PBX should prefer to send and receive audio. Supported codecs are alaw, ulaw, G.729, G.722
Dialout Settings	Allow or deny the preferred dial out plans for user.

4.1.2 SIP Extension Group

Navigate through **Setup** > **Extensions** > **SIP Extension Group**

By Default there is an extension group called "Default SIP". You can limit or enable PBX features for extensions by creating Extensions group.

Figure 8: SIP Extension Group

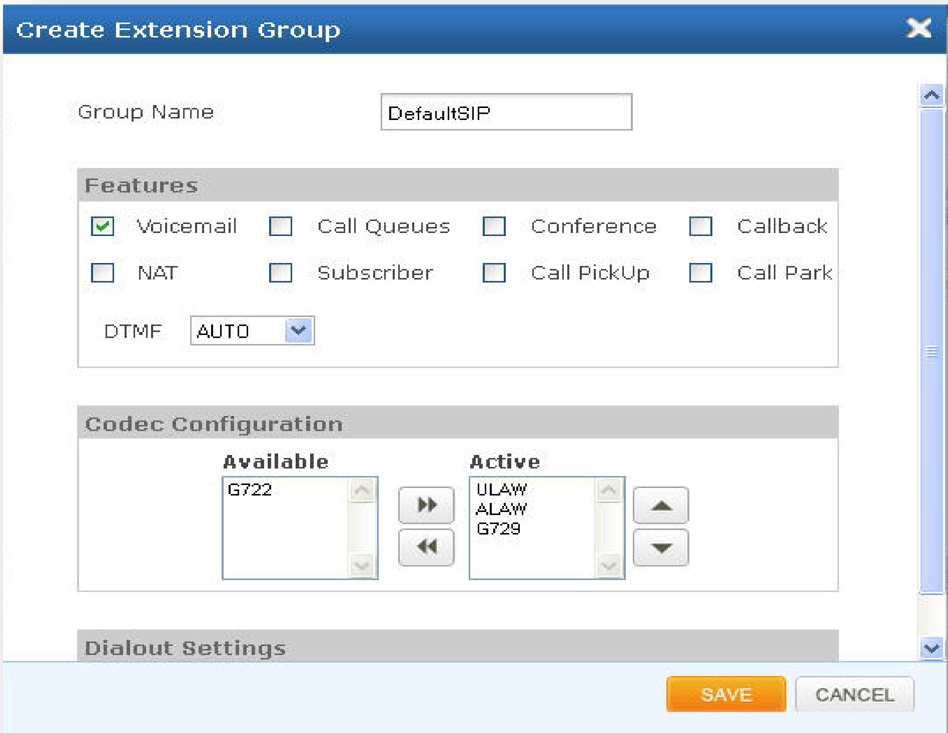


Figure 9: Create Extension Group

Group Name	Descriptive name for extension group
DTMF	Set default DTMF mode for sending DTMF digits. Options: • INBAND – sent along with audio (requires 64 kbit codec - alaw, ulaw) • INFO – sent as SIP INFO messages • RFC2833 – sent as RTP packets • AUTO – System automatic selects the mode. Uses RFC2833 if offered, inband otherwise. Default: **AUTO**
Features	Enable or Disable the desired features like voicemail, call queue, conference, and callback, call pickup, call park etc…
Codec Configuration	Choose the available Codecs and set priority in the order in which Mega PBX should prefer to send and receive audio. Supported codecs are alaw, ulaw, G.729, G.722.
Dialout Settings	Allow or deny the preferred dial out plans for this extension group.

4.2 Trunks

4.2.1 SIP Trunks

Navigate through **Setup** >**Trunks** > **SIP Trunks**

SIP Trunks provide the interface to any SIP companion such as VoIP service provider, any SIP server or SIP clients. Add different types of interfaces, and configure the signaling & media settings for each trunk.

Figure 10: SIP Trunks

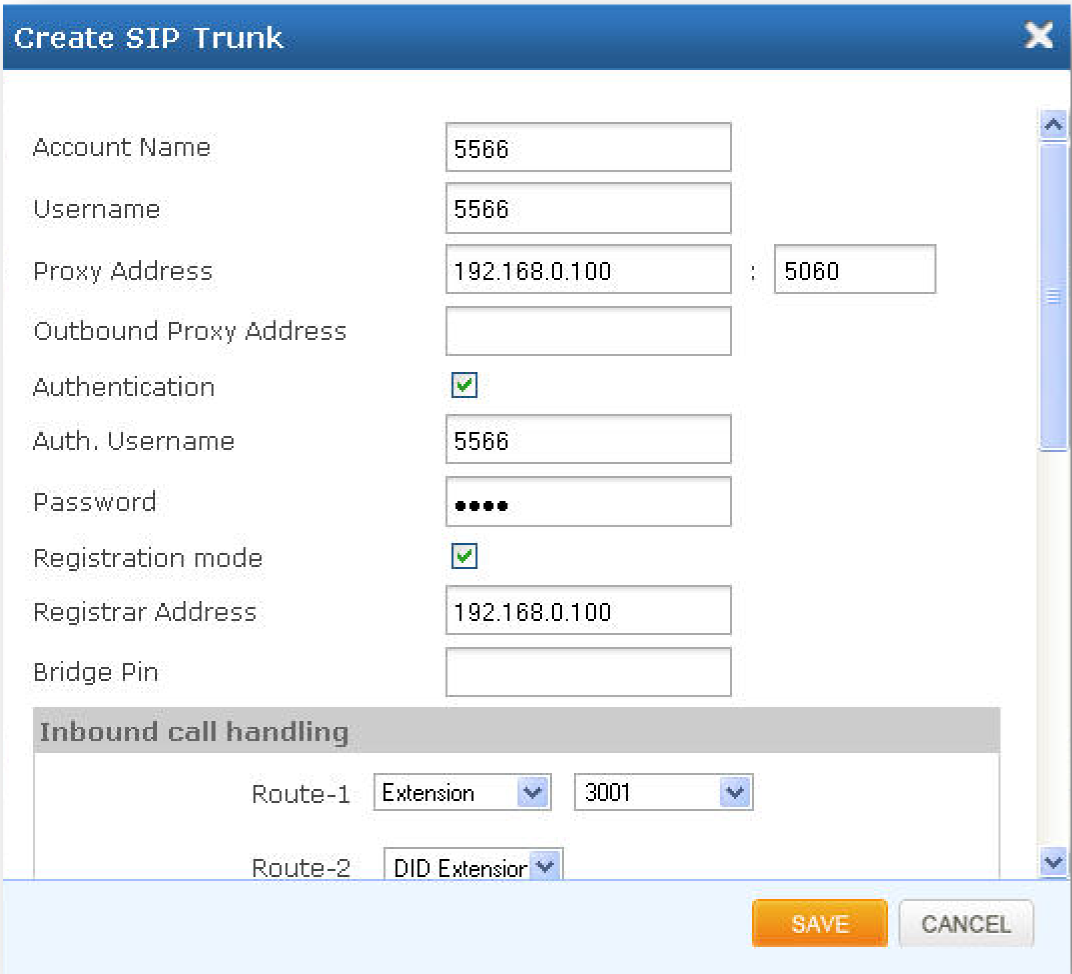


Figure 11: Create SIP Trunk

Account Name	Descriptive name for the SIP Trunk for user's reference. e.g.: 5656.
Username	Username of endpoint (e.g.: IPPBX) will use to authenticate with the Mega PBX, e.g.: 5656
Proxy Address	IP address or hostname with port of the endpoint (VOIP Service Provider or IPPBX) where the calls will be diverted. Default port no.: **5060,** e.g.: 192.168.0.100.
Outbound Proxy Address	IP address or hostname with port of the outbound proxy server. This ensures that all the SIP packets are sent via specified proxy. Specifying the port is not mandatory. Default port no.: **5060** e.g.: 192.168.0.222:5062 OR 192.168.0.222
Authentication	Enable, if Authentication is required by the End point (VOIP Service Provider or IPPBX)
Auth. Username	A username to use only for registration. e.g.: 5656.
Password	Password to authenticate registrations and inbound & outbound calls.
Registration Mode	Enable, if Registration to the End point (VOIP Service Provider or IPPBX) is required.
Registrar Address	IP address or hostname with port of the Registrar server where Mega PBX must register to. Specifying the port is not mandatory. Default port no.: **5060** e.g.: 192.168.0.222:5062 OR 192.168.0.222
Bridge PIN	You can set a PIN for outgoing calls on SIP trunk, thus you will set one more level of security. Leave it blank for unsecured mode. E.g.: 111.
Inbound Call Handling	Route incoming calls to any destination like extension, queue, and voice mail group, IVR, conference, DID Extensions etc. E.g.: Route To Extension 2000.

Show/ Hide Advance Options

Advance options, allows enabling the options like Features, Codec Configuration and Dial out Settings for a specific Trunk.

Advanced Options	
DTMF	Set default DTMF mode for sending DTMF digits. Options: • INBAND – sent along with audio (requires 64 kbit codec - alaw, ulaw) • INFO – sent as SIP INFO messages • RFC2833 – sent as RTP packets • AUTO – System automatic selects the mode. Uses RFC2833 if offered, inband otherwise. Default: **AUTO**
Registration Timeout	You can set the length of time in seconds between registration attempts (the default is 20 seconds).
Features	Enable or Disable the desired features like NAT, FAX, DID Routing, video calling and call recording.
Codec Configuration	Choose the available Codecs and set priority in the order in which gateway should prefer to send and receive audio. Supported codecs are alaw, ulaw, G.729, G.722

4.2.2 PRI Trunks

(Only for Mega PBX products with PRI support)

Navigate through **Setup** > **Trunks** > **PRI Trunks**

PRI Trunks provides the interface to any ISDN PRI companion such as PRI service provider or any other ISDN PBX. Create an interface for each span.

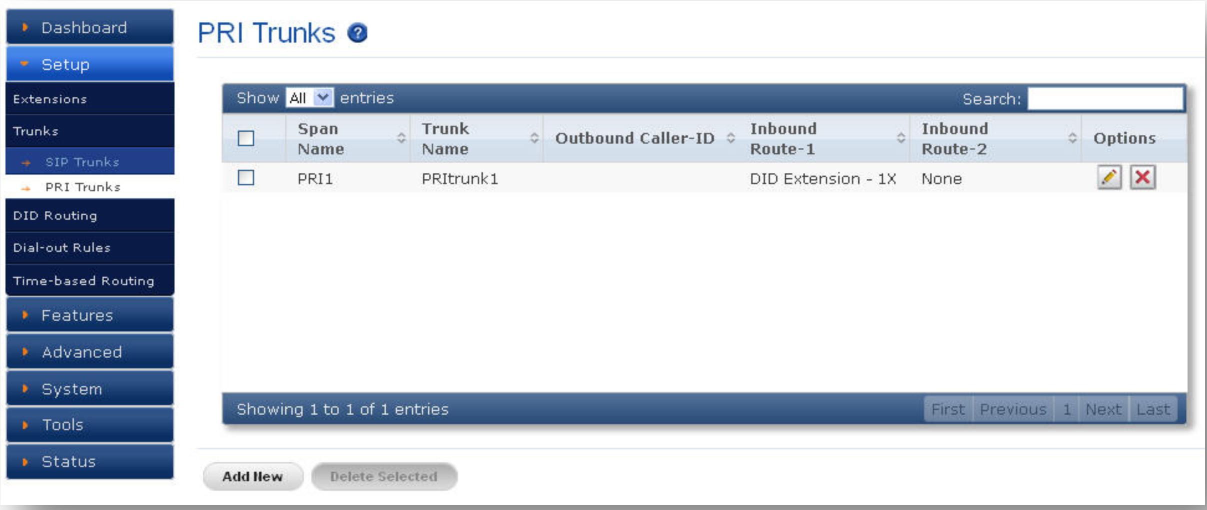


Figure 12: PRI Trunks

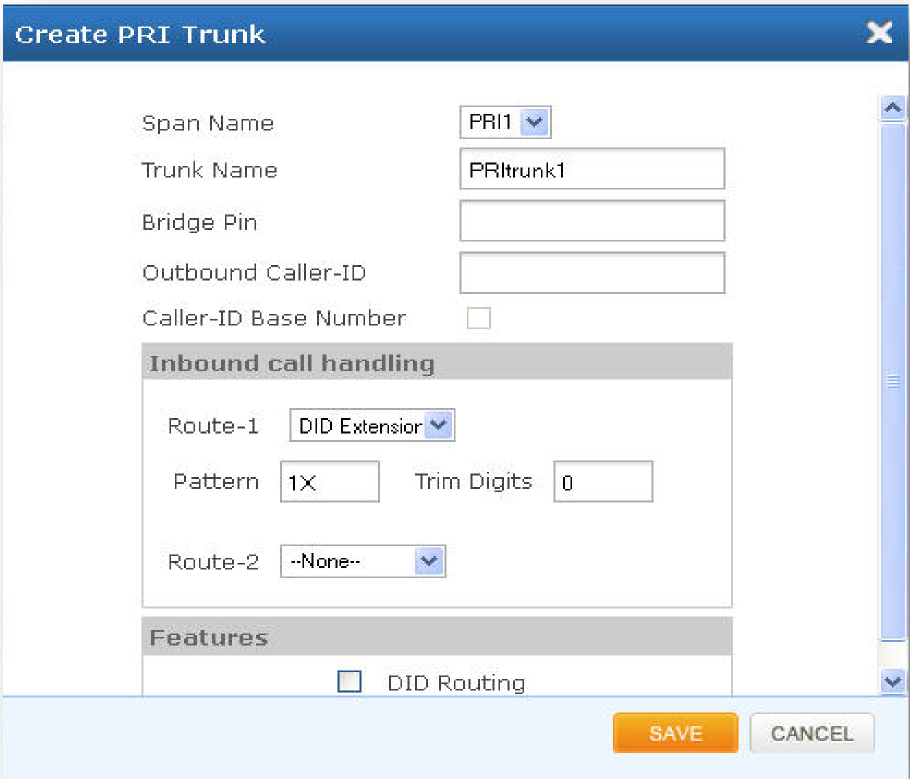


Figure 13: Create PRI Trunk

Span Name	Select the appropriate PRI Spans.
Trunk Name	Descriptive name for the PRI Trunk for user's reference.
Bridge PIN	You can set a PIN for outgoing call on PRI trunks, thus you can set

	one level of security. Leave it blank for unsecured mode.
Outbound Caller ID	Configure the Caller ID Number that would be applied for outbound calls over this trunk.
Inbound Call Handling	Route incoming calls to DID extensions or Dial out rules. Route-1 and route-2 can be defined. E.g.: Route-1: DID Extension pattern: 1X. Trim Digits : 0.
DID Routing	Enable or Disable the DID Routing for this trunk

4.3 DID Routing

Navigate through **Setup** >**DID Routing**

DID Routing is a feature that enables incoming calls to be routed directly to the selected extension or trunk.

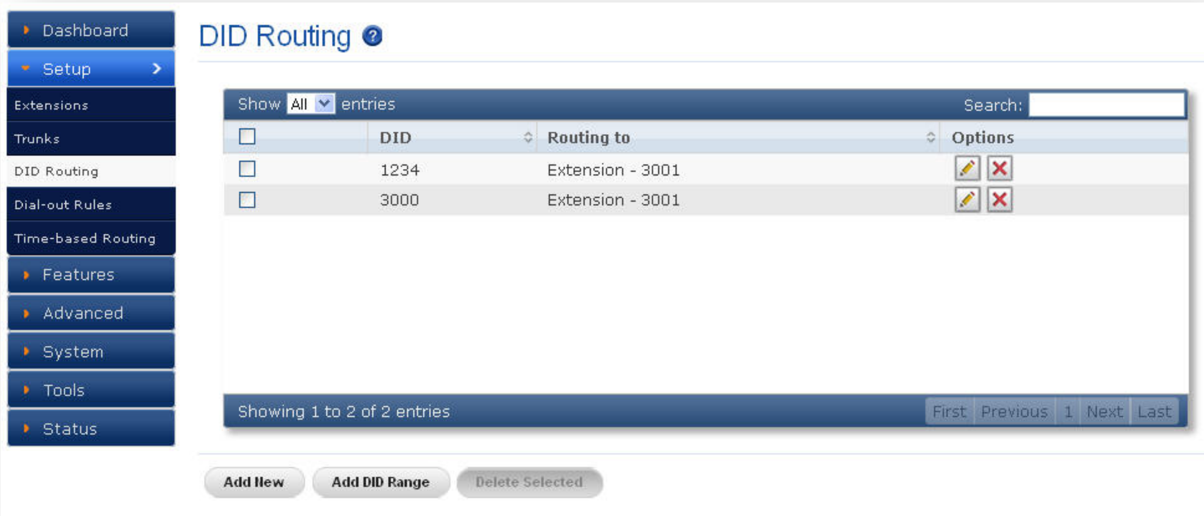


Figure 14: DID Routing

DID Number	It is the number provided by the service provider for the incoming calls. e.g.: 5023
Route To	Specify where the incoming calls should be routed. e.g.: Conference.
ADD DID Range	Here you can specify the range of the DID numbers provided by the service provider, e.g.: 5000 to 5010.

Start Extension to Map	Here you can specify the starting number of the SIP extension which is to be mapped with the DID numbers. e.g.:2000

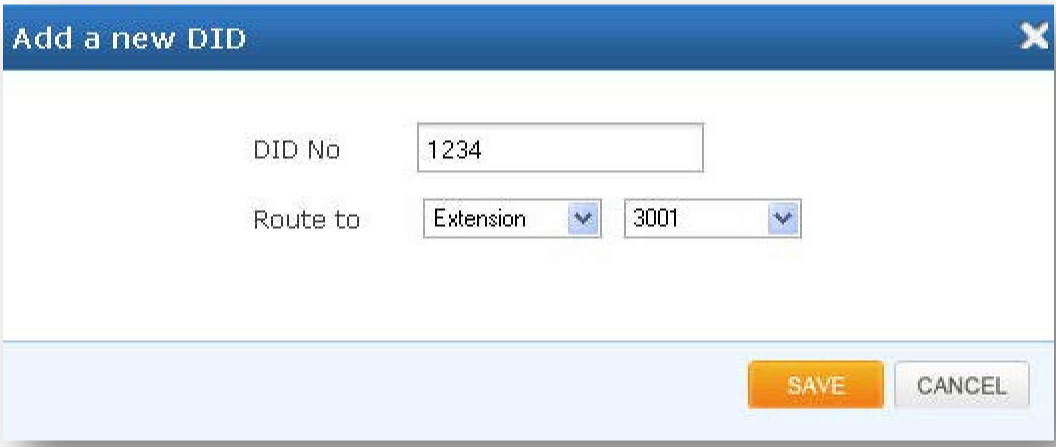


Figure 15: Add a new DID

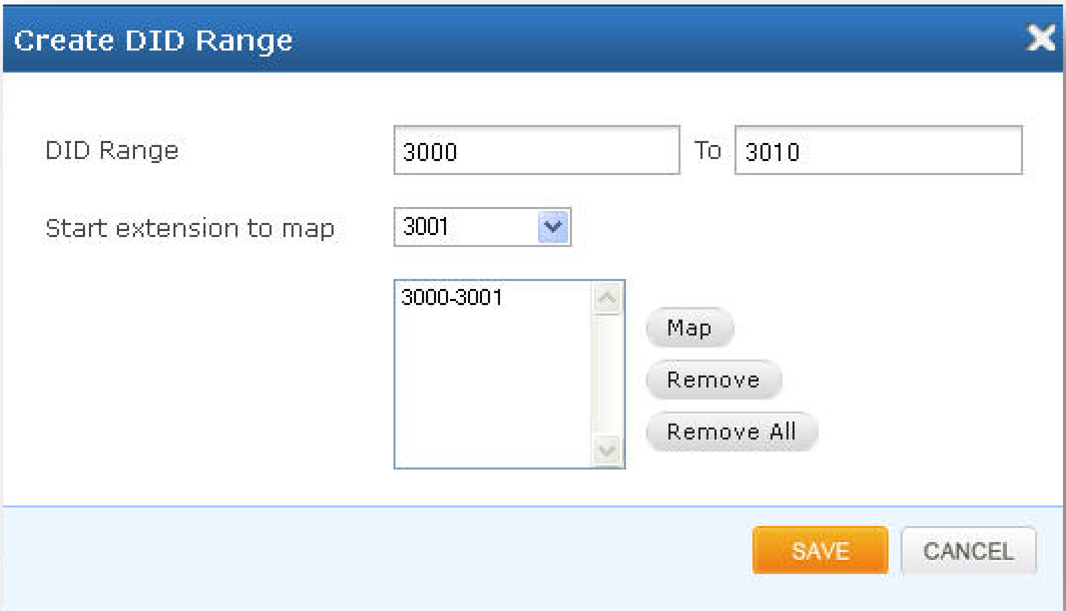


Figure 16: Create DID Range

4.4 Dial-out Rules

Navigate through **Setup** > **Dial-Out Rules**

Dial out Rules is used to configure the system to judge outgoing calls via trunks and also used to select a least cost routing provider based on the prefix configured.

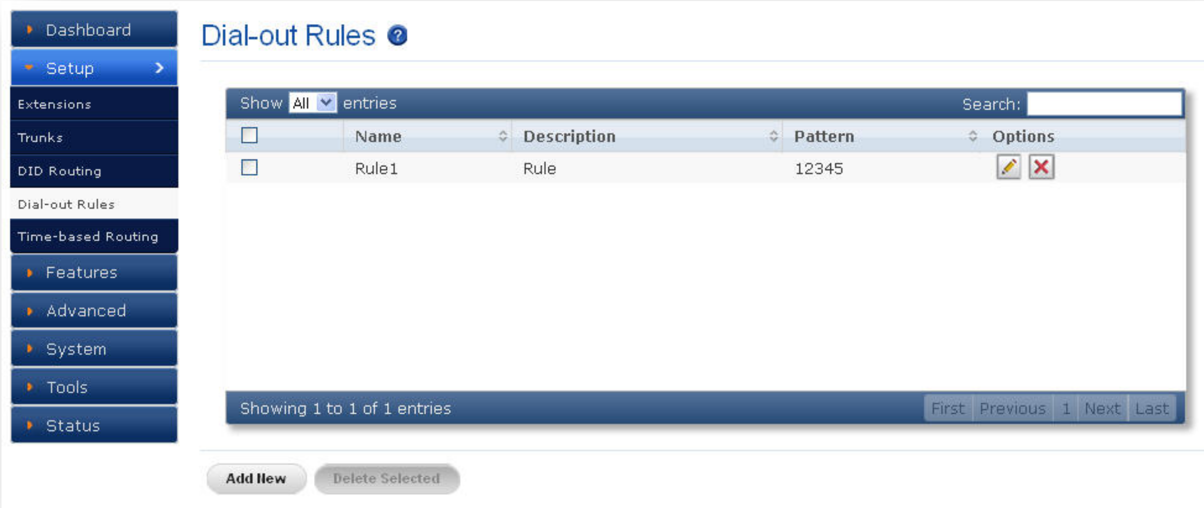


Figure 17: Dial-Out Rules

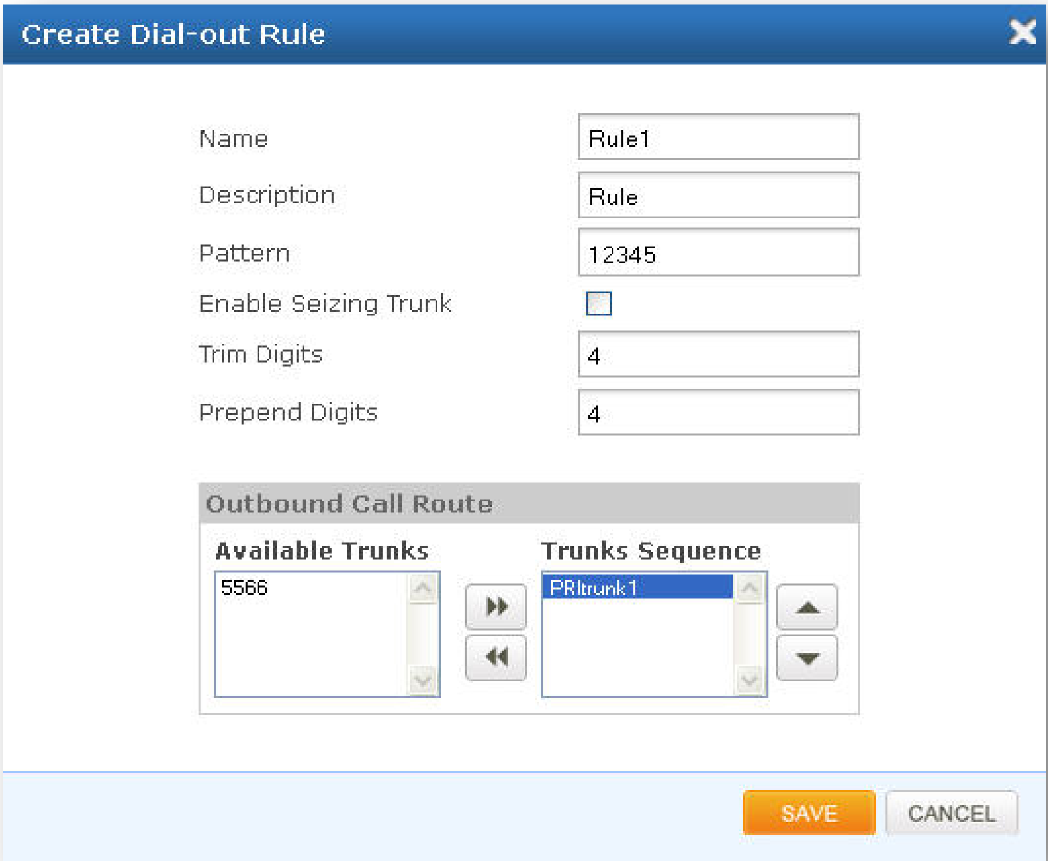


Figure 18: Create Dial-Out Rule

Name	Descriptive name for the Dial-Out rule for user's reference.
Description	Provide the description for the Dial-Out rule. (Optional)
Pattern	Specify the pattern to match the dialed string of the incoming call. Pattern: 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. e.g.: X. – match at least one digit 988XXXX – match 988 followed by 4 digits
Enable Ceasing Trunk	Seize the line and dial out.
Trim Digits	Number of digits to trim from the beginning of dialed number. Eg. Pattern=52xx , Trim Digits=1, dial out digits = 2xx
Prepend Digits	Entered digits add to the beginning of pattern to be Dialout.
Outbound Call Route	Select the preferred trunks where calls are to be routed for this Dial Out rule. Ordering of the trunks in the "Selected" column indicates the order in which call flows on failure.

4.5 Time -based Routing

Navigate through **Setup** > **Time-Based Routing Groups**

Time-base routing, routes calls to different locations based on the time of day and day of week, when a call is made.

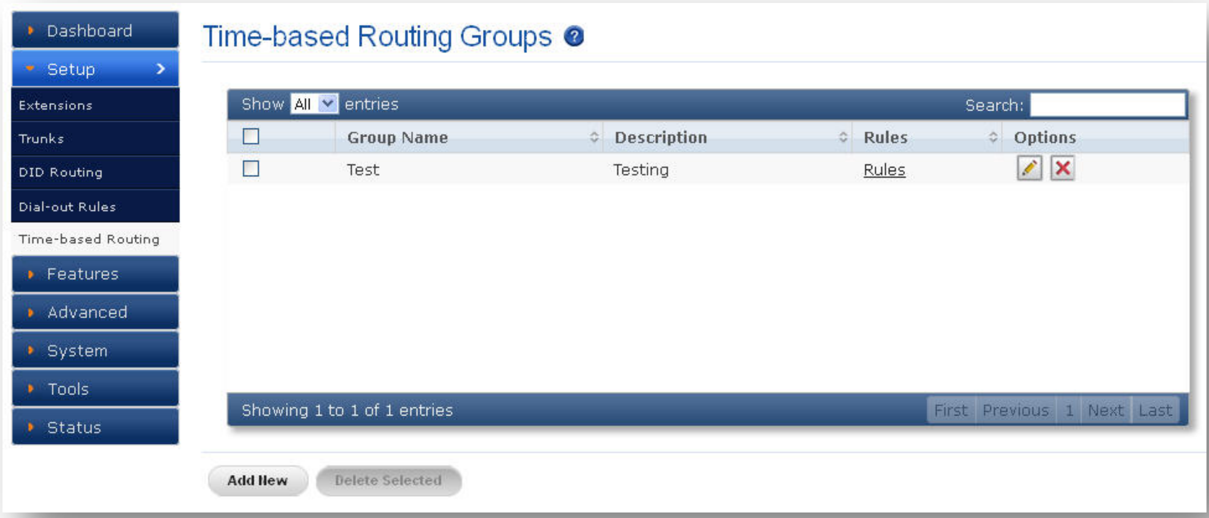


Figure 19: Time-based Routing Groups

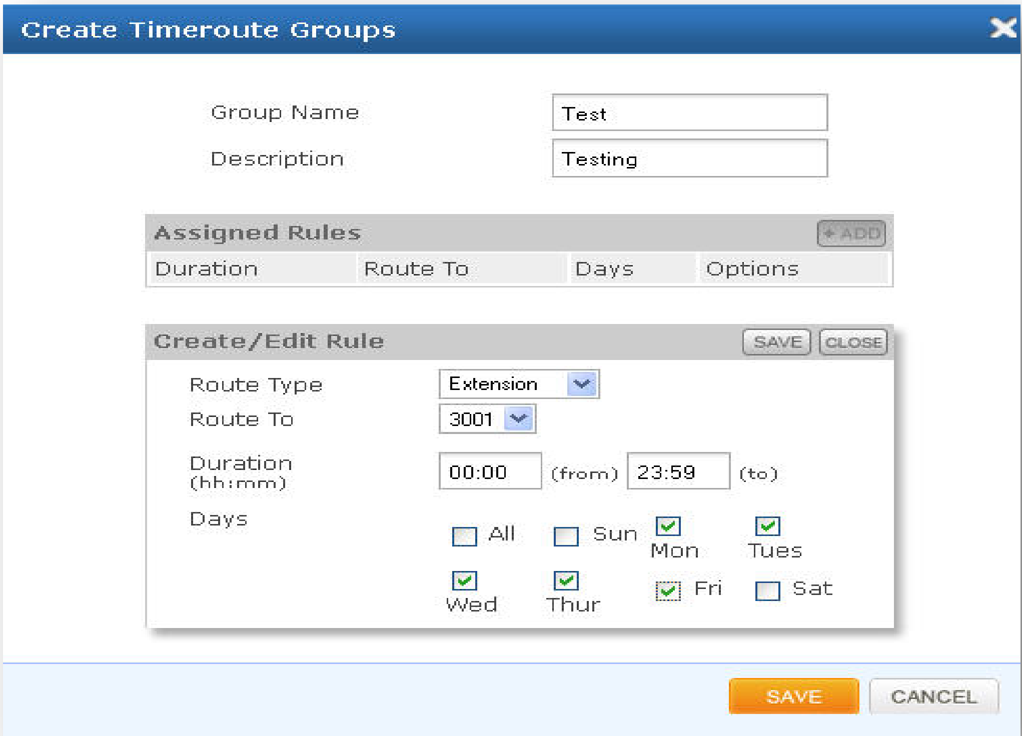


Figure 20: Create Time route Groups

Group Name	Descriptive name for the Time-Based Routing Group for user's reference.
Description	Provide the proper description for the Time based routing rule. (Optional)

Route Type	Select the destination where the call is routed to on matching the time. The destination can be any of these – Extension, Trunk, IVR, Queue, Voicemail Group, Fax to E-Mail
Route To	Setting the destination for the incoming route. The destination can be any of the created extensions, Trunks, IVRs, Queues and so on.
Duration	Specify the time range for which this routing rule will apply. Format: hh:mm
Days	Select the day/days during which this routing rule will apply.

*Make sure that the current date and time are configured correctly under **System**> **Date/Time Configuration.***

5. Features

5.1 IVR

Navigate through **Features** >**IVR**

Interactive voice response is a pre recorded interactive operator that allows an automatic separation of the incoming calls through the sequence of interaction with a multiple choices of menu with telephone callers.

Figure 21: IVR (Interactive Voice Response)

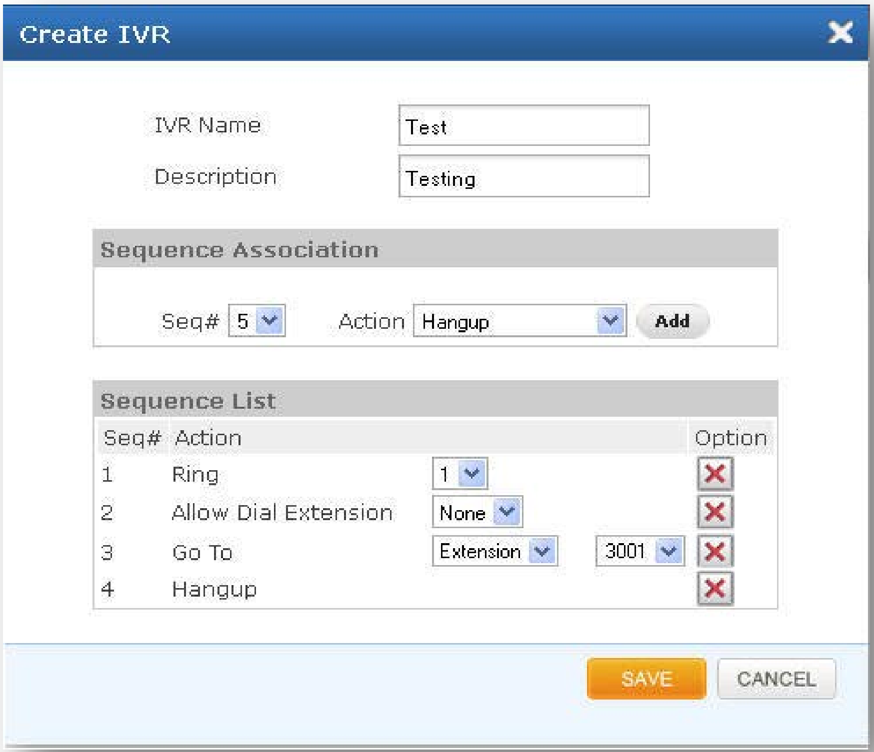


Figure 22: Create IVR

IVR Name	A character based name for the IVR.
Description	Detailed description of the IVR.
Sequence Association	Specifying how incoming calling should be handled sequentially.
Sequence List	List of operations to be handled.

5.2 Voice Files

Navigate through **Features** > **Voice Files**

Voice files are the audio files that are going to be played during the IVR play back and these files can be uploaded or they can be recorded, the user also has an option to associate key press with his voice file.

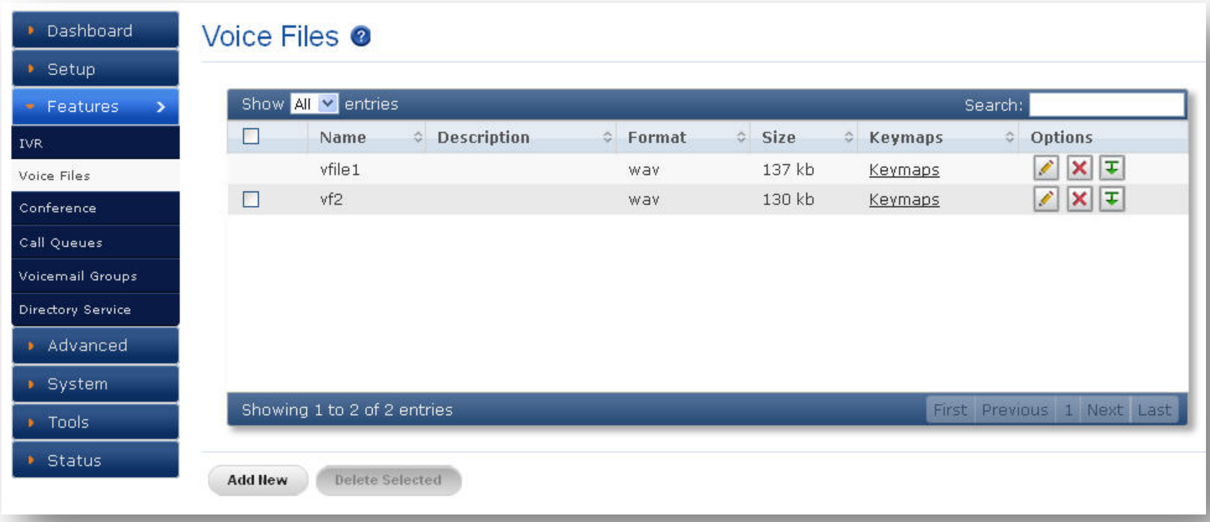


Figure 23: Voice Files

Example 1: Voice file without Key press

"Welcome to Allo.com! If you know your party's extension, please dial it now.

Example 2: Voice file with Key press

Press 1 for Customer Service

Press 2 for Sales

Press 0 for Reception

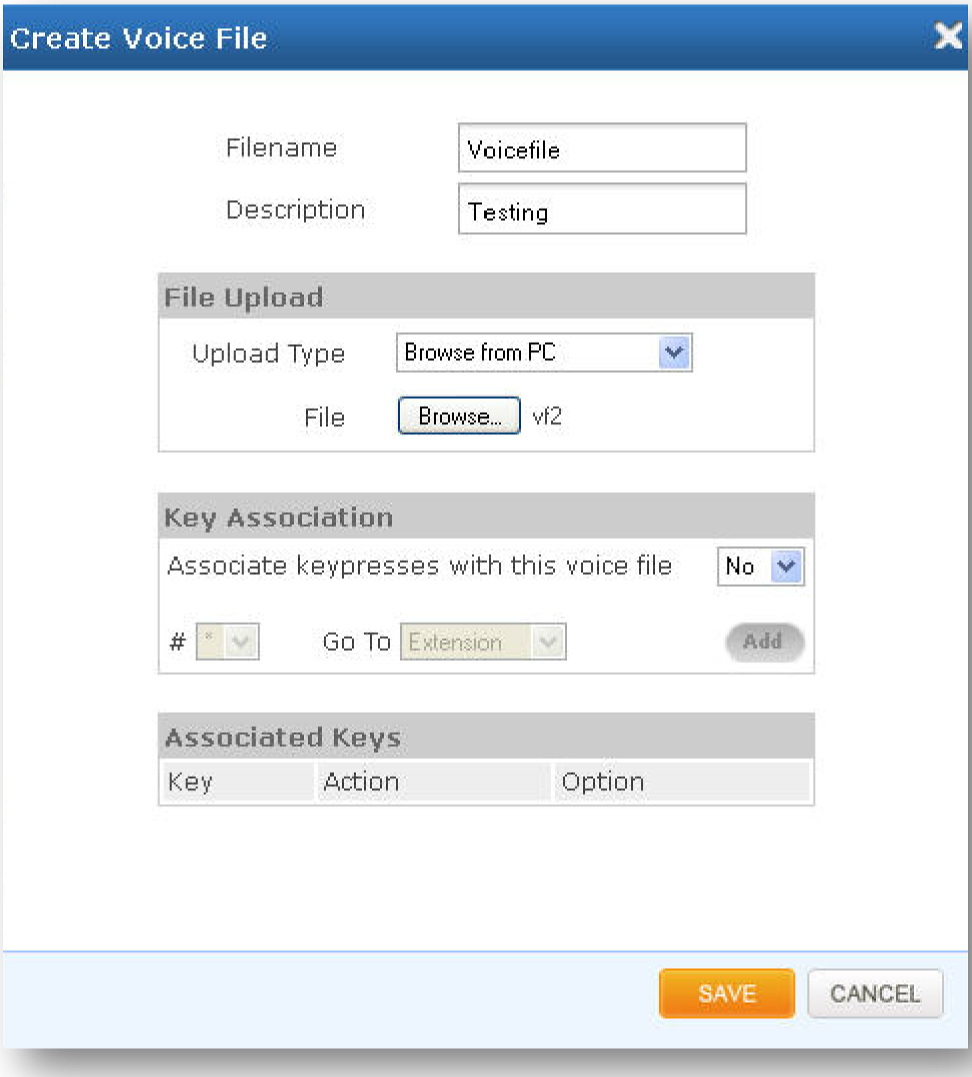


Figure 24: Create Voice file

File Name	A character based name for a voice file.
Description	Detailed description for the voice file.
File Upload	Upload the voice file to device.
Key Association	With associated key press option it will route incoming call to the specified destination.
Associated Keys	The sequence in which a key and its associated action is given.

*Mega PBX Supports”**.wav**” format only*

5.3 Conference

Navigate through **Features** > **Conference**

The Mega PBX supports password protected conference bridges that allow up to 50 simultaneous participants from any trunks or Internal Extensions.

Conference Number	Specifies a unique number which can be used to enter the conference room, e.g.: 6000.
Date	Specifies the date when the conference has to be scheduled, e.g.: 26/2/2014.
Start Time	Specifies the time to schedule the conference. E.g.: 12:30.
Duration	The duration during which the conference will be active, e.g.: 180 minutes.
Admin Pin	Specifies the PIN for the admin user to join the conference, e.g.: 555.
Speaker Pin	Specifies the PIN for the speaker and this user can speak and listen, e.g.: 666.
Listener Pin	Specifies the PIN for the listener and this user can only listen but cannot speak, e.g.: 777.
Participants	Specifies the privilege to be assigned and it can be Admin, Speaker or Listener.
Add Dynamic Participants	This option enables the user to add the participants dynamically from the web GUI.
Outbound Dialing	This option enables the user to add the participants dynamically over the trunks.
Maximum Participants	Specify the maximum participants to be allowed, e.g.: 10.

5.4 Call Queues

Navigate through Features > **Call Queues**

Call Queues are used to distribute calls in the order of arrival to the first available agent. The system answers each call immediately and if necessary holds it in a queue until it can be directed to the next available agent. This feature is used to balance the workload among the agents.

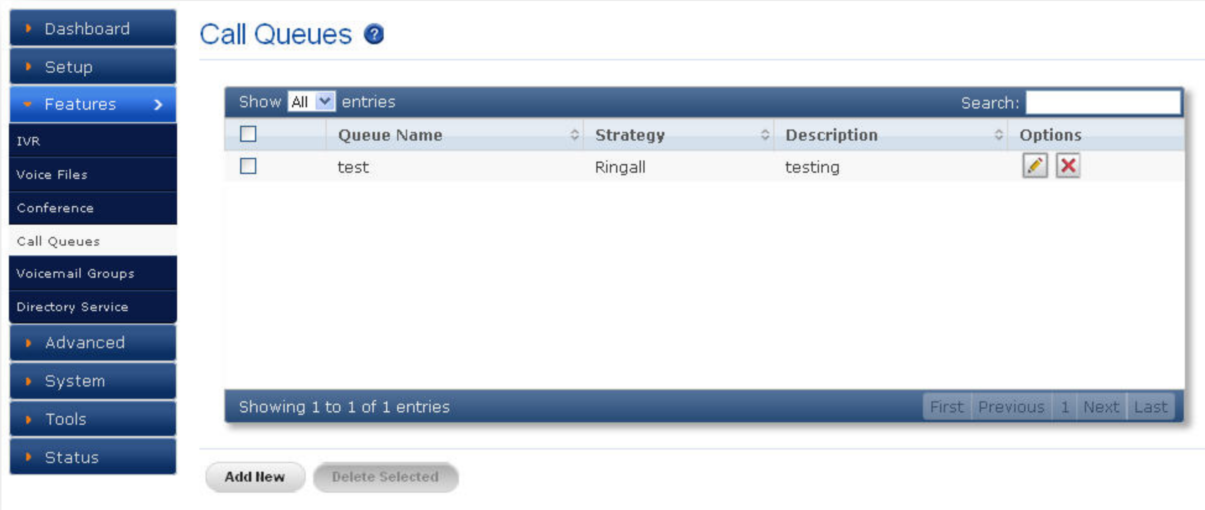


Figure 25: Call Queues

Figure 26: Create New Call Queue

Queue Name	A character based unique name for the queue.
Strategy	Calls are distributed among the members handling a queue

	with one of the several strategies like Ring all, Round robin, Random etc.,.
Description	Detailed description of the queue.
Queue Length	Used to decide the length of the queue.
Queue Members	Add the members to distribute the incoming calls.
Announcement Options	You can set timeout for queue, member and wrap up.

5.5 Voicemail Groups

Navigate through **Features** > **Voicemail Groups**

Voicemail Groups feature allows sending the voice message to multiple people or a group of people.

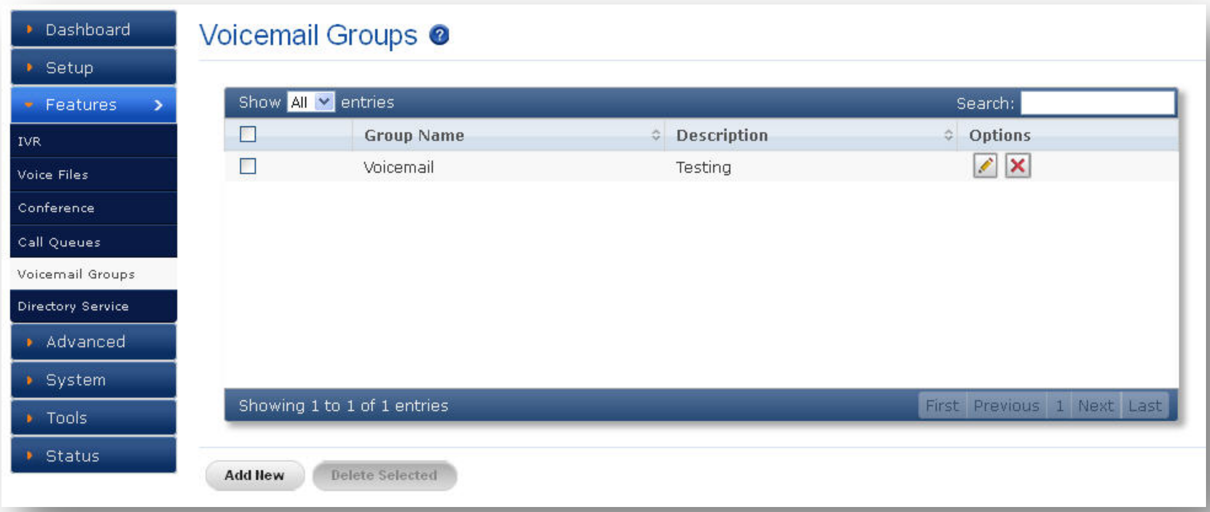


Figure 27: Voicemail Groups

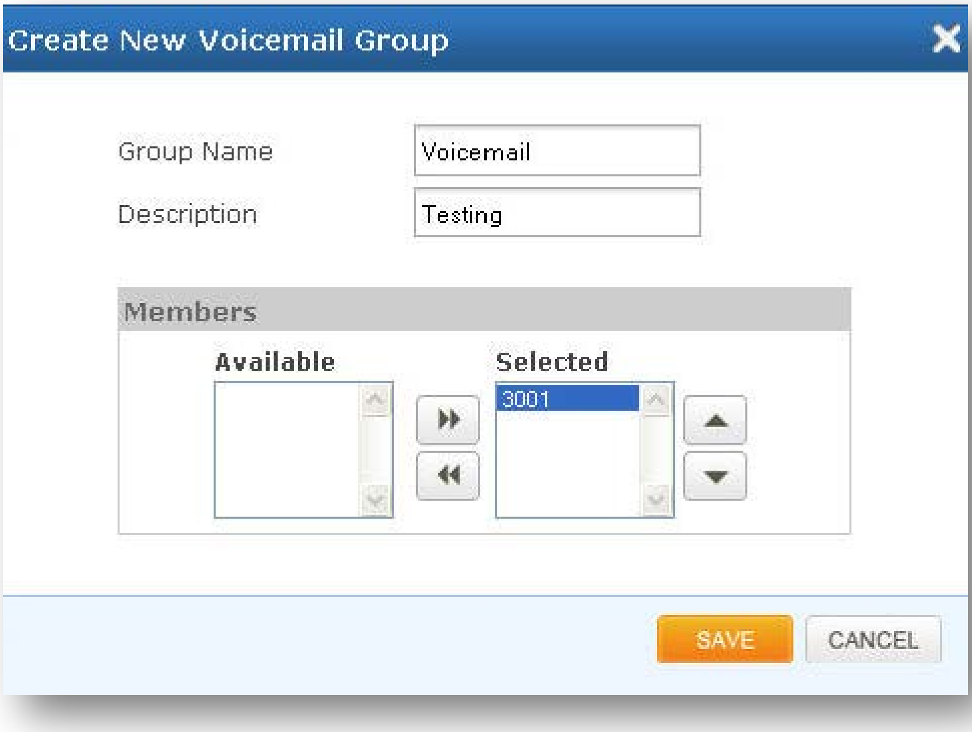


Figure 28: Create New Voicemail group

Group Name	A character based unique name for a voicemail Group
Description	Detailed description of the Voicemail Group.
Members	Specifies the members to receive the voicemail from the list of available members.

5.6 Directory Entries

Navigate through **Features** > **Directory Service**

Directory is where you can configure the Directory option for the extensions to search users by their First or last name. Dialing the 'Directory Extension' would present to the caller, a directory of users listed in the system telephone directory - from which they can search by First or Last Name.

Figure 29: Directory Entries

Extension Number	Specify the Extension number to dial for accessing the Name Directory, e.g.: 555.
First Name	Allow the caller to enter the first name of a user in the directory, e.g.: JOHN.
Last Name	Allow the caller to enter the Last name of a user in the directory, e.g.: RAJ.
Import Directory	Import the directory entries by browsing the corresponding file. File has to be in .CSV format.
Export Directory	Export the Directory Entries by specifying the filename. The File will be exported to the local computer in .CSV format

6. Advanced

6.1 Feature Settings

Navigate through **Advanced** > **Feature Settings**

Feature Settings allows to modify the basic call related functionalities like call pickup, Conference, Callback etc…

6.1.1 Extension

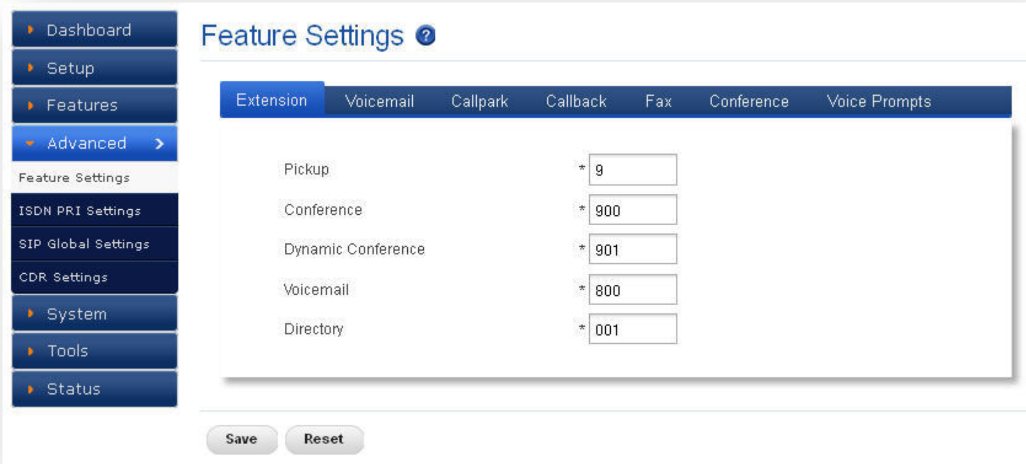


Figure 30: Feature Settings

Pickup	This feature code is to pick up the ringing extension from another extension if the party is not available in the desk. By Default Feature code is "*9". For e.g. if *9 is your call pick up code, by dialing *91002 from any other extension you can attend 1002 phone extension.
Conference	Using this feature code you can enter into the conference room. By Default the Feature code is "*900", i.e. Dial *900 from your phone and It will prompt you for the Conference Number and the PIN and it will let you enter into the conference room.
Dynamic conference	Using this feature code you can enter into the conference room and also add users in to it. By Default the Feature code is "*901", i.e. Dial *901 from your phone. It will prompt you

	for the Conference Number and the PIN and it will let you enter into the conference room. To add users into the conference *0 then followed by extensions.
Voicemail	This is to assign the code for accessing the voice mail. This will allow end users to change their personal settings for voice mail handling. By dialing to this number, any users who are registered to Mega PBX can access the Voice Mail. I.e. dial *800 from your phone and follow the instructions.
Directory	Using this Feature code you can dial the other extension with first and last name configured in the PBX.

6.1.2 Voicemail

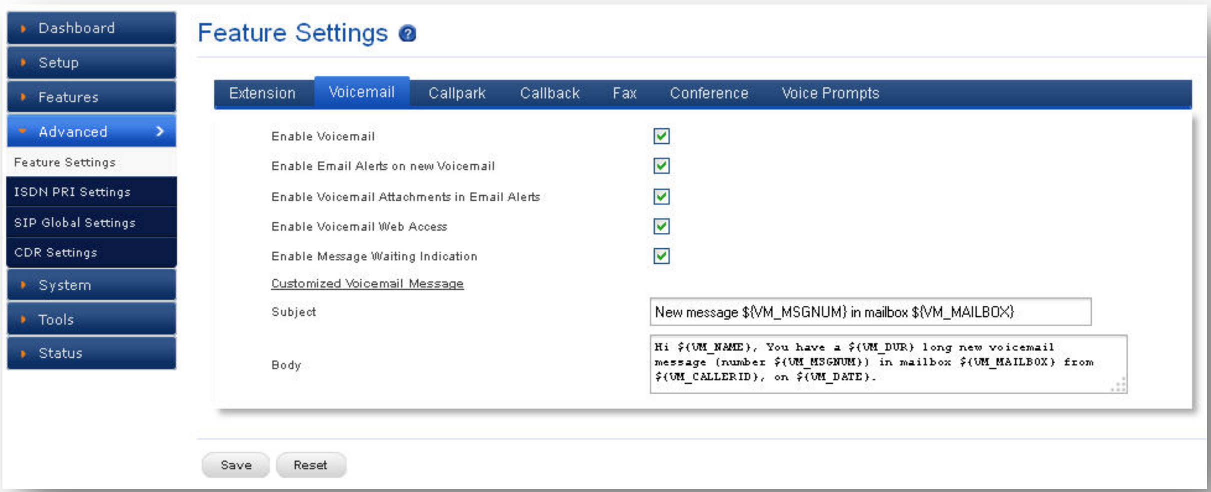


Figure 31: Voicemail

Enable Voice mail	With voice mail, callers can leave messages when you are busy, unable to answer phone calls, or when the IP phone is off-line.
Enable Email alerts on new Voicemail	Helps to send an email when someone leaves a voice message.
Enable Voicemail attachments in Email alerts	Helps to attach voicemails in your Emails.
Enable Voicemail access	Helps to access your voicemail by using the voicemail access

	number followed by the password.
Enable message waiting indication	Helps the subscribers to know that a voice message is waiting.
Customized Voicemail message	Allows you to customize the Voicemail message Notifications, which will be sent to the specified Email-ID's.

6.1.3 Call Park

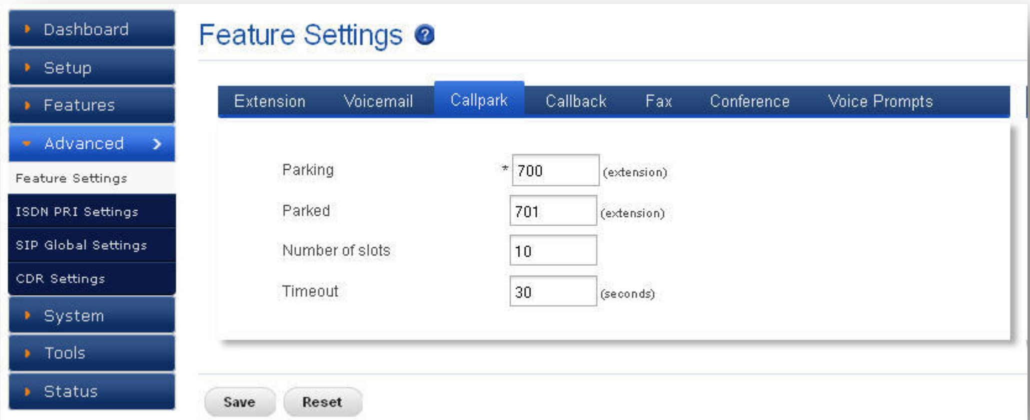


Figure 32: Call Park

Parking	Using this feature code you can park the number of calls. By Default the Feature code is "*700". i.e. park a caller by dialing *700 by using transfer button on your phone
Parked	Using this feature code you can retrieve the parked calls. By Default the Feature code is "*701. To retrieve the call, the user can go to any phone in the group and press the feature code "*701" and dial the parked extension.
Number of slots	Specifies the number of parking slots. For example, to configure six parking slots : (701, 702, 703, 704, 705, and 706)
Timeout	It is the timeout interval for calls parked at a call-park slot.

6.1.4 Call Back

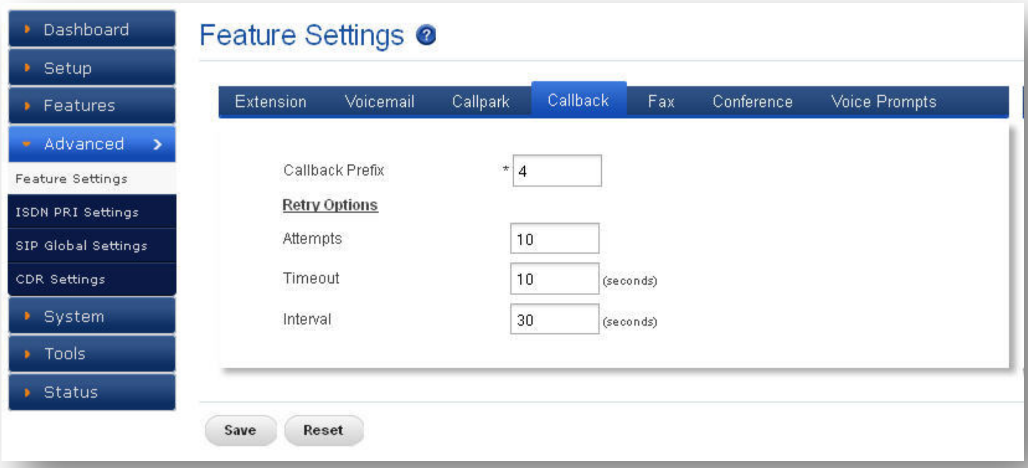


Figure 33: Callback

Callback prefix	To register callback function, user has to dial destination number with this prefix code.
Retry Options	This option helps to callback with number of attempts and timeout for the callback and also the time interval between callback calls.

6.1.5 FAX

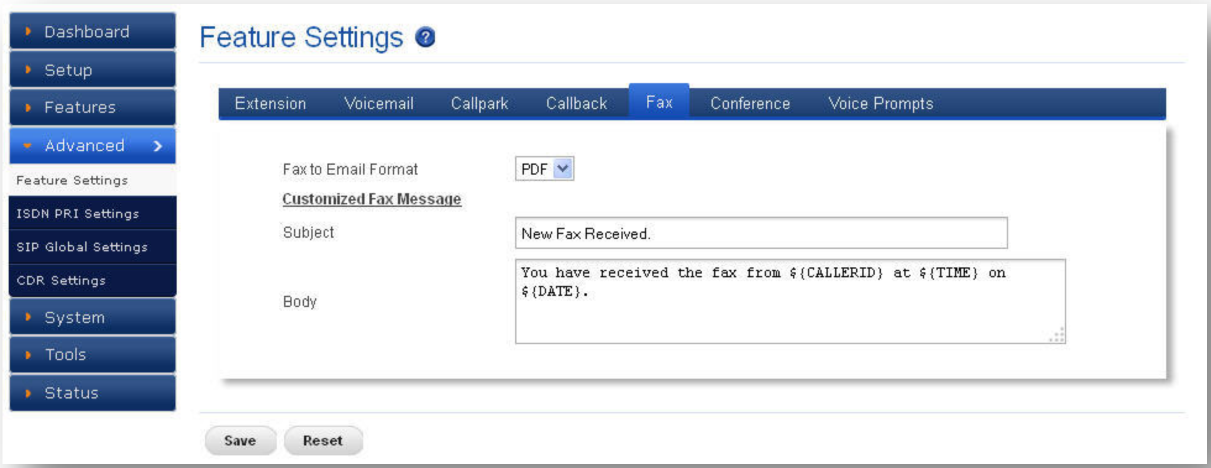


Figure 34: Fax

Fax to Email format	Specify the format to receive the FAX by email either PDF or TIFF format.
Customized Fax message	Allows you to customize the FAX message Notifications, which will be sent to the specified Email-ID's.

6.1.6 Conference

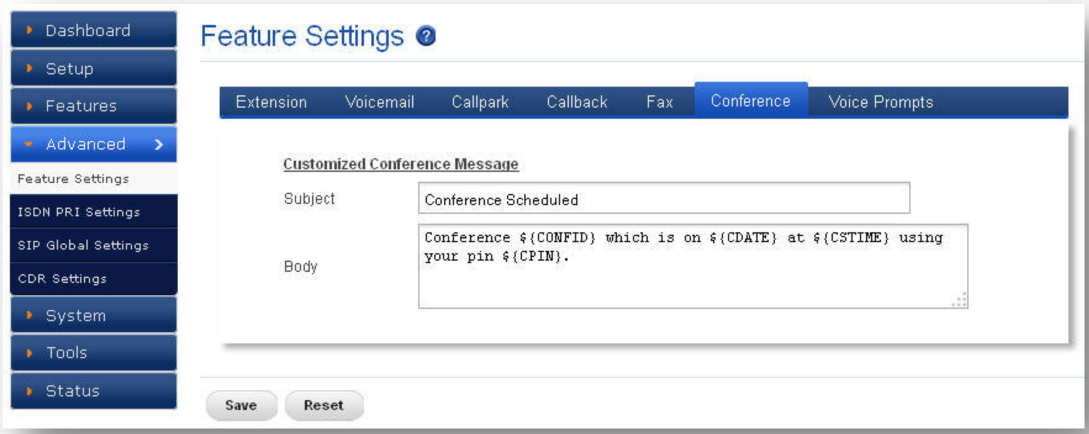


Figure 35: Conference

Customized conference message	Allows you to customize the Conference message Notifications, which will be sent to the specified Email-ID's.

6.1.7 Voice Prompts

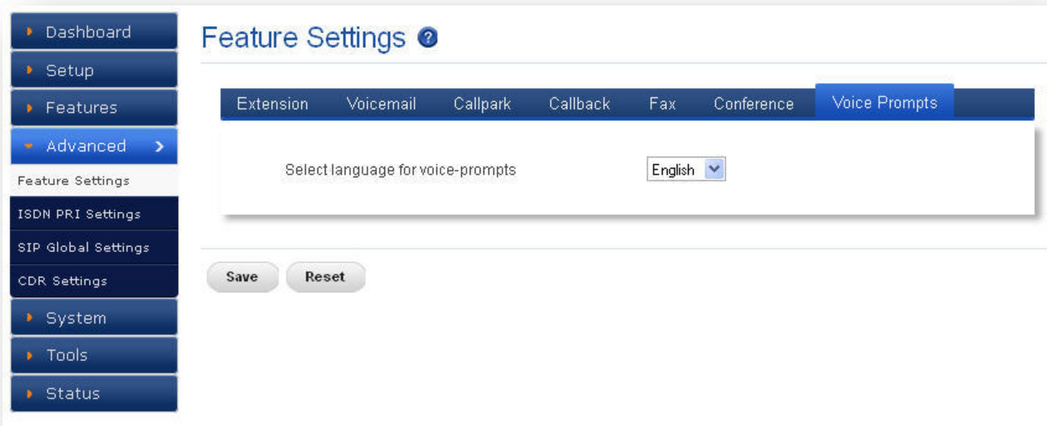


Figure 36: Voice Prompts

Select language for Voice-	Allows you to select the language for the Voice prompts.

prompts.	MegaPBX- BRI supports four language voice prompts such as English, French, Turkish, and Spanish.

6.2 ISDN PRI Settings

(Only for Mega PBX products with PRI support)

Navigate through **Advanced** > **ISDN PRI Settings**

This section provides the ability to modify the PRI settings depending on the carrier (T1/E1), signaling, switch type, etc with respect to the service provider or any other mate.

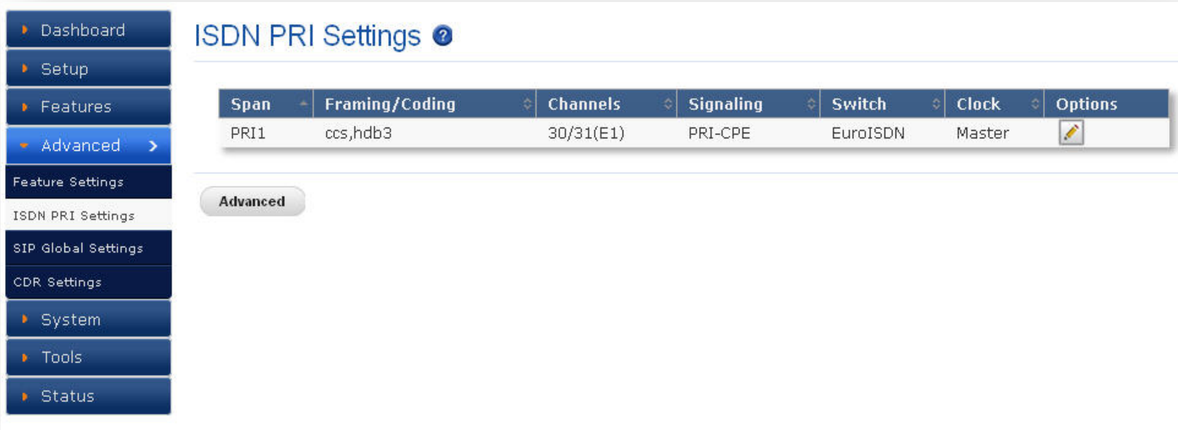


Figure 37: ISDN PRI Settings

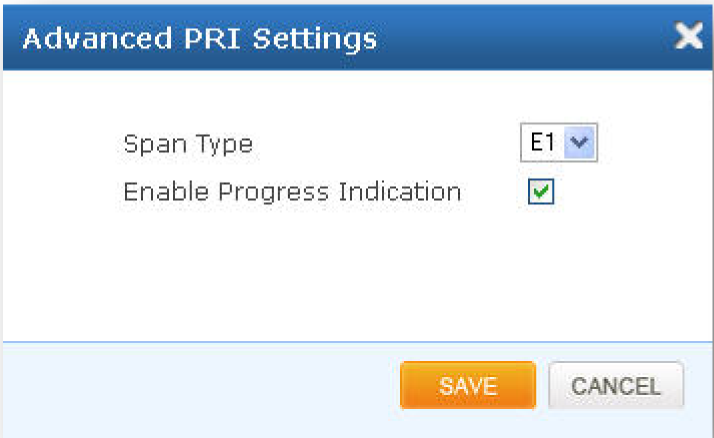


Figure 38: Advanced PRI Settings

Framing/Coding	Select Proper Framing & coding by checking with your service Provider. If CRC is enabled from Telco side, Please select CCS/HDB3/CRC4. If not select CCS/HDB3.

	Default for E1: **CCS/HDB3** Default for T1: **ESF/B8ZF**
Signaling	Select if gateway should work as customer premises equipment (CPE) or network device (NET). If Gateway is connecting to PRI service provider, select PRI-CPE. If connected with other Digital PBX's configured as CPE, then select PRI-NET. Default: **PRI-CPE**
Switch Type	Select the switch type as indicated by the ISDN service provider. Default: **E1- Euro ISDN & T1- National ISDN2**
Sync/Clock Source	Specify a transmit clock source. Master clocking uses the device's own system clock. Primary or secondary clocking uses a signal received from the T1 or E1 interface. Default: **Master**
Line Build Out	LBO depends on the line length for which attenuation is defined. Default: **0db**. Check with your service provider for appropriate Line build out settings if you face any issue.
Channels	Indicate the number of channels to be used on this span.
Advanced Options	
Dialed Numbering Plan	ISDN-level Type Of Number (TON) or numbering plan, used for the dialed number, which is dependant on geographical location. Default: **unknown**
Dialing Numbering Plan	Sets the calling number's numbering plan. Default: **National**
Prefix	Prefixes specified under international, national, private, local, and unknown, will be used with dynamic dialing numbering plan, which dynamically sets the Type Of Number in the ISDN messages. If the called number matches the national prefix specified, it will automatically set the ISDN TON of calling number to national. Similarly for international & local prefixes. This will also strip off the digits in the prefix from the called number as well, but only if national or international prefix is matched.

Overlap Dialing	Send overlap digits

Advanced PRI settings	
Span Type	Select the carrier type, E-carrier (E1) or T-carrier (T1) which depended on lines provided in your country. Default: **E1**
Enable Progress indication	Enabling this will provided call progress tones. Required, if service provided fails to provide call progress tones.

6.3 SIP Global Settings

Navigate through **Advanced** > **SIP Global Settings**

SIP Global settings apply to all VoIP traffic.

6.3.1 Port Settings

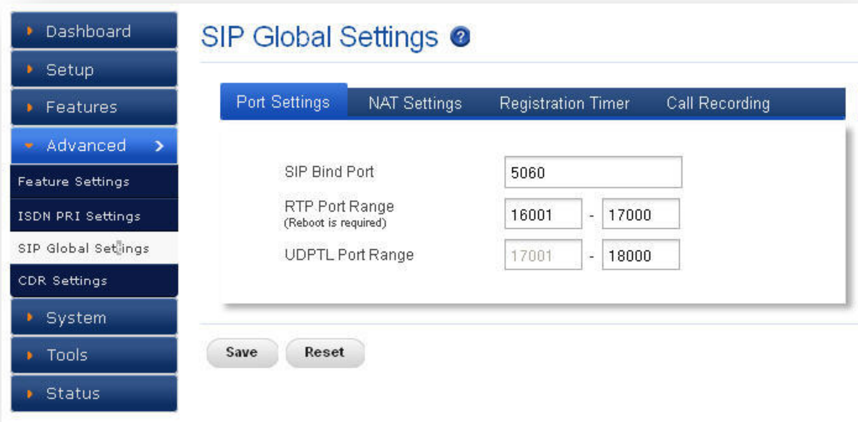


Figure 39: Port Settings

SIP Bind Port	Choose a port on which to listen for SIP UDP traffic. Default: 5060
RTP Port Range	Range of port numbers to be used for RTP traffic. Default: **16001- 17000** Make sure you configure this dynamic range of ports on your NAT Router. When the Mega PBX is behind a NAT and

	the NAT is configured to do port forwarding with above mentioned port range for UDP ports.
UDPTL Port Range	Port range for T38 Faxing is 17001 to 18000.

6.3.2 NAT Settings

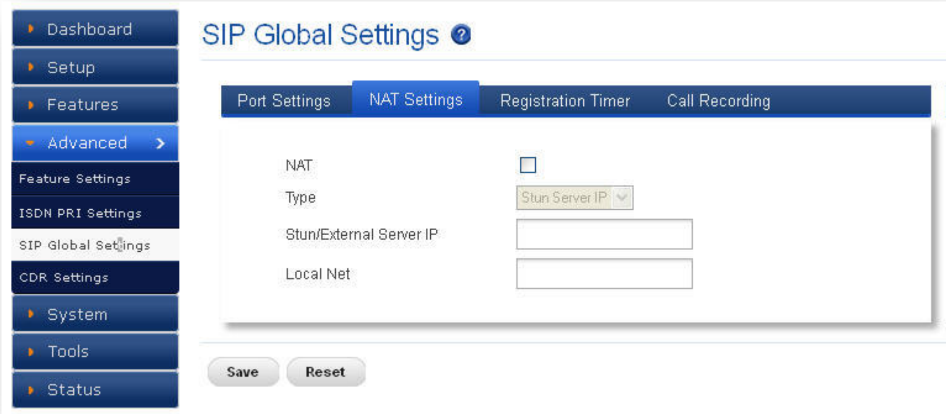


Figure 40: NAT Settings

NAT	NAT option is checked, when the Mega PBX is behind the Router/Firewall. Select either Stun Server IP or External IP. Default: **disabled**
Stun Server IP	If the Mega PBX is behind a non-symmetric NAT router, it may be necessary to use STUN to allow PBX to reliably communicate via IP through the router. Enter a STUN server IP address or domain name in the STUN Server field. For a list of public STUN servers, please Refer to: http://www.voip-info.org/wiki/view/STUN
External IP	Enter the NAT Traversal IP address i.e. Public IP Address of your internet, to communicate with Public Network when PBX is behind the NAT. This IP address will substitute in all outgoing SIP messages instead of Local IP address.
Local Netmask	Entering the Net mask of the local network of the Mega PBX allows it to identify the hosts falling within the same

	network. E.g.: 192.168.2.0/255.255.255.0

6.3.3 Registration Timer

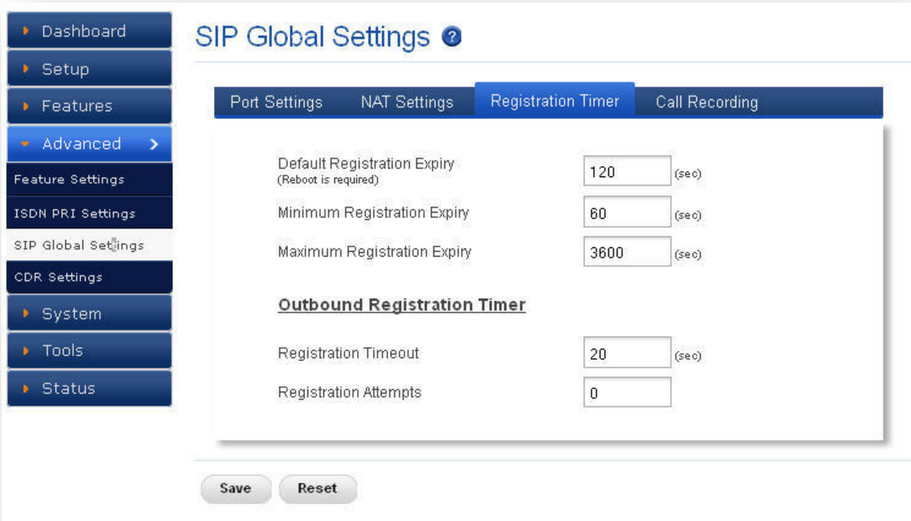


Figure 41: Registration Timer

Default Registration Expiry	Default duration (in seconds) of incoming/outgoing registrations. Default: **120 sec**
Min Registration Expiry	Minimum duration (in seconds) of registrations allowed by the Mega PBX. Default: **60 sec**
Maximum Registration Expiry	Maximum duration (in seconds) of incoming registrations allowed by the Mega PBX. Default: **3600 sec**
Registration Attempts	Number of registration attempts before giving up with registrar (Outbound Registrations only). Default: **0** (never give up)
Registration Timeout	Registration attempt will be retried till this duration (in seconds), if no response from the Registrar. (Outbound Registrations only). Default: 20 sec

6.3.4 Call Recording

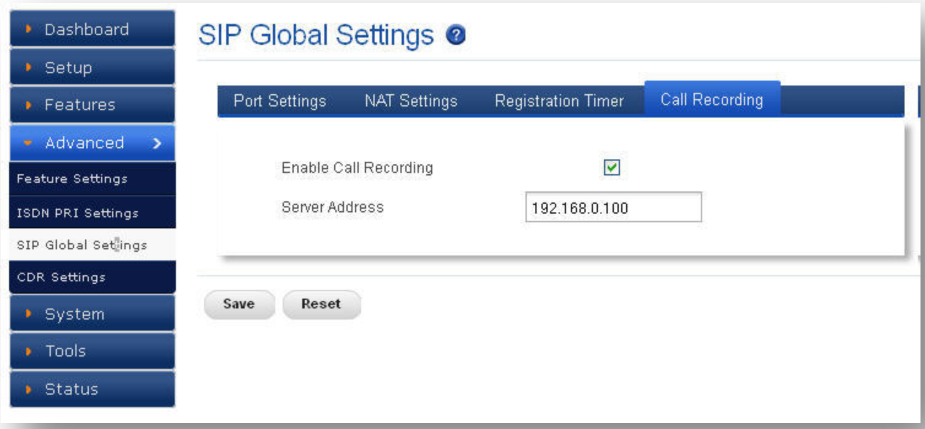


Figure 42: Call Recording

Server Address	It enables Call Recording with the given server where the ORK audio application is running.

6.4 CDR Settings

Navigate through **Advanced** > **CDR Settings**

CDR Settings feature allows managing the call records via radius server and FTP server.

6.4.1 Radius Configuration

Radius Server Configuration allows configuring the radius server information where all the call records will be saved in the form of vendor-specific attributes (VSAs).

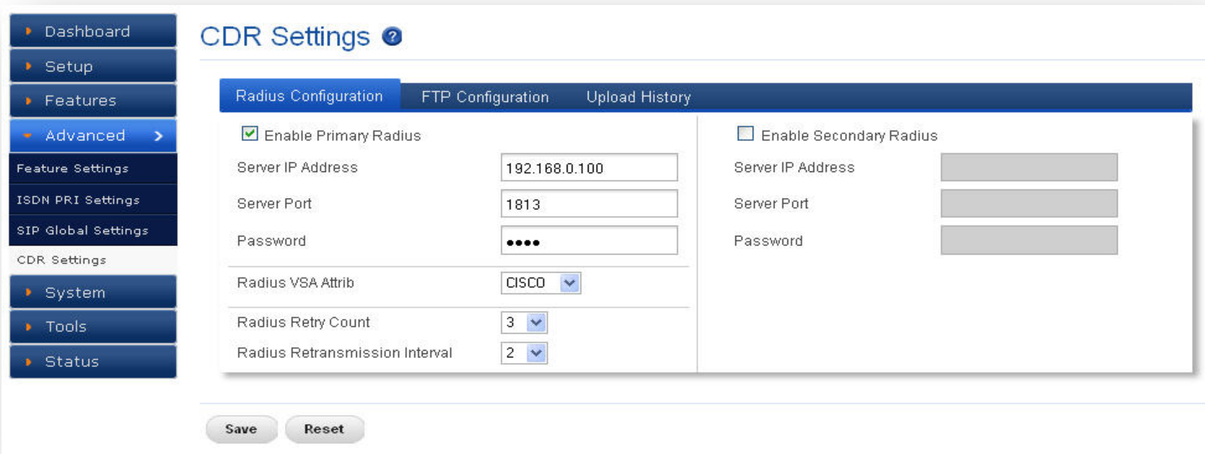


Figure 43: Radius Configuration

To upload CDR entries via Radius, we need to install Radius Server on PC and the same has to be configuring it.

Server IP Address	User can enter the IP Address of Radius Server. Ex: 192.168.xxx.xxx
Server Port	It specifies the Port Number on which Radius Server can be connected. By default Radius Server works on 1813 Port. It can be changed depending upon Radius Server's Configuration.
Password	User can enter the Radius Server's password to authenticate the Radius configuration.
Radius VSA Attrib	MegaPBX supports two attributes such as CISCO, DIGIUM. User can select anyone from the drop down list.
Radius Retry Count	It specifies No. of retries to upload CDR entries to Radius Server.
Radius Retransmission Interval	It specifies the time duration between retries.

Note: User can use secondary Radius Server.

6.4.2 FTP Configuration

This will allow you to configure the FTP server information for CDR billing and CDR upload scheduling by setting the Frequency of schedule, Day and Time.

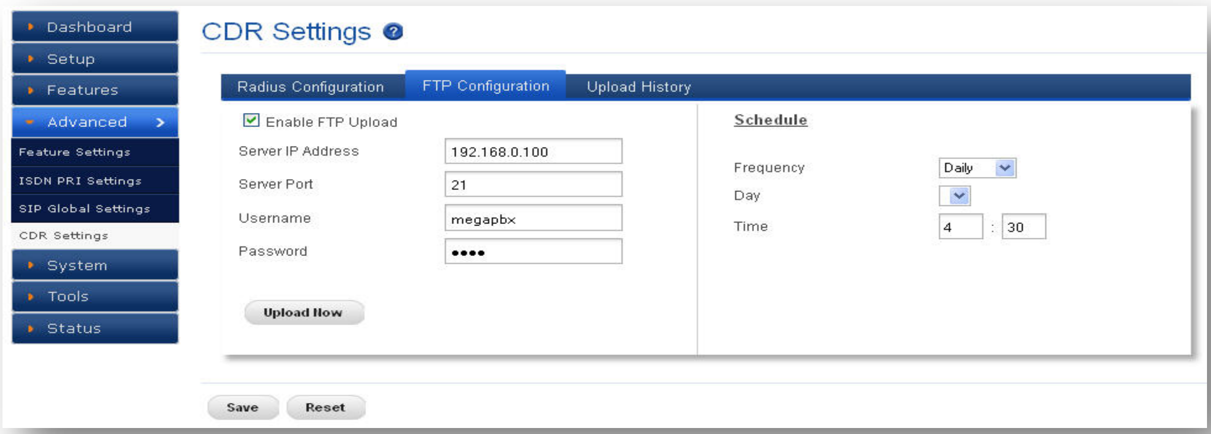


Figure 44: FTP Configuration

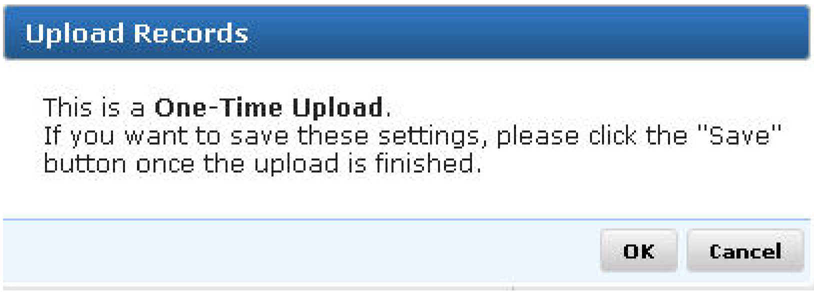


Figure 45: Upload Records

Server IP address	User can enter the IP Address of Radius Server. Ex: 192.168.xxx.xxx
Server Port	It specifies the Port Number on which Radius Server can be connected. By default Radius Server works on 1813 Port. It can be changed depending upon Radius Server's Configuration.
Username	It specifies the username of endpoint (e.g.: IPPBX) will use to authenticate with the Mega PBX, e.g.: 5656
Password	User can enter the Radius Server's password to authenticate the FTP Configuration Settings.
Schedule	A user can schedule uploading CDR entries to FTP server. There are three frequency options: Daily, Weekly and Monthly.

6.4.3 Upload History

It shows the upload history of CDR with the details like time, server address and status of the uploaded CDR.

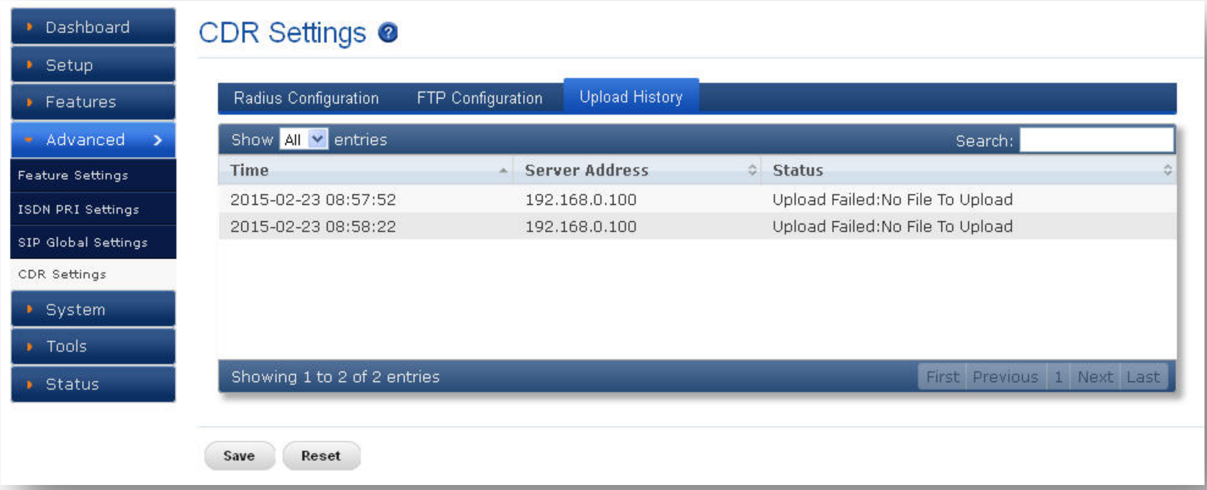


Figure 46: Upload History

7. System

7.1 Network

Navigate through **System** > **Network Settings**

Network Settings allows modifying the device network settings.

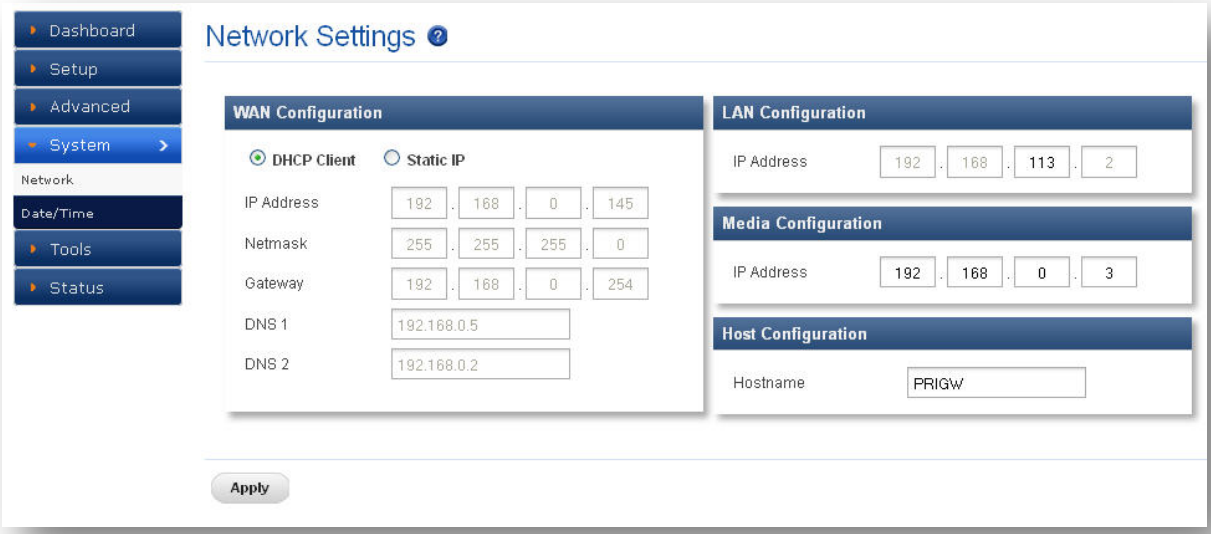


Figure 47: Network Settings

WAN Configuration:

DHCP	When enabled and a DHCP server is available, the Mega PBX will auto configure itself. If DHCP server is not available, select “Static”, and fill in the Network Configuration.
IP Address	The static IP address corresponding to your WAN configuration.
Net mask	The Net mask corresponding to your WAN configuration.
DNS	The IP address corresponding to a DNS server.

LAN Configuration:

LAN Port is a management port. Mega PBX can be connected back-to-back to a PC or to a LAN network for configuration. It is always recommended to connect back-to-back to a PC. In case, connected to LAN network & if IP series clash is found, IP series can be changed here.

WAN port IP address and LAN port IP address should not be in the same network segment.

Media Configuration:

It is applicable only for 4. 0.0 Version onwards. If the media IP address is not configured, there won't be voice path in the calls.

Media IP Address needs to be configured manually and it should be in the same network segment of WAN IP address. This IP address is used for Voice Media Configuration.

For E.g.: If the WAN IP address is 192.168.0.145 and media IP address should be in the same Network Segment (available IP address) side. 192.168.0.3

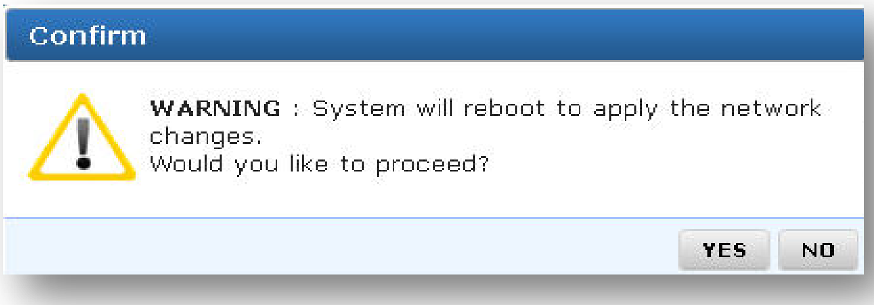


Figure 48: Warning

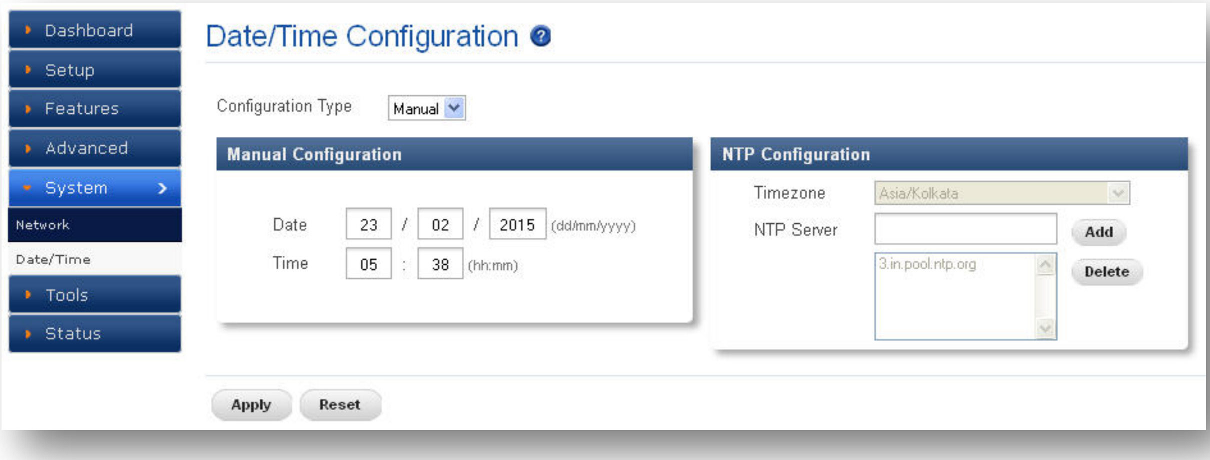
System will reboot to apply the network changes.

Host Configuration:

Host name: Label or IP of your system identity.

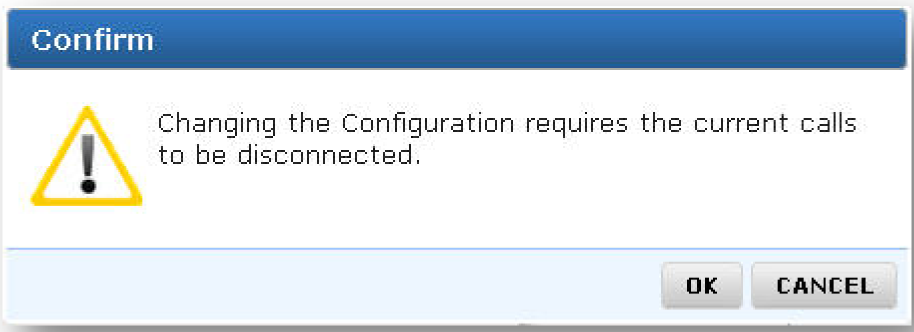
7.2 Date/Time

Navigate through **System** > **Date/Time**



Configuration Type	Date and Time of the Mega PBX can be either set manually (uses RTC) or automatically (through NTP). Default: **NTP**
NTP Configuration	**Time Zone:** Select the correct time zone for the location where the Mega PBX is installed using the Time Zone dropdown box. Default: **Asia/Kolkata** **NTP Server:** URI or IP address of the NTP (Network Time Protocol) server, which will be used to synchronize the date and time. E.g.: 3.in.pool.ntp.org

Click on **"APPLY "**button, followed by "**SAVE ALL**" button to update the configuration changes.



Changing the configuration requires the current calls to be disconnected.

8. Tools

8.1 Diagnostics

Navigate through **Tools** > **Diagnostics**

Analyze the functionality of the Mega PBX with some of these diagnostic tools provided.

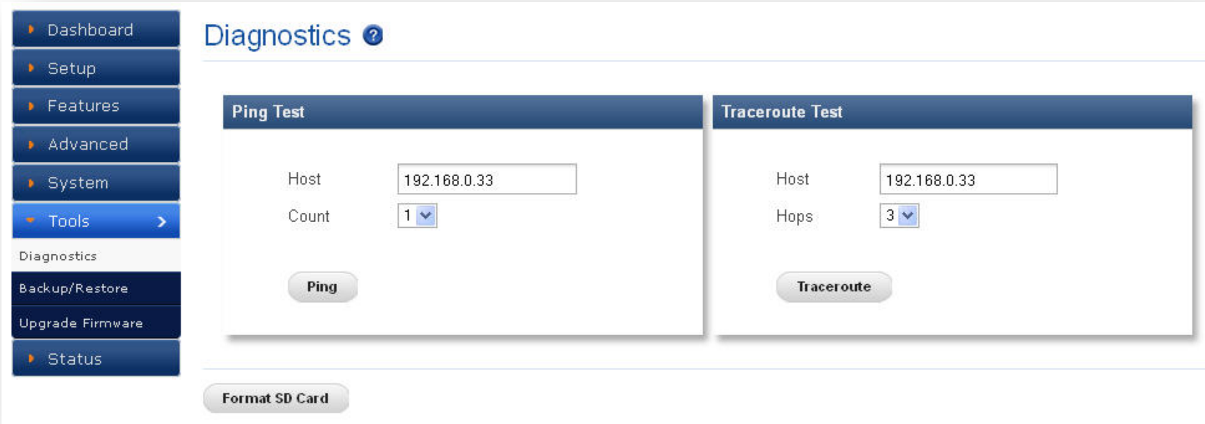


Figure 49: Diagnostics

Ping Test:

It is used to check the packet loss and latency time from your SIP end client like IP Phone/ FXS gateways to check the quality of your network connections.

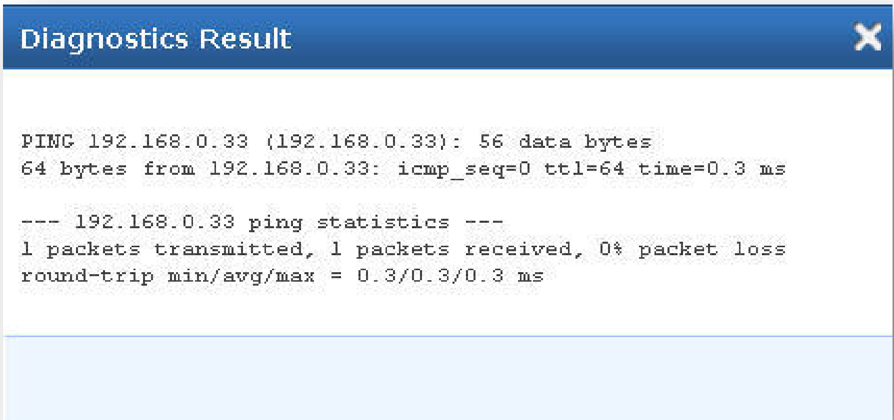


Figure 50: Diagnostics Result

Trace route Test: It is used to determine the route taken by packets across an IP network.

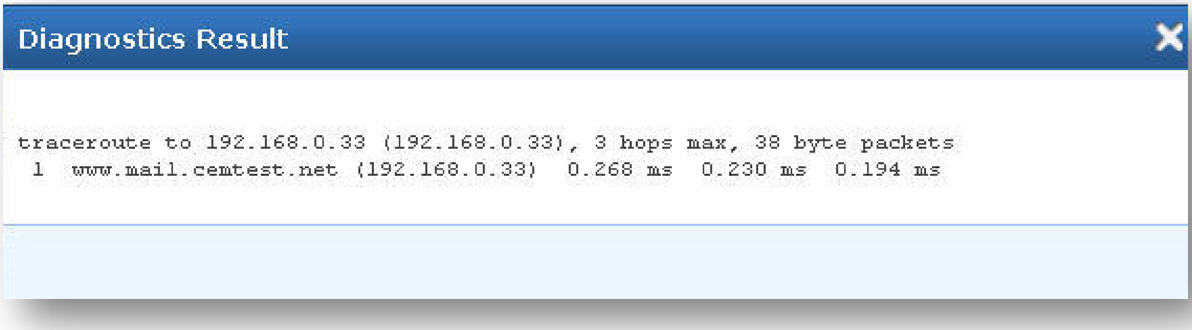


Figure 51: Diagnostics Result

Format SD Card / Format USB Drive: This will allow you to completely erase all the contents in the SD card/ USB Drive (MegaPBX-BRI supports USB Drive).

If you are formatting the USB Drive/SD card details, kindly take the back of the MegaPBX configuration.

8.2 Backup/Restore

Navigate through **Tools** > **Backup/Restore**

Back Up:

Allow you to take the back up of the System configurations & save it to the local PC.

Restore:

Restoring from a new upload or backup file will destroy all current configurations and require a system reboot. All calls will be dropped and all current configurations will be destroyed.

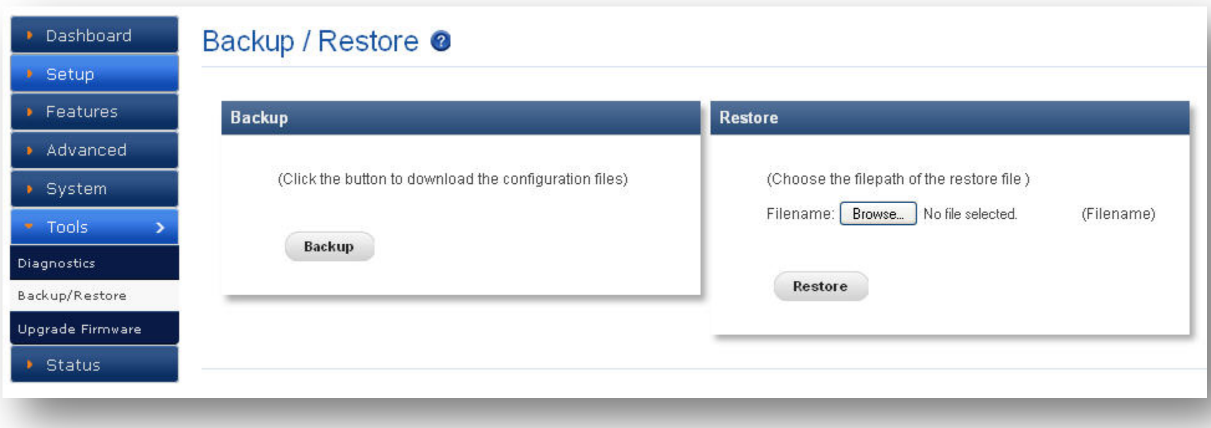


Figure 52: Backup/Restore

Administrator password will not be restored on restoration. So you should still use same credentials as before restoration.

8.3 Upgrade Firmware

Navigate through **Tools** > **Upgrade Firmware**

Update Mega PBX with the latest release available, which can contain key updates, added functionalities and bug fixes. When a new release is available, download it and save to your local PC. Then, browse for the file, and click the Upload button. Now your Mega PBX will display a Progress Screen and will prompt when it is about to reboot. Reboot and wait for blue LED's turn ON.

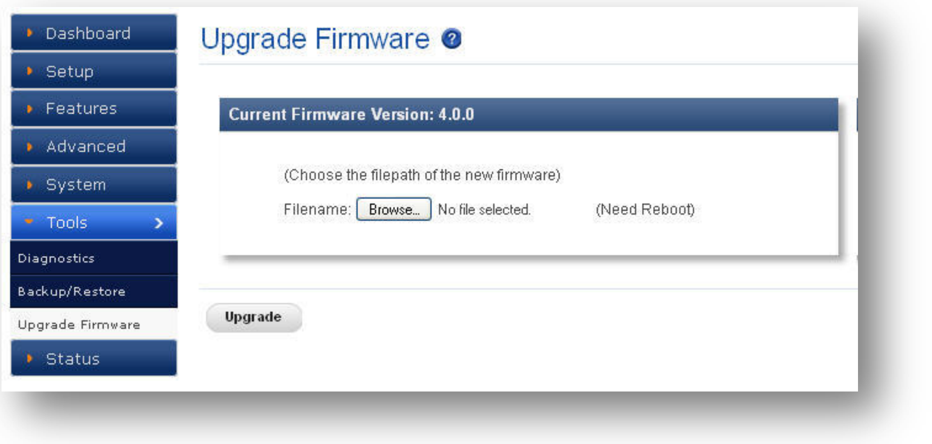


Figure 53: Upgrade Firmware

During firmware upgrade, there should not be any power or network disturbances, which may leads to Mega PBX board faulty. Firmware up-gradation process will take few minutes.

9. Status

This section generates the various status of the MegaPBX-PRI are explained below.

9.1 Call Reports

Navigate through **Status** > **Call Reports**

Call Reports displays a detailed list of calls pass through the MegaPBX-PRI. The list can be generated on the bases of date range, CDR count, latest 50 entries or all entries. Generated report can also be exported to local PC as CSV file or directed to a printer.

To create a new Report, select the Extension Range or Date range, and click the **"Generate"** Report button. A list with call details will display in the Call Reports section.

You can either export to your local PC or Print the Call reports.

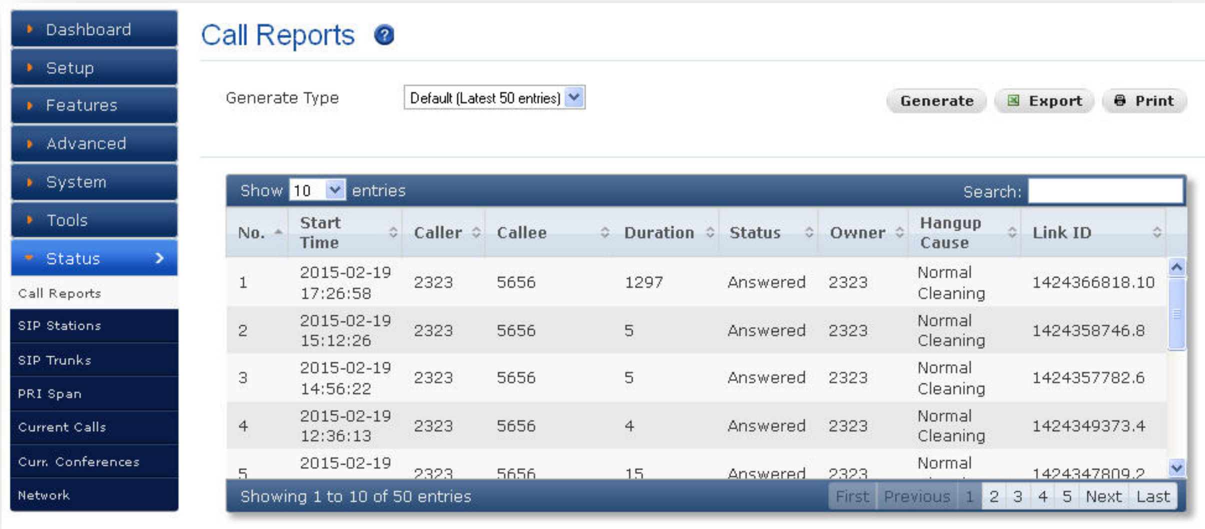


Figure 54: Call Reports

9.2 SIP Station

Navigate through **Status** > **Call Reports**

SIP Station Status page displays detailed status of each SIP Extensions available on the MegaPBX-PRI.

E.g.:

No.	Extension	Host	Status
1	1001	Unspecified	Registration Failed
2	1002	192.168.0.127	Registered

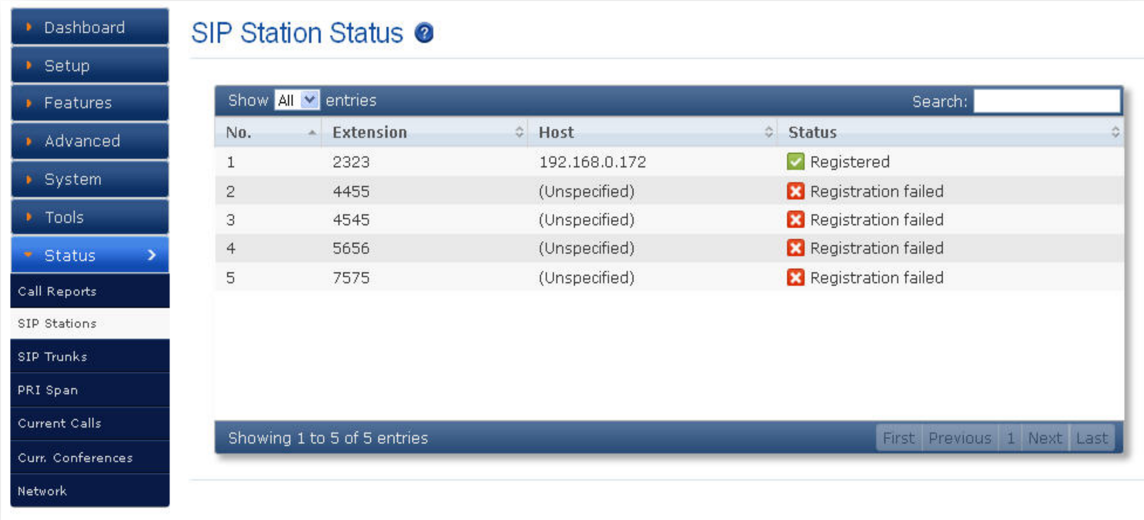


Figure 55: SIP Station Status

9.3 SIP Trunks

Navigation: **Status > SIP Trunks**

SIP Trunk Status page displays detailed status of each SIP trunks available on the MegaPBX-PRI.

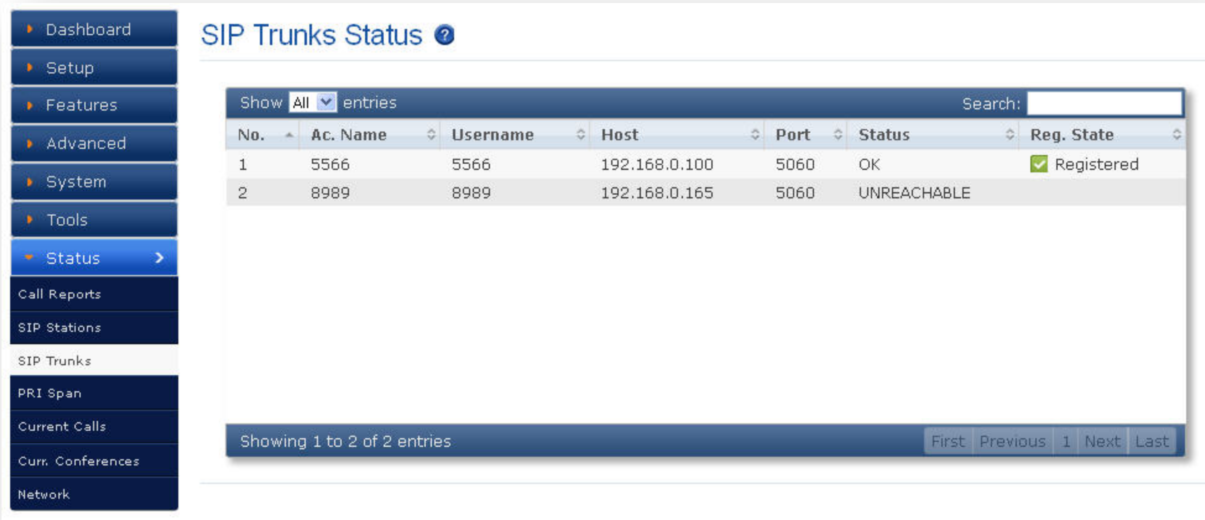


Figure 56: SIP Trunks Status

Status	Reg. State	Description	
OK	Registered	Configured, Registered & reachable	-
OK	-	Configured & Reachable, but	-

		no Registration	
OK	Request Sent	Configured, but Host not responding or unreachable	Check Registrar Address
OK	Rejected	Configured & reachable, but Registration failure	Check Authentication
UNREACHABLE	Registered	Configured, Registered, but not reachable	Check Proxy Address
UNREACHABLE	-	Configured, but not reachable	Check Proxy Address
UNKNOWN	-	Not Registered	Client not registered

Dynamic: Host IP is obtained dynamically on registration.

9.4 PRI Span

(Only for Mega PBX products with PRI support)

Navigate through **Status** > **PRI Span**

PRI Span Status page displays detail status of each E1/T1 Port with individual channel info, available on the PRI Gateway.

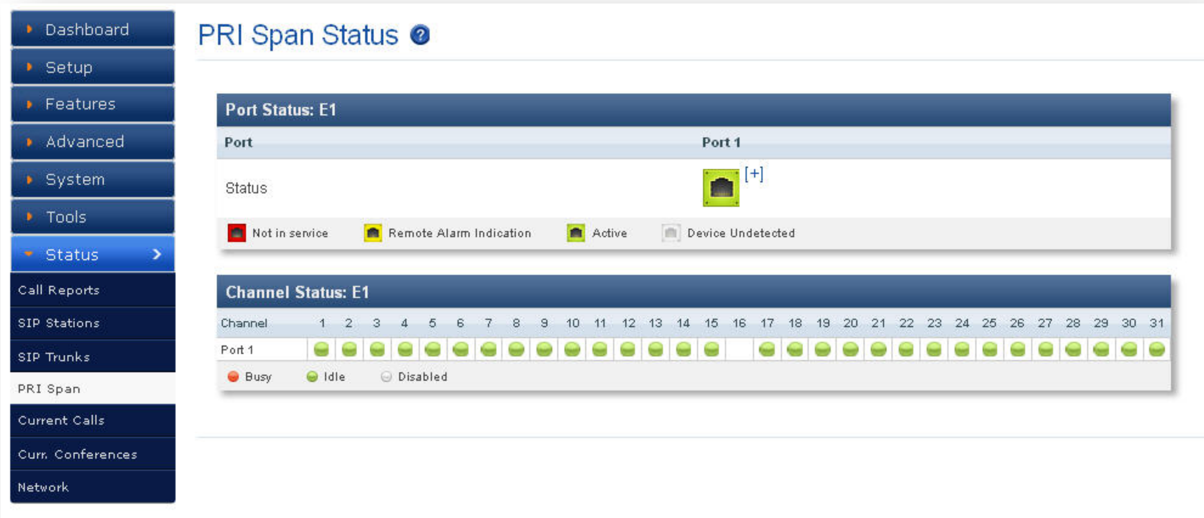
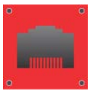
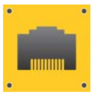


Figure 57: PRI Span Status

E1/T1 Port Status	Description
	Appliance is not seeing far end, circuit is not up, or cable is bad.
	Appliance is synchronizing or is receiving a yellow alarm from the far end.
	PRI Link is Active. Appliance is in-sync with the far end.
	T1/E1 driver is not initialized or device undetected.

Channel Status	Description
	Channel is Busy
	Channel is Idle and ready to receive or make calls
	Channel is not active

9.5 Current calls

Navigate through **Status** > **Current Calls**

Current Calls page displays detailed status of the real time calls available on Mega PBX.

9.6 Current Conferences

Navigate through **Status** > **Current Conferences**

Current Conferences page displays detailed status of the real time conference available on Mega PBX.

9.7 Network

Navigate through **Status** > **Network Status**

Network Status page displays detailed status of the network configuration on Mega PBX.

10. Administrator

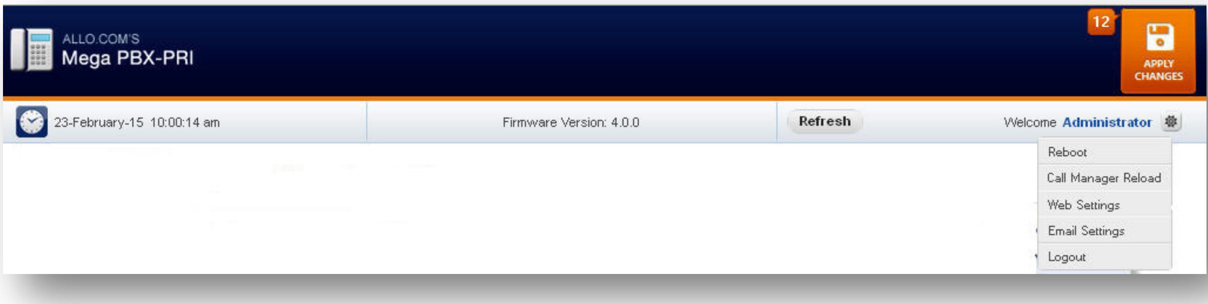


Figure 58: Administrator

10.1 Reboot

Navigate through > Reboot

Using this option administrator can reboot (SOFT reboot) MegaPBX System remotely

10.2 Call Manager Reload

Navigate through > Call Manager Reload

Reloading Call Manager will restart call manager and drop all current calls.

10.3 Web Settings

Navigate through > **Web Settings**

Session Timeout	Duration after which current web login session expires. Default: 3600 sec
Pagination	Number of entries in a table per page to be displayed.
Change Password	Modify Administrator password here.

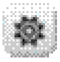
10.4 Email Settings

Navigate through > **Email Settings**

To configure the Mega PBX to send out voicemail/FAX via email, the related SMTP setting must be configured.

Mail Server	Enter the domain name of the Email Server address of the particular authorized email client account
Email ID	Enter the email ID of the particular authorized email client account.
Username	Email ID given by the Mail Server administrator.
Password	Password of the Email ID
TLS Support	To secure the server to server transfer of emails, the provider needs to enable a technology called Transport Layer Security (TLS).

10.5 Logout

Navigate through  > **Logout**

Administrator Logout option after use.

Thank you for choosing

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