DGW-L1 User Manual

1. Overview

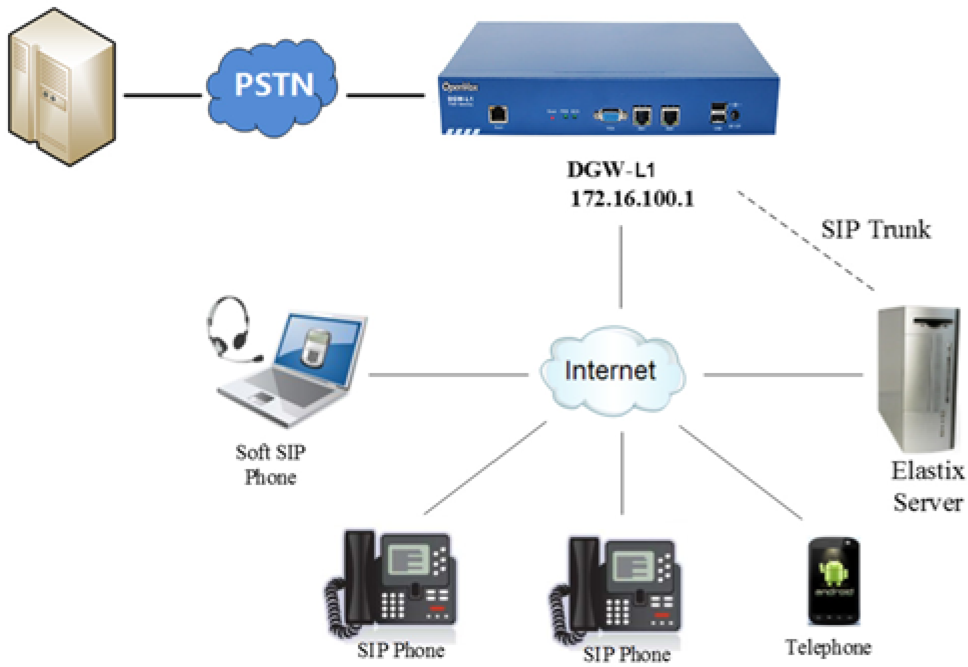
What is DGW-L1

OpenVox T1/E1 Gateway is an open source asterisk-based VoIP Gateway solution for operators and call centers. It is a converged media gateway product. This kind of gateway connects traditional telephone system to IP networks and integrates VoIP PBX with the PSTN seamlessly. With friendly GUI, users may easily setup their customized Gateway. Also secondary development can be completed through AMI (Asterisk Management Interface). The DGW-L1 could support 12v power supply.

It is developed with a wide selection of codecs and signaling protocol, including G.711A, G.711U, G.729, G.722, G.723 and GSM. It supports PRI/SS7/R2 protocol. OpenVox T1/E1 Gateway has good processing ability and stability. The T1/E1 gateway will be 100% compatible with Asterisk, Elastix, trixbox, 3CX, FreeSWITCH SIP server and VOS VoIP operating platform.

Sample Application

Figure 1-2-1Topological Graph



Product Appearance

The picture below is appearance of DGW-L1.

Figure 1-3-1 Product Appearance

Figure 1-3-2 Front Panel

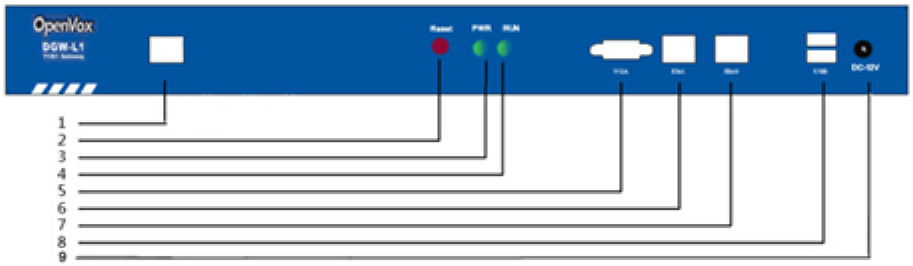


Table 1-3-1 Description of Front Panel

Interface	Function	Color	Work Status
1 Port	E1/T1 port. There is only one port.		
2 Reset	Reset button is used to restore the device.		
3 RUN	Register indicator	Green	Slow blinking(Green 2s and Flash 0.1s):Work normally
			Fast blinking(Green 0.5s and Flash 0.5s): Work abnormally
			Crazily blinking(Green 0.1s and Flash 0.1s): Preparing restore the device
			No blinking: Dahdi Error
4 PWR	Power Status indicator	Green	On: Power is on
			Off: Power is off
5 VGA	VGA monitor connector		
6 Eth1	Network interface		
7 Eth0	Network interface		
8 USB	USB interface		
9 DC-12v	Power supply		

Main Features

- Based on Asterisk^{R}
- Editable Asterisk^{R}configuration file
- Wide selection of codecs and signaling protocol
- Support 512 routing rules and flexible routing settings
- Stable performance, flexible dialing, friendly GUI
- Codecs support: G.711A, G.711U, G.729, G.723, G.722,GSM
- Support ports group management
- Connect legacy PBX systems to low-cost VoIP services
- Connect legacy PBX systems to remote sites over private VoIP links
- Connect IP PBX systems to legacy TDM services

Physical Information

Table 1-5-1 Description of Physical Information

Weight	1314g

Size	31cm*17cm*5cm
Temperature	-40~85°C (Storage)
	0~40°C (Operation)
Operation humidity	5%~95% non-condensing
Max power	12W
LAN port	1
WAN port	1

Software

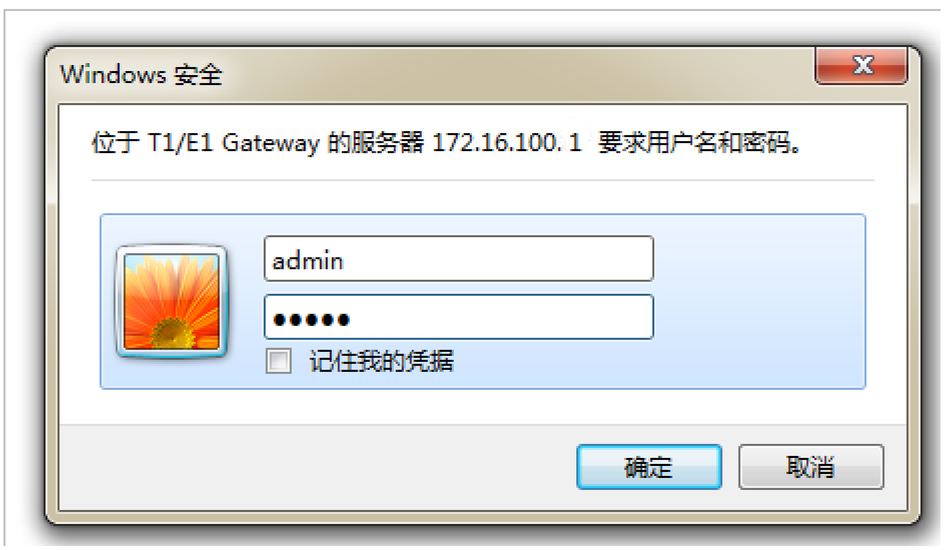
Default IP: 172.16.100.1(Eth0)192.168.100.1(Eth1)

Username: admin

Password: admin

Notice: Log in

Figure 1-6-1 LOG IN Interface



2. System

Status

On the “Status” page, you will find all Interface, Status, Time, Login Settings, General, Auto Provision, Tools and Information.

Figure 2-1-1 System Status

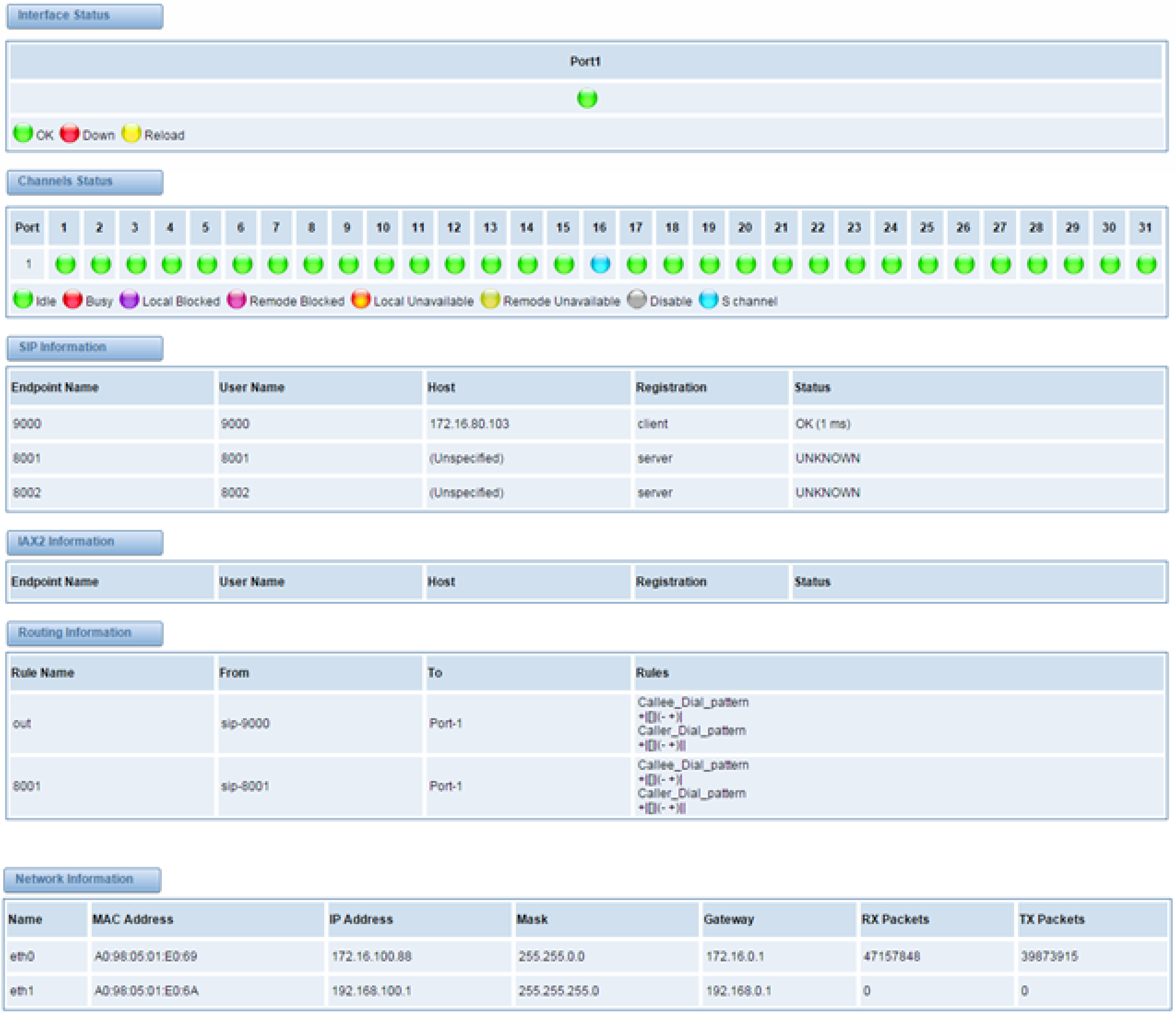


Table 2-1-1 Description of System Status

Options	Definition
Interface Status	Show the status of port, include "RED" and "OK". "RED" means no trunk line connected; "OK" means the trunk line of port is available.
Signaling Status	Show the signaling status of port, include "Down" and "UP". "Down" means it is unavailable; "UP" means the port is available.

Time

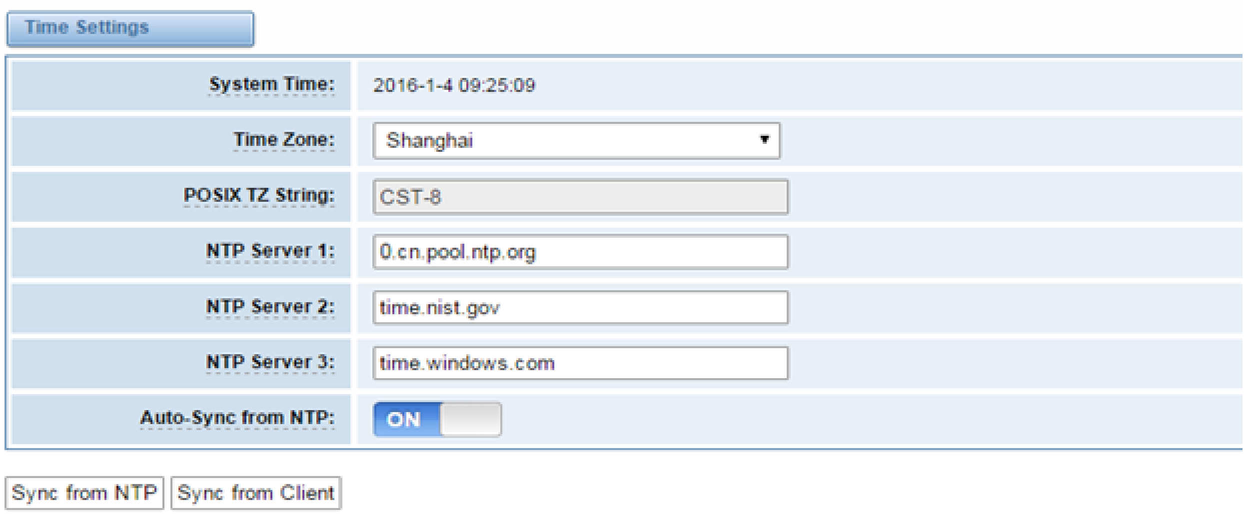
Table 2-2-1Description of Time Settings

Options	Definition
System Time	Your gateway system time.
Time Zone	The world time zone. Please select the one which is the same or the closest as your city.
POSIX TZ String	Posix timezone strings.

NTP Server 1	Time server domain or hostname. For example, [time.asia.apple.com].
NTP Server 2	The first reserved NTP server. For example, [time.windows.com].
NTP Server 3	The second reserved NTP server. For example, [time.nist.gov].
Auto-Sync from NTP	Whether enable automatically synchronize from NTP server or not. ON is enable, OFF is disable this function.
Sync from NTP	Sync time from NTP server.
Sync from Client	Sync time from local machine.

For example, you can configure like this:

Figure 2-2-1 Time Settings



You can set your gateway time Sync from NTP or Sync from Client by pressing different buttons.

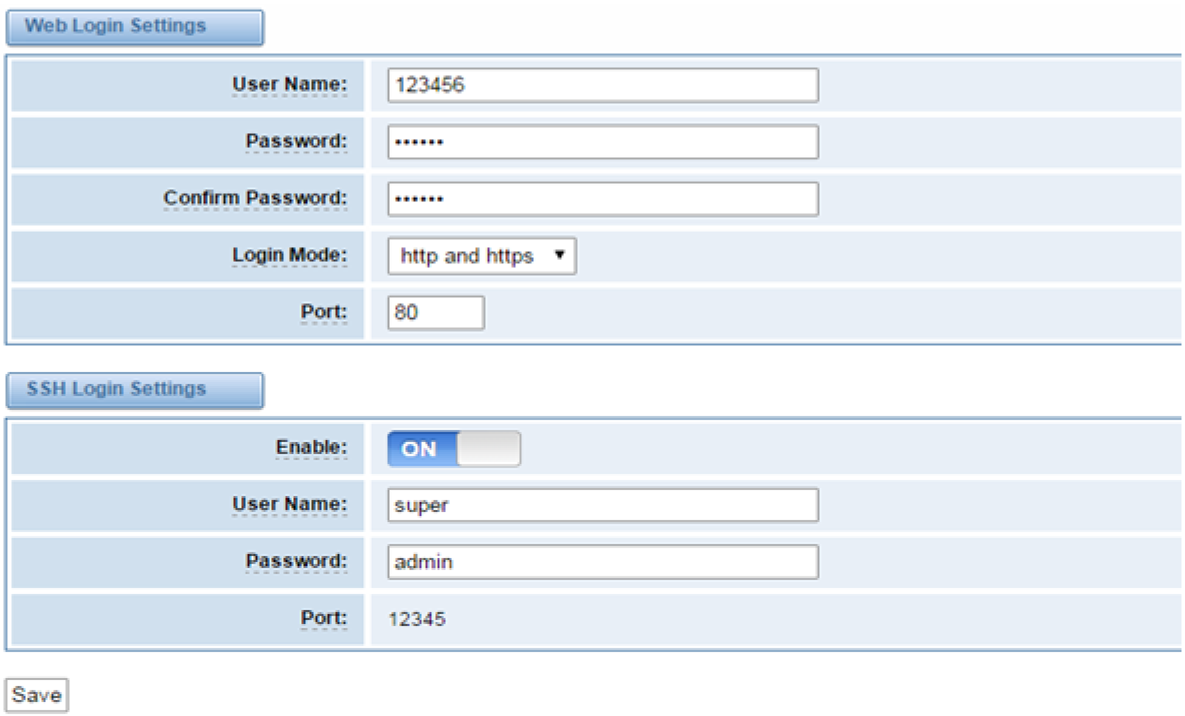
Login Settings

Your gateway doesn't have administration role. All you can do here is to reset what new username and password to manage your gateway. And it has all privileges to operate your gateway. You can modify “Web Login Settings” and “SSH Login Settings”. If you have changed these settings, you don’t need to log out, just rewriting your new user name and password will be OK. Also you can specify the web server port number.

Table 2-3-1Description of Login Settings

Options	Definition
User Name	Define your username and password to manage your gateway, without space here. Allowed characters "-_+. <>&0-9a-zA-Z". Length: 1-32 characters.
Password	Allowed characters "-_+. <>&0-9a-zA-Z". Length: 4-32 characters.
Confirm Password	Please input the same password as 'Password' above.
Port	Specify the web server port number.

Figure 2-3-1 Login Settings



Notice: Whenever you do some changes, do not forget to save your configuration.

General

Language Settings

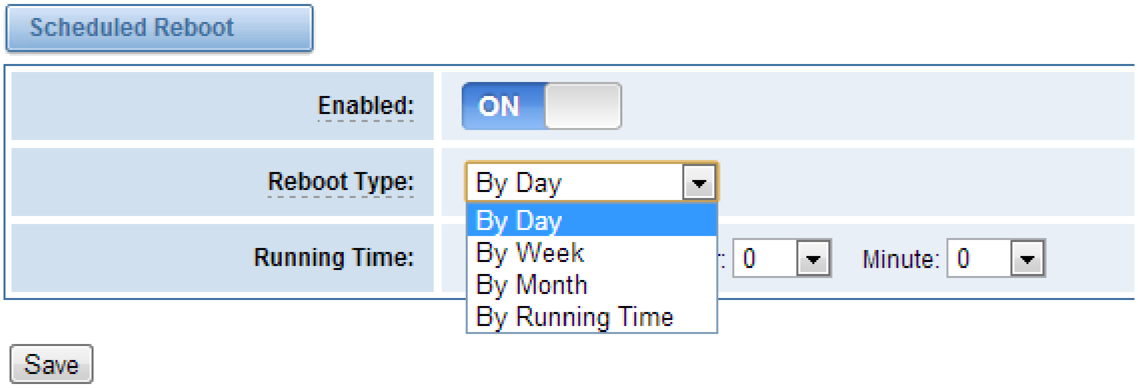
You can choose different languages for your system. If you want to change language, you can switch "Advanced" on, then "Download" your current language package. After that, you can modify the package with the language you need. Then upload your modified packages, "Choose File" and "Add".

Figure 2-4-1 Language Settings

Scheduled Reboot

If switch it on, you can manage your gateway to reboot automatically as you like. There are four reboot types for you to choose, "By Day, By Week, By Month and By Running Time".

Figure 2-4-2 Reboot Types



If use your system frequently, you can set this enable, it can helps system work more efficient.

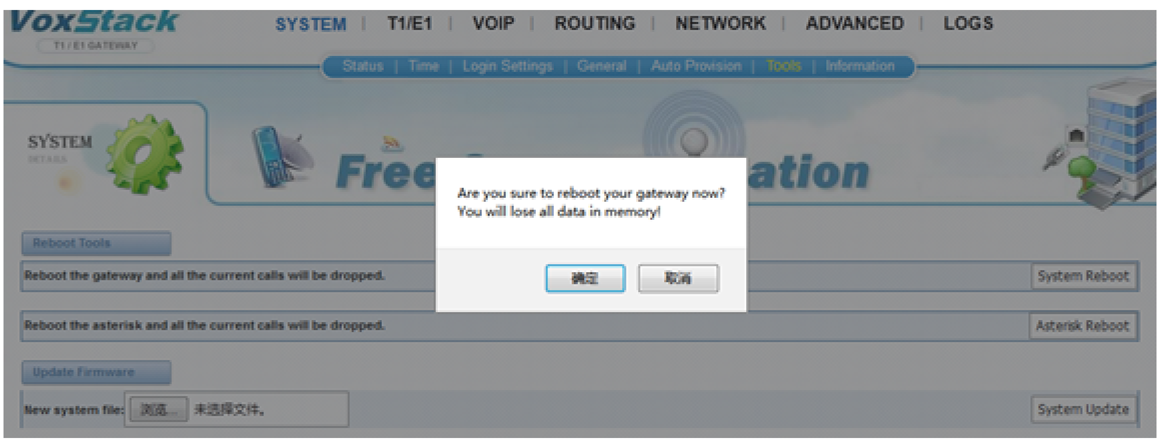
Tools and Information

On the “Tools” pages, there are reboot Tools, update Firmware, upload Configuration, backup Configuration and Restore Configuration toolkits.

Reboot Tools

You can choose system reboot and Asterisk reboot separately.

Figure 2-6-1 Reboot Prompt



If you press “OK”, your system will reboot and all current calls will be dropped. Asterisk Reboot is the same.

Table 2-6-1 Instruction of reboots

Options	Definition
System Reboot	This will turn off your gateway and then turn it back on. This will drop all current calls.
Asterisk Reboot	This will restart Asterisk and drop all current calls.

Update Firmware

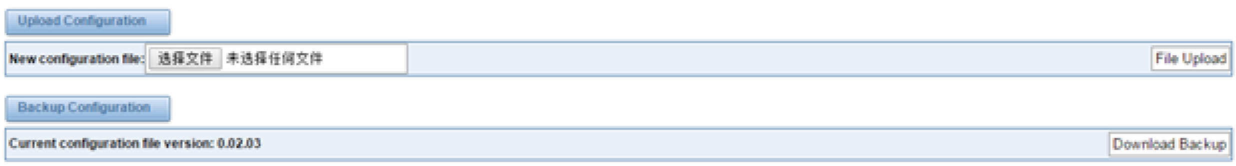
We offer 2 kinds of update types for you, you can choose System Update or System Online Update. System Online Update is an easier way to update your system, if you choose that, you will see some information below.

Figure 2-6-2Prompt Information

Upload and Backup Configuration

If you want to update your system and remain your previous configuration, you can first backup configuration, then you can upload configuration directly. That will be very convenient for you.

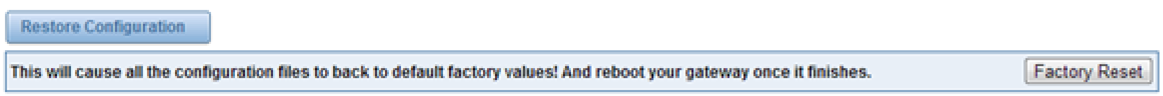
Figure 2-6-3 Upload and Backup



Restore Configuration

Sometimes there is something wrong with your gateway that you don’t know how to solve it, mostly you will select factory reset. Then you just need to press a button, your gateway will be reset to the factory status.

Figure 2-6-4 Factory Reset



Information

On the “Information” page, there shows some basic information about the T1/E1 gateway. You can see software and hardware version, storage usage, memory usage and some help information.

Figure 2-6-5 System Information

3. T1/E1

General

Figure 3-1-1 General Settings

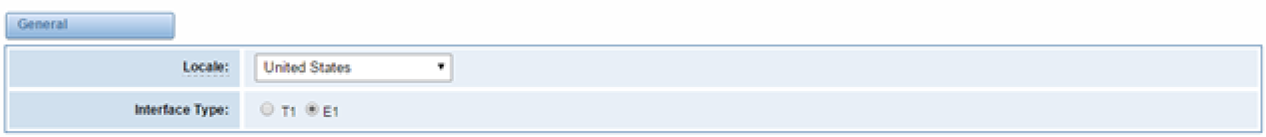


Table 3-1-1 Definition of General Settings

Options	Definition
Local	Your locale. This will be used for the tone style used when in-call indications need to be generated such as ring back, busy, congestion, and other call-oriented inband tone signals.

Figure 3-1-2 Port Details

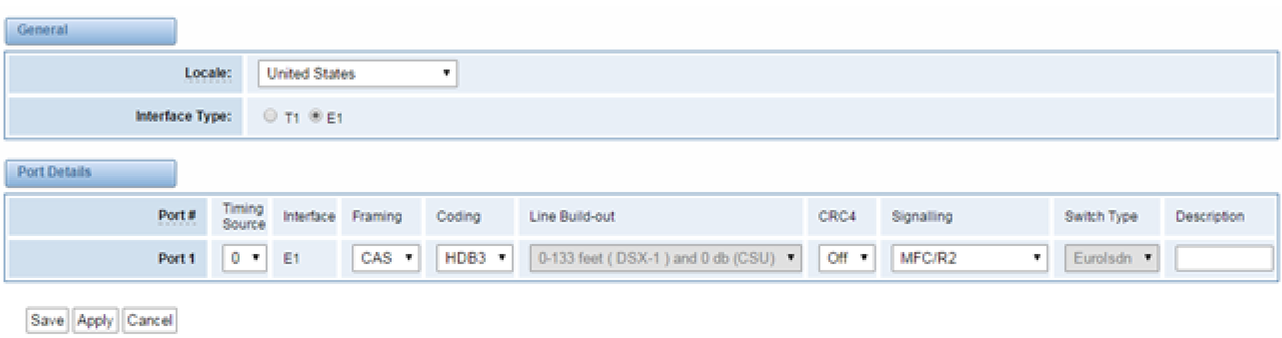


Table 3-1-3 Definition of Port Details

Options	Definition
Timing Source	Timing Source indicate the ports as to which should be used to recover the clock.(0 for master mode, upper for client mode, small number have higher priority)
Interface	Choose a line type for this interface, all ports must be the same type.
Framing	Framing method for this interface
Coding	Coding method for this interface
Line Build-out	Line build-out represents the length of the cable form the port on this gateway to the next device.
CRC4	Enable cyclic redundancy checking for error checking on line. CRC-4 support is required for all network switches in Europe, but many older switches and PBXs don’t support it.
Signaling	It shows you what signaling the port uses.
Switch Type	Only used for PRI
Description	An optional description of this interface to be used for reference only.

ISDN-PRI

Advanced: Interface Type

Figure 3-2-1 Advanced: Interface Type

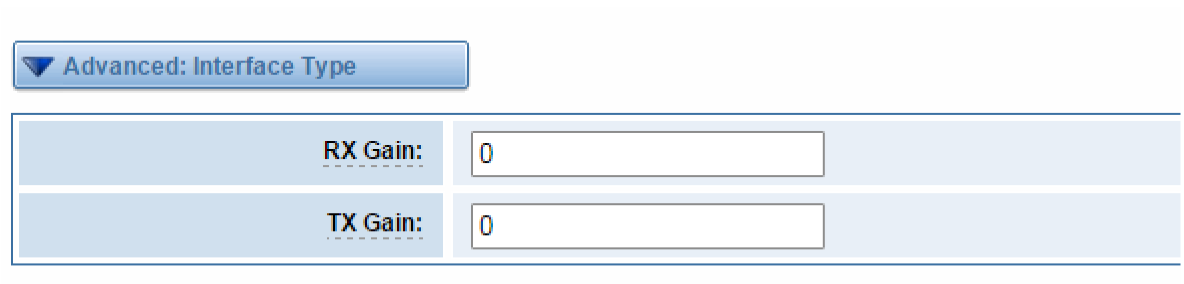


Table 3-2-1Definition of Interface Type

Options	Definition
RX Gain	Gain for the rx channel.Default:0.0
TX Gain	Gain for the tx channel.Default:0.

ISDN: Signaling

Figure 3-2-2 ISDN: Signaling

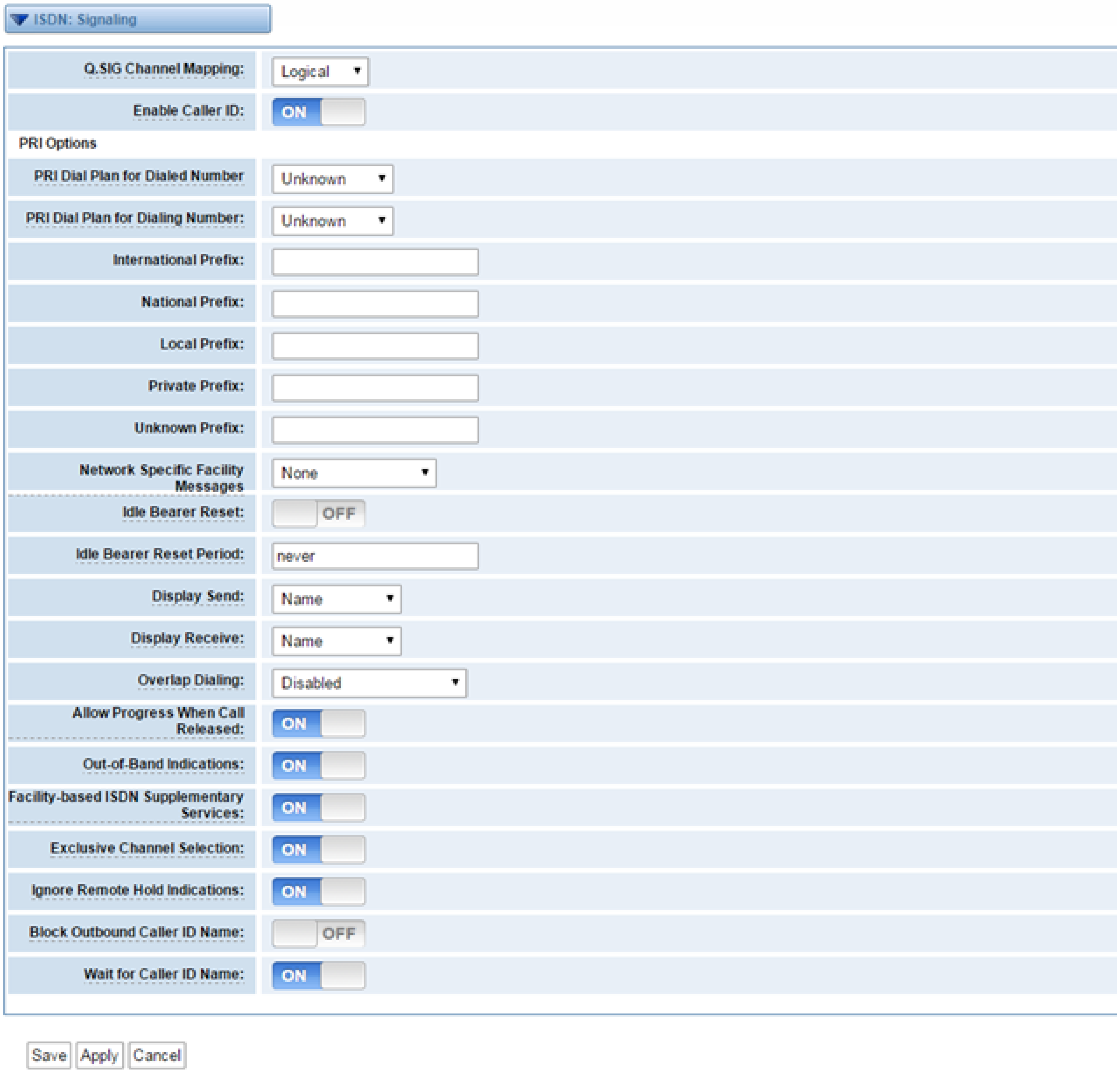


Table 3-2-2 Definition of Signaling

Options	Definition
Q.SIG Channel Mapping	Sets logical or physical channel mapping. In logical channel mapping, channels are mapped to 1-30. In physical channel mapping, channels are mapped to 1-15, 17-31, skipping the number used for the data channel. Default is physical.
Enable Caller ID	Whether or not to enable caller ID.
PRI Dial Plan for Dialed Number	PRI Dialplan: The ISDN_level Type Of Number or numbering plan, used for the dialed number. Leaving this as ‘unknown’ works for most case. In some very unusual circumstances, you may need to set this to ‘dynamic’ or ‘redundant’.
PRI Dial Plan for Dialing Number	PRI Local Dialplan: Only RARELY used for PRI(sets the calling numbre's numbering plan). In North America, the typical use is sending the 10 digit; callerID number and setting the prilocaldialplan to 'national' (the default); Only VERY rarely will you need to change this.
Network Specific Facility (NSF) Messages	Some switches (AT&T especially) require network specific facility IE supported values are currently 'none', 'sdn', 'megacom', 'tollfreemegacom', 'accunet'
Idle Bearer Reset	Whether or not to reset unused B channels.

Idle Bearer Reset Period	Time in seconds between reset of unused B channels.
Display Send	Send /receive ISDN display IE option. The display option are a comma separated list of the following option: Block: Do not pass display text data. Name_initial: Use display text in SETUP/CONNECT messages as the party name. Name_update: Use display text in other messages NOTIFY/FACLITY for CLOP name update. Name: Combined name_initial and name_update options. Text: Pass any unused display text data as an arbitrary display message during a call. Send text goes out in an INFORMATION message. Defaults to name.
Display Receive	Send /receive ISDN display IE option. The display option are a comma separated list of the following option: Block: Do not pass display text data. Name_initial: Use display text in SETUP/CONNECT messages as the party name. Name_update: Use display text in other messages NOTIFY/FACLITY for CLOP name update. Name: Combined name_initial and name_update options. Text: Pass any unused display text data as an arbitrary display message during a call. Send text goes out in an INFORMATION message. Defaults to name.
Overlap Dialing	Enable overlap dialing mode--sending overlap digits.
Allow Progress When Call Released	Allow inband audio (progress) when a call is RELEASEd by the far end of a PRI.
Out-of-Band Indications	PRI Out of band indications. Enable this to report Busy and Congestion on a PRI using out-of-band notification. Inband indication, as used by the gateway doesn't seem to work with all telcos.
Facility-based ISDN Supplementary Services	To enables transmission of facility-based ISDN supplementary services (such as caller name from CPE over facility). Cannot be changed on a reload.
Exclusive Channel Selection	If you need to override the existing channels selection routine and force all PRI channels to be marked as exclusively selected, set this to yes. priexclusive cannot be changed on a reload.
Ignore Remote Hold Indications	If you wish to ignores remote hold indications enable this option.

Block Outbound Caller ID Name	Enable if you need to hide the name and not the number for legacy PPBX use. Only applies to PRI channels.
Wait for Caller ID Name	Support Caller ID on call waiting.

SS7

Link Set Settings

Figure 3-3-1 Link Set Settings

You can click button as shown below, when there are several link set, only one can be set to the default.

Figure 3-3-2 SS7 Link Set Settings

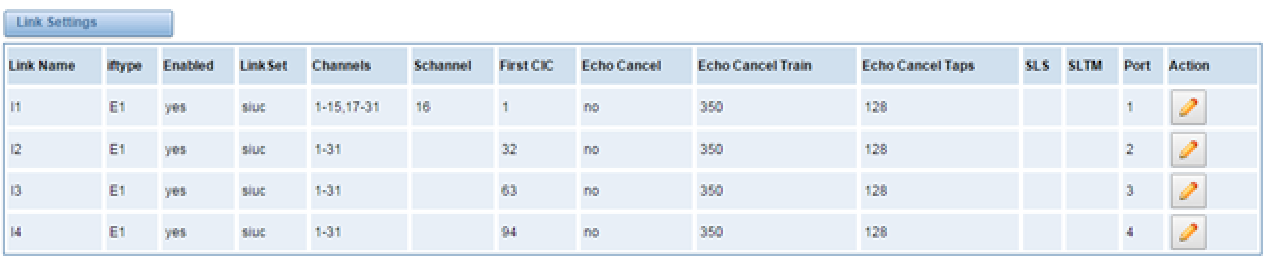
Table 3-3-1 Definition of SS7 Link Set Settings

Options	Definition
Name	The linkset's name
Enabled	The linkset is enable or disable
Enabled_ st	The end_of_pulsing (ST) is not used to determine when incoming address is complete

Use Connect	Reply incoming call with CON rather than ACM and ANM
Hunting Policy	The CIC hunting policy (even_mu, odd_lru, seq_lth, seq_htl) is even CIC numbers, most recently used
Subservice	The subservice field: national (8), international l(0), auto or decimal/hex value; The auto means that the subservice is obtained from first received SLTM.
t35	The value and action for t35. Value is in msec, action is either st or timeout; if you use overlapped dialing dial plan, you might choose:t35=>4000,st
variant	Running under SS7 standard
OPC	The point code for this SS7 signaling point
DPC	The destination point (peer) code
Set to Default	Set the linkset as the default linke set

Link Settings

Figure 3-3-3 Link Settings

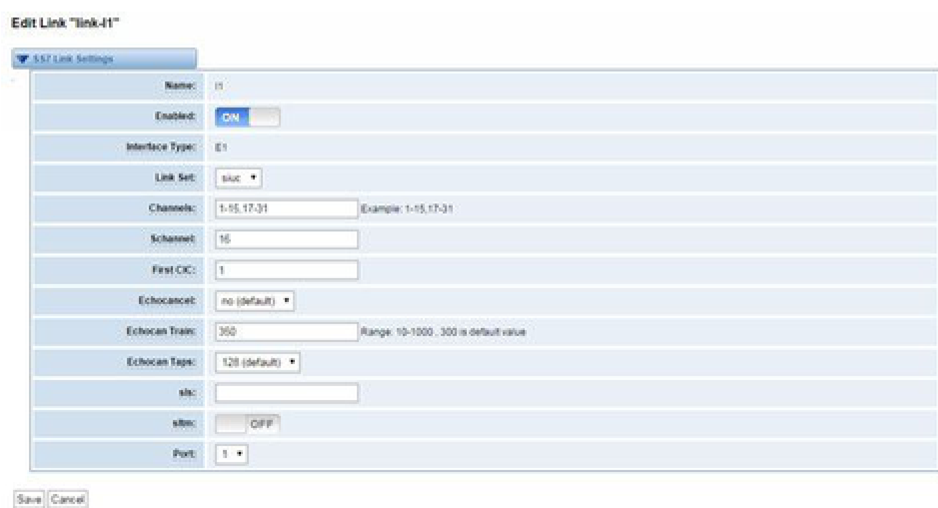


Link Settings

Link Name	iftype	Enabled	Link Set	Channels	Schannel	First CIC	Echo Cancel	Echo Cancel Train	Echo Cancel Taps	SLS	SLTM	Port	Action
l1	E1	yes	siuc	1-15,17-31	16	1	no	350	128			1	
l2	E1	yes	siuc	1-31		32	no	350	128			2	
l3	E1	yes	siuc	1-31		63	no	350	128			3	
l4	E1	yes	siuc	1-31		94	no	350	128			4	

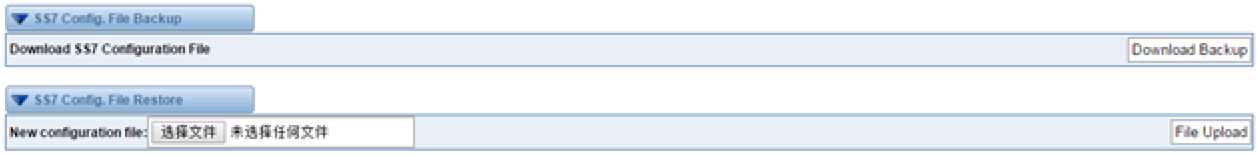
You can click button as shown below.

Figure 3-3-4 SS7 Link Settings



SS7 Config. File Backup and Restore

Figure 3-3-5 Config. File Backup and Restore



MFC/R2

Advanced: Interface Type

Figure 3-4-1 Advanced: Interface Type

Table 3-4-1 Definition of Interface Type

Options	Definition
RX Gain	Gain for the rx channel. Default:0.0
TX Gain	Gain for the tx channel. Default:0.0

MFC/R2: Signaling

Figure 3-4-2 MFC/R2: Signaling

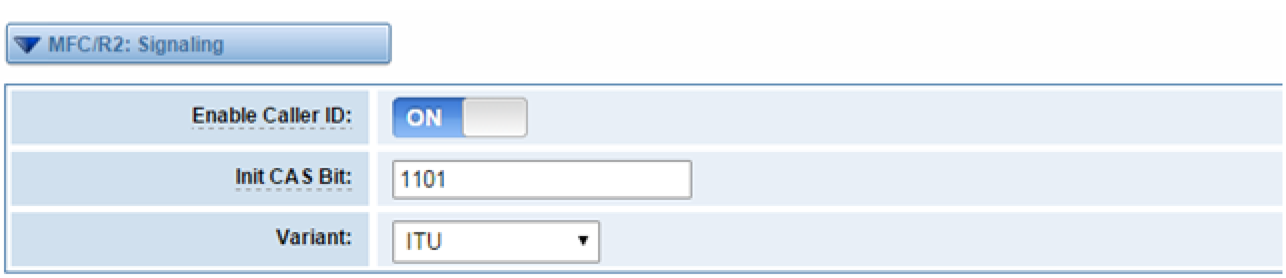
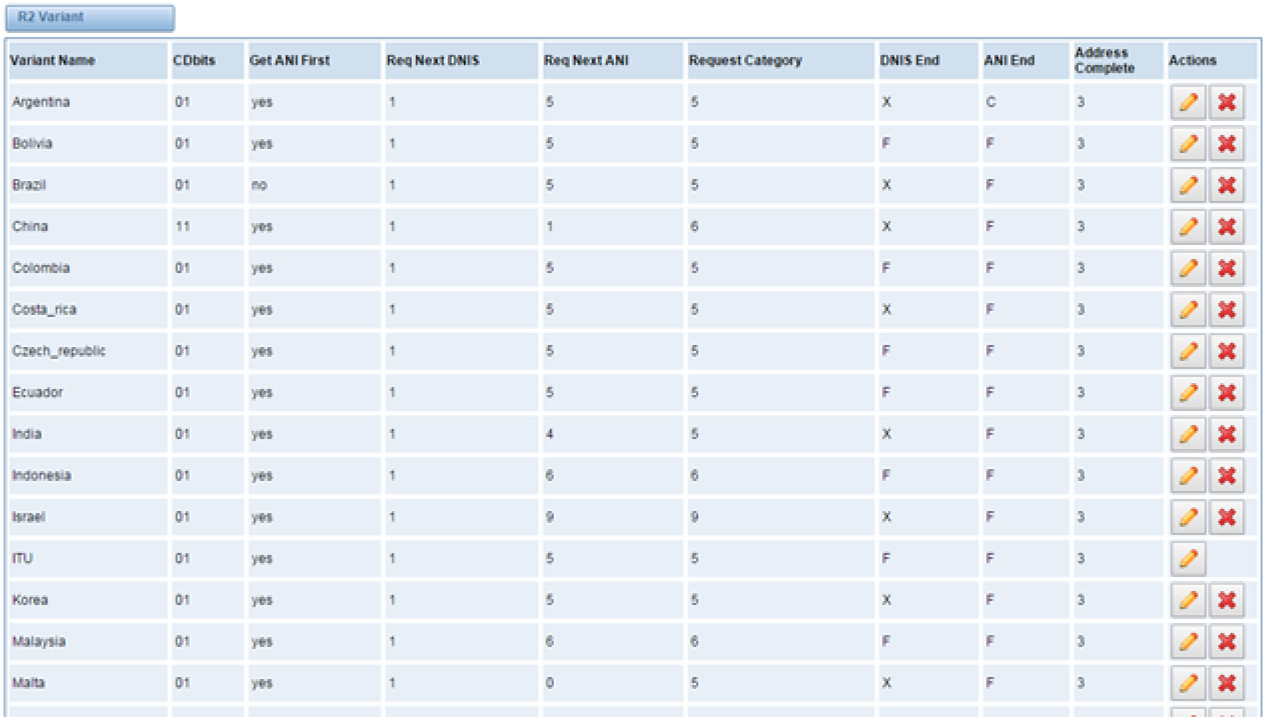


Table 3-4-2Definition of MFC/R2: Signaling

Options	Definition
Enable Caller ID	Whether or not to use caller ID
Init CAS Bit	The initial position of the CAS bits.

Figure 3-4-3 R2 Variant



R2 Variant

Variant Name	CDbits	Get ANI First	Req Next DNIS	Req Next ANI	Request Category	DNIS End	ANI End	Address Complete	Actions
Argentina	01	yes	1	5	5	X	C	3	
Bolivia	01	yes	1	5	5	F	F	3	
Brazil	01	no	1	5	5	X	F	3	
China	11	yes	1	1	6	X	F	3	
Colombia	01	yes	1	5	5	F	F	3	
Costa_rica	01	yes	1	5	5	X	F	3	
Czech_republic	01	yes	1	5	5	F	F	3	
Ecuador	01	yes	1	5	5	F	F	3	
India	01	yes	1	4	5	X	F	3	
Indonesia	01	yes	1	6	6	F	F	3	
Israel	01	yes	1	9	9	X	F	3	
ITU	01	yes	1	5	5	F	F	3	
Korea	01	yes	1	5	5	X	F	3	
Malaysia	01	yes	1	6	6	F	F	3	
Malta	01	yes	1	0	5	X	F	3	

you can click button, then you could fine the below.

Modify R2 Variant

Figure 3-4-4 General

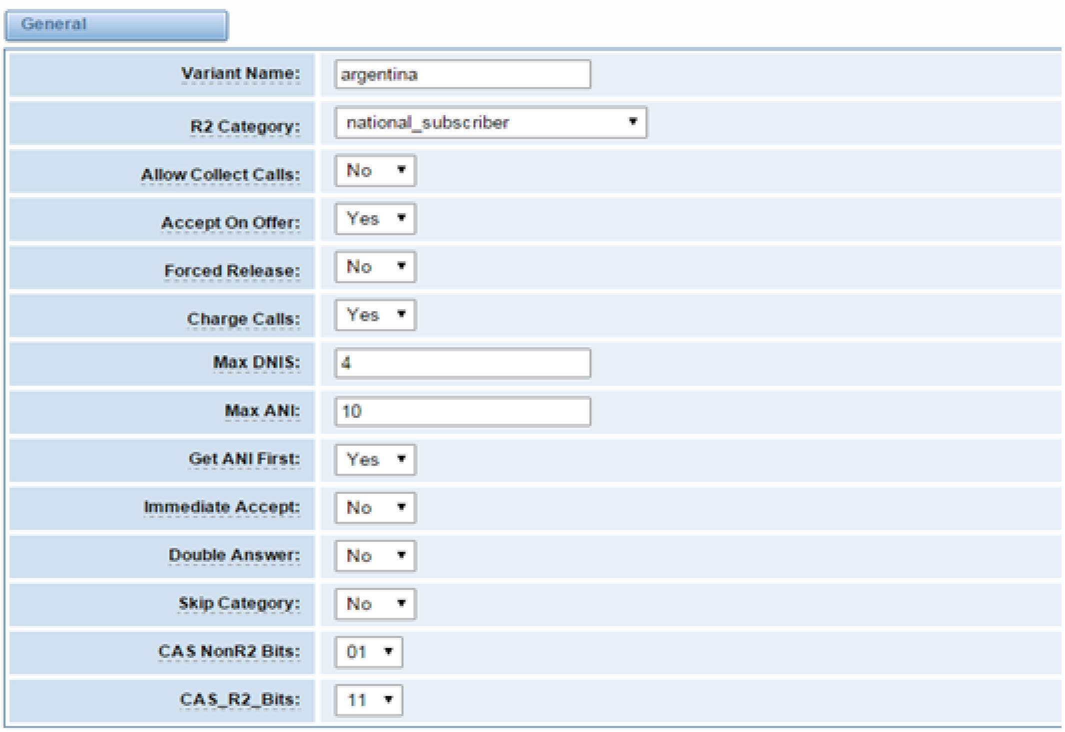


Table 3-4-3 Definition of General

Options	Definition

<table>
<tr><td>Variant Name</td><td>
<table>
<tr><td>Variant Name</td><td></td></tr>
<tr><td>Usually national_subscriber works just fine</td><td></td></tr>
<tr><td>Default is to block collect calls</td><td></td></tr>
<tr><td>With this set to ‘no’ then the call will NOT be accepted on offered, and the call will start irs execution in extensions. Conf until the channel is answered.</td><td></td></tr>
<tr><td>Brazil use a special signal to force the release of the line instead of the normal clear back signal</td><td></td></tr>
<tr><td>Whether or not report to the other end ‘accept call with charge’, when interconnecting with old PBXs this may be useful</td><td></td></tr>
<tr><td>Max amount of DNIS to ask for</td><td></td></tr>
<tr><td>Max amount of ANI to ask for</td><td></td></tr>
<tr><td>Whether or not get the ANI before getting DNIS</td><td></td></tr>
<tr><td>This feature allows to skip the use of Group B/II signals and go directly to the accepted state for incoming calls</td><td></td></tr>
<tr><td>This will cause that every answer signal is changed by answer->clear back->answer, sort of flash</td><td></td></tr>
<tr><td>Skip request of calling party category and ANI</td><td></td></tr>
<tr><td>Which bits are never used</td><td></td></tr>
<tr><td>Which bits are be used</td><td></td></tr>
</table>
</td></tr>
<tr><td>R2 Category</td><td>Usually national_subscriber works just fine</td></tr>
<tr><td>Allow Collect Calls</td><td>Default is to block collect calls</td></tr>
<tr><td>Accept On Offer</td><td>With this set to ‘no’ then the call will NOT be accepted on offered, and the call will start irs execution in extensions. Conf until the channel is answered.</td></tr>
<tr><td>Forced Release</td><td>Brazil use a special signal to force the release of the line instead of the normal clear back signal</td></tr>
<tr><td>Charge Calls</td><td>Whether or not report to the other end ‘accept call with charge’, when interconnecting with old PBXs this may be useful</td></tr>
<tr><td>Max DNIS</td><td>Max amount of DNIS to ask for</td></tr>
<tr><td>Max ANI</td><td>Max amount of ANI to ask for</td></tr>
<tr><td>Get ANI First</td><td>Whether or not get the ANI before getting DNIS</td></tr>
<tr><td>Immediate Accept</td><td>This feature allows to skip the use of Group B/II signals and go directly to the accepted state for incoming calls</td></tr>
<tr><td>Double Answer</td><td>This will cause that every answer signal is changed by answer->clear back->answer, sort of flash</td></tr>
<tr><td>Skip Category</td><td>Skip request of calling party category and ANI</td></tr>
<tr><td>CAS NonR2 Bits</td><td>Which bits are never used</td></tr>
<tr><td>CAS_R2_Bits</td><td>Which bits will be used</td></tr>
</table>

Figure 3-4-5 Timer

Table 3-4-4 Definition of Timer

Options	Definition
MF Back Cycle	Max amount of time our backward MF signal can last
MF Back Resume Cycle	Amount of time we set MF signal ON to resume the MF cycle with a MF pulse
MF Fwd Safety	Safety FORWARD timer
R2 Seize	How much time do we wait for a response to our seize signal
R2 Answer	How much to wait for an answer once the call has been accepted
Metering Pulse	Hoe much to wait for metering pulse detection
R2 Double Answer	Interval between ANSWER-CLEAR BACK-ANSWER when double answer is in effect
R2 Answer Delay	Minimum delay time between the Accept tone signal and the R2 answer signal
CAS Persistence Check	Time to wait for to CAS signaling before handing the new signal
DTMF Start Dial	Safety time before starting to dial DTMF
DTMF Detection End	Safety time to decide when to stop detecting DTMF DNIS.

Figure 3-4-6 Group A

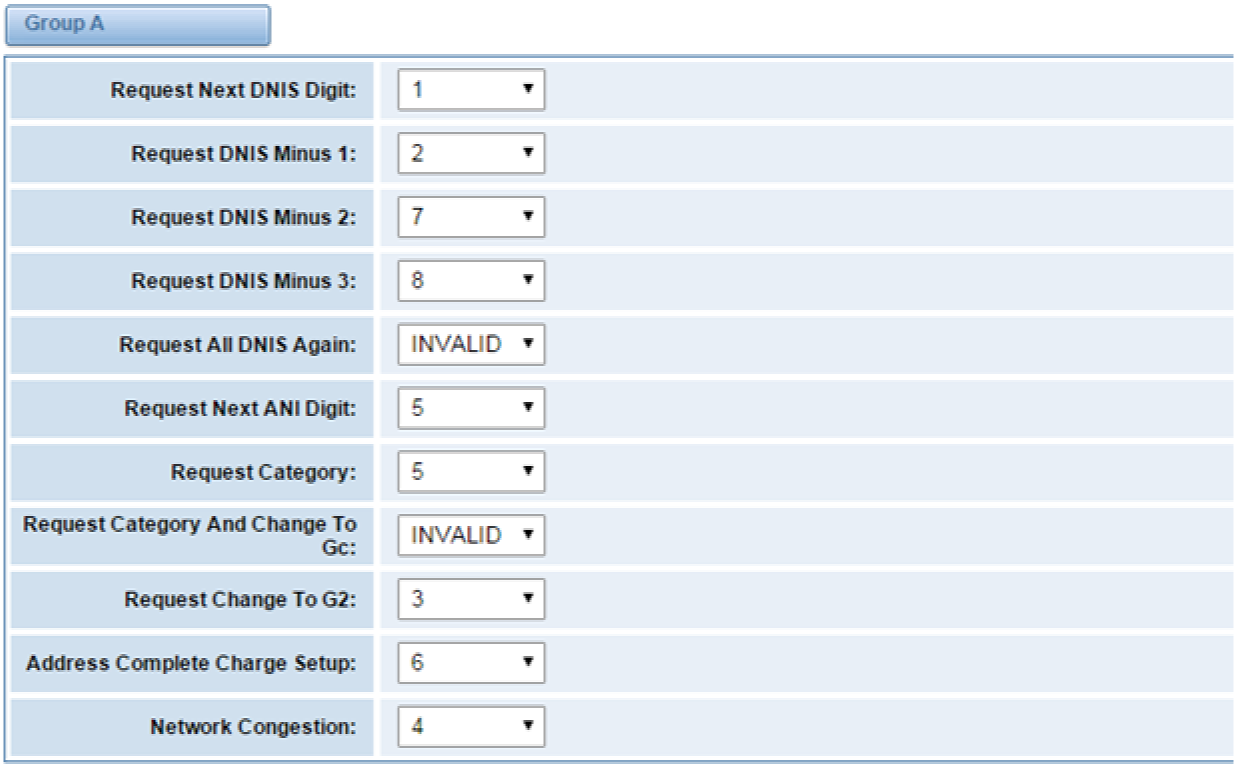


Figure 3-4-7 Group B

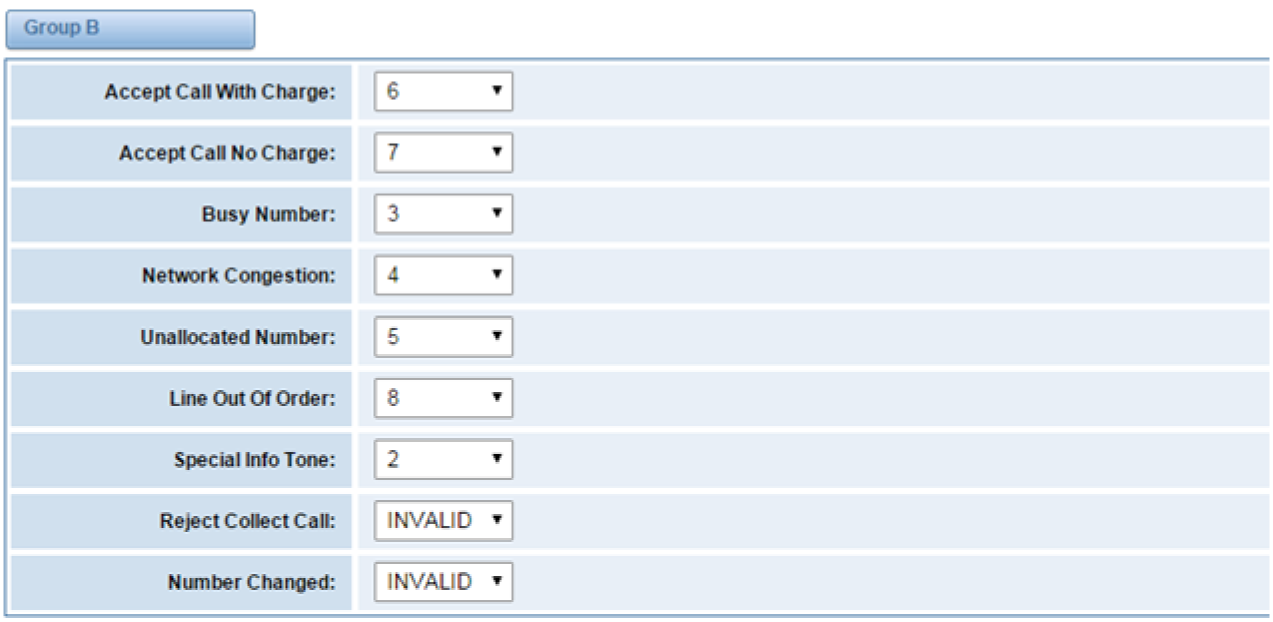


Figure 3-5-8 Group C

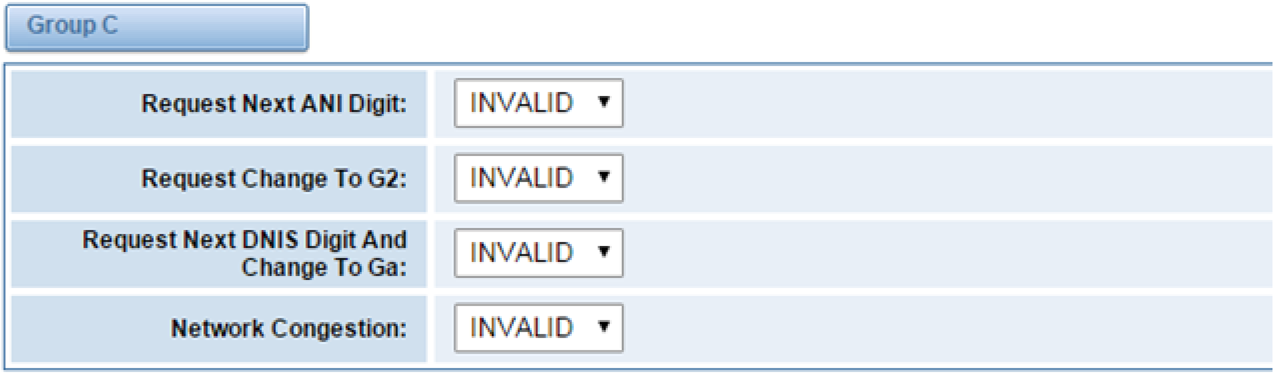


Figure 3-4-9 Group 1

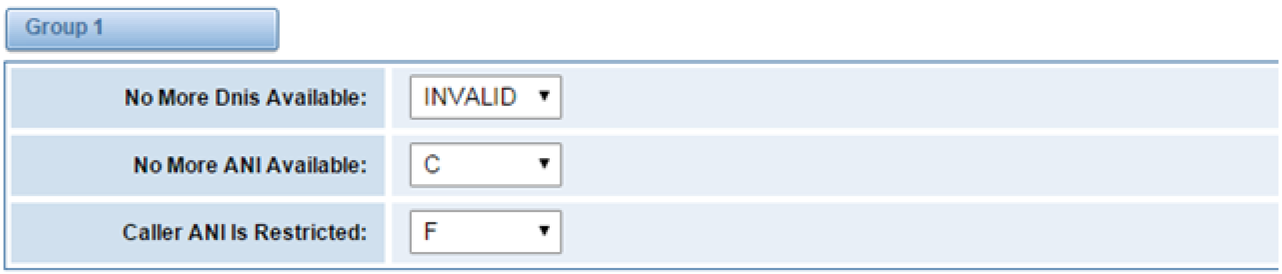
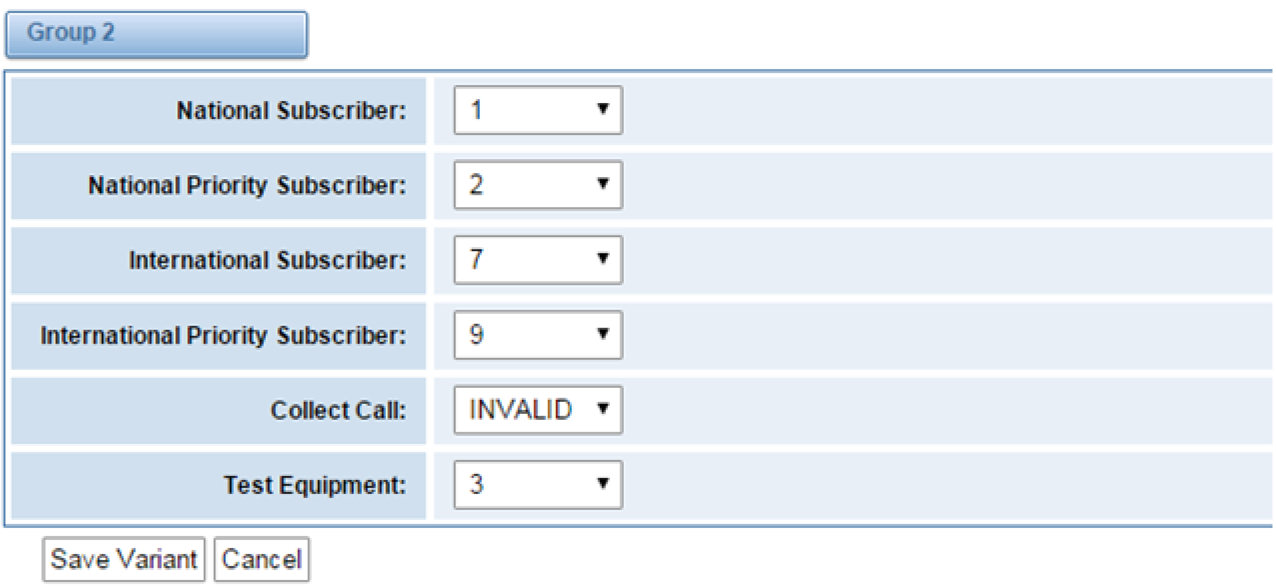


Figure 3-4-810Group 2



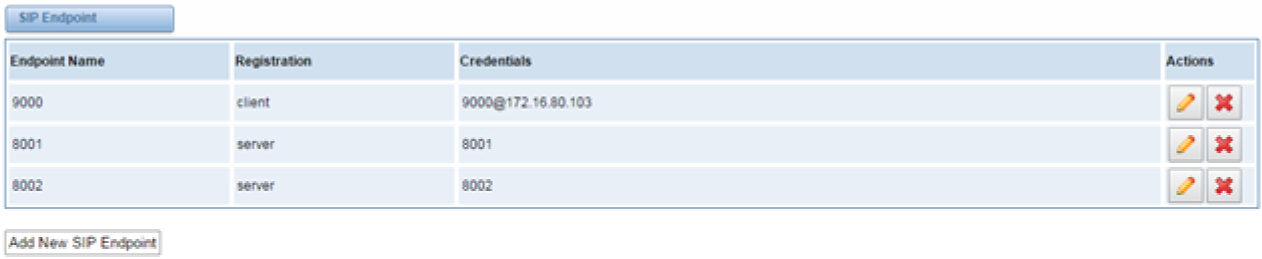
4.VOIP

VOIP Endpoints

SIP Endpoints

This page shows everything about your SIP, you can see status of each SIP.

Figure 4-1-1 SIP Status



Main Endpoint Settings

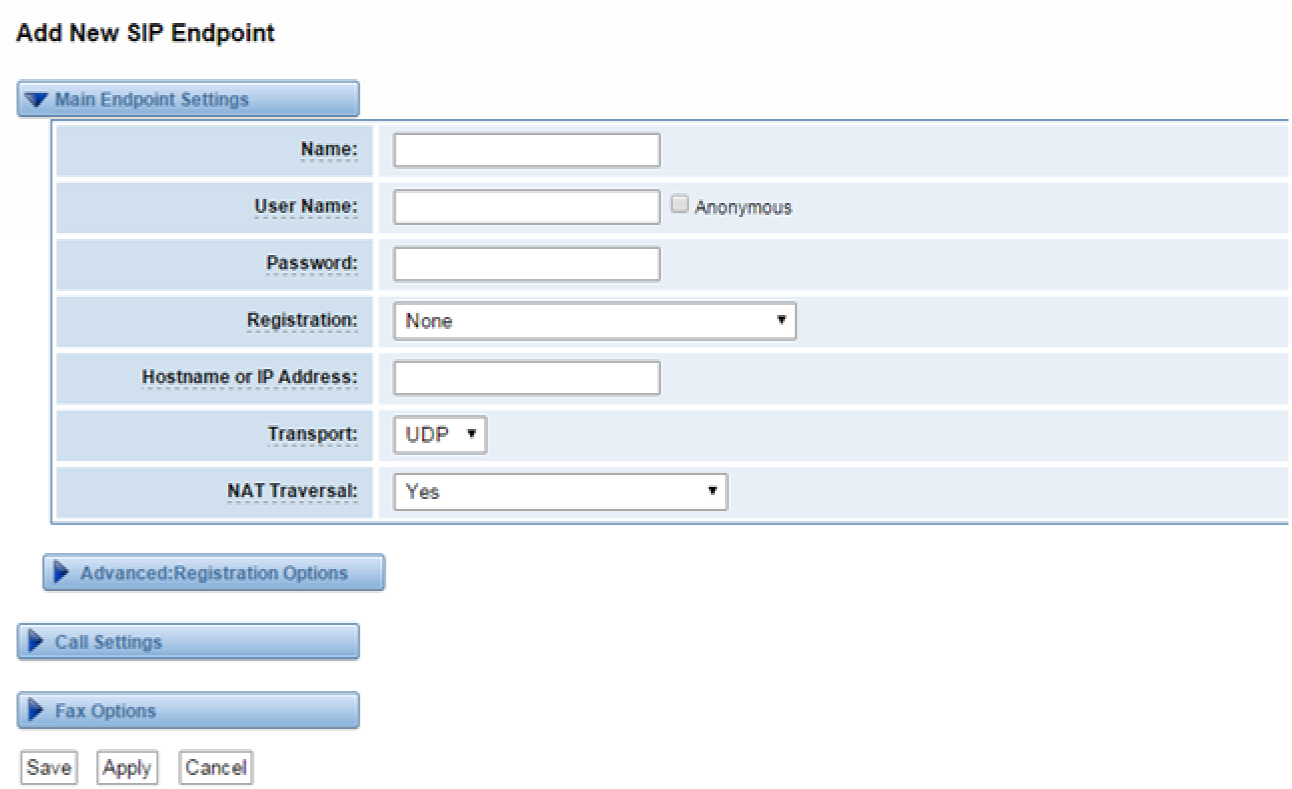
You can click button to add a new SIP endpoint, and if you want to modify existed endpoints, you can click button.

There are 3 kinds of registration types for choose. You can choose Anonymous, Endpoint registers with this gateway or This gateway registers with the endpoint.

You can configure as follows:

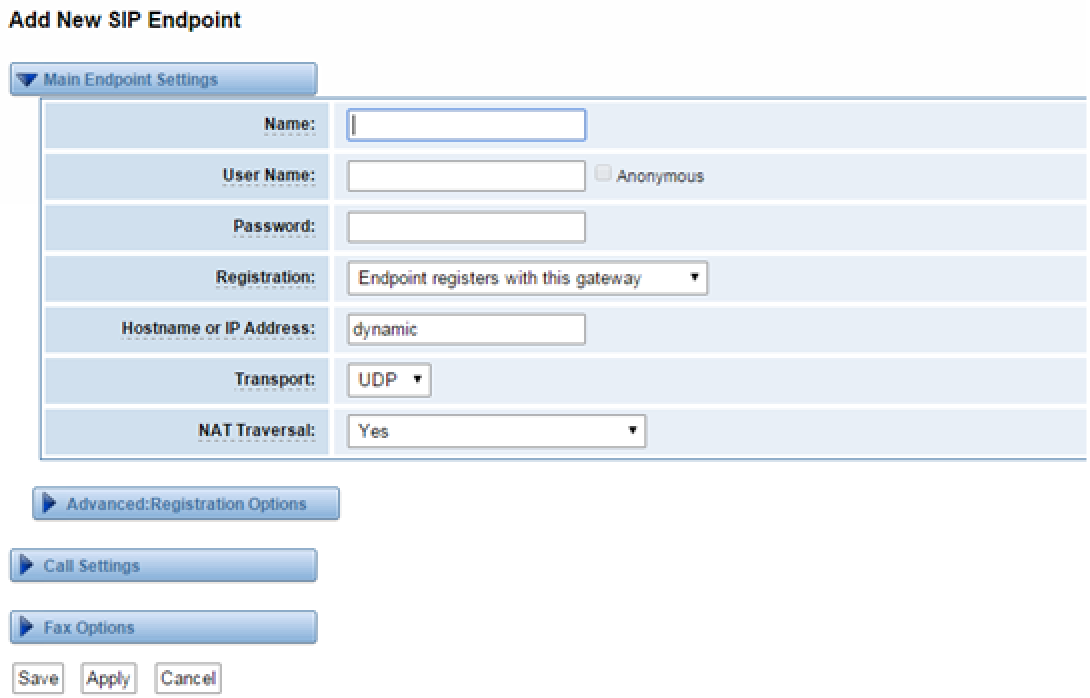
If you set up a SIP endpoint by registration “None” to a server, then you can’t register other SIP endpoints to this server. (If you add other SIP endpoints, this will cause Out-band Routes and Trunks confused.)

Figure 4-1-2 None Registration



For convenience, we have designed a method that you can register your SIP endpoint to your gateway, thus your gateway just work as a server.

Figure 4-1-3 Endpoint Register with Gateway



Also you can choose registration by “This gateway registers with the endpoint”, it’s the same with “None”, except name and password.

Figure 4-1-4 This Gateway Register with the Endpoint

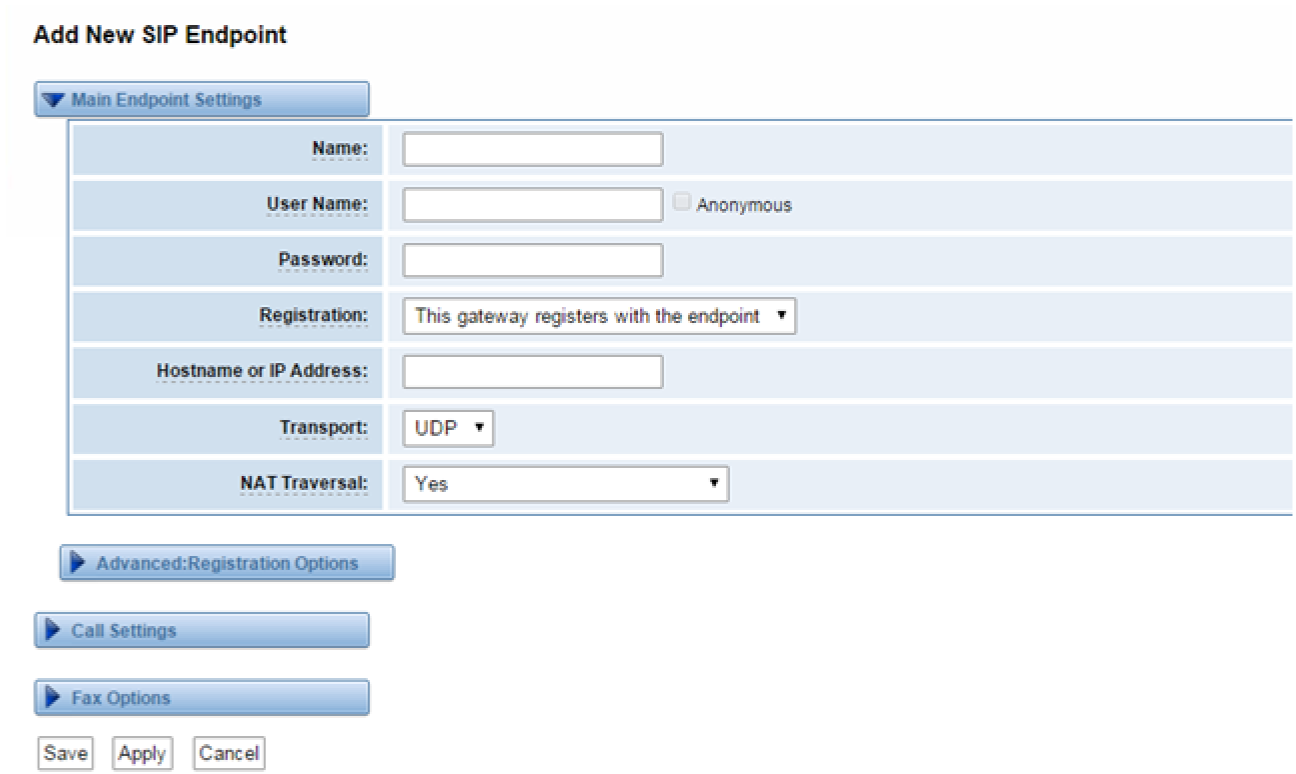


Table 4-1-1 Definition of SIP Options

Options	Definition
Name	A name which is able to read by human. And it’s only used for user’s reference.

Username	User name the end point use to authenticate with the gateway
Password	Password the endpoint will use to authenticate with the gateway. Allowed characters
Registration	Whether this endpoint will registers with this gateway.
Hostname or IP Address	IP address or hostname of the endpoint or 'dynamic' if the endpoint has a dynamic IP address. This will require registration. Notice: if the input here is hostname and your DNS has changed, you must reboot asterisk.
Transport	This sets the possible transport types for outgoing. Order of usage, when the respective transport protocols are enabled, is UDP, TCP, TLS. The first enabled transport type is only used for outbound messages until a Registration takes place. During the peer Registration the transport type may change to another supported type if the peer requests so.
NAT Traversal	Addresses NAT-related issues in incoming SIP or media sessions.

Advanced: Registration Options

Table 4-1-2 Definition of Registration Options

Options	Definition
Authentication User	A username to use only for registration.
Register Extension	When Gateway registers as a SIP user agent to a SIP proxy (provider), calls from this provider connect to this local extension.
From User	A username to identify the gateway to this endpoint.
From Domain	A domain to identify the gateway to this endpoint.
Remote Secret	A password which is only used if the gateway registers to the remote side.
Port	The port number the gateway will connect to at this endpoint.
Qualify	Whether or not to check the endpoint's connection status.
Qualify frequency Frequency	How often, in seconds, to check the endpoint's connection status.
Outbound Proxy	A proxy to which the gateway will send all outbound signaling instead of sending signaling directly to endpoints.

Call Settings

Table 4-1-3 Definition of Call Options

Options	Definition
DTMF Mode	Set default DTMF Mode for sending DTMF. Default: rfc2833. Other options: 'info', SIP INFO message (application/ dtmf-relay); 'Inband', Inband audio (require 64kbit codec - alaw, ulaw).
Trust Remote-Party-ID	Whether or not the Remote-Party-ID header should be trusted.
Send Remote-Party-ID	Whether or not to send the Remote-Party-ID header.
Caller ID Presentation	Whether or not to display Caller ID.

Advanced: Signaling Settings

Table 4-1-4Definition of Signaling Options

Options	Definition
Progress Inband	If we should generate in-band ringing. Always use ‘never’ to never use in-band signalling, Even in cases where some buggy devices might not render it. Valid values: yes, no, never. Default: never.
Append user=phone to URI	Whether or not to add;’ user=phone’ to URIs that contain a valid phone number.
Add Q.850 Reason Headers	Whether or not to add Reason header and to use it if it is available.
Honor SDP Version	By default, the gateway will honor the session version number in SDP packets and will only modify the SDP session if the version number changes. Turn This option off to force the SDP session version number and treat all SDP data as new data. This is require for devices that send non-standard SDP packets (observed with Microsoft OC S).By default This option is on.
Allow Transfers	Whether or not to globally enable transfers. Choosing ‘no’ will disable all transfers (unless enable in peers or users). Default is enabled.
Allow Promiscuous Redirects	Whether or not to allow 302 or REDIR to non-local SIP address .Note that promiscredir when redirects are made to the local system will cause loops since this gateway is incapable of performing a ‘hairpin’ call.
Max Forwards	Setting for the SIP Max-Forwards header (loop prevention).
Send TRYING on REGISTER	Send 100 Trying when the endpoint registers.

Advanced Timer Settings

Table 4-1-5 Definition of Timer Options

Options	Definition
Default T1 Timer	This timer is used primarily in INVITE transactions. The default for Timer T1 is 500ms or the measured run-trip time between the gateway and the device if you have qualify=yes for the device.
Call Setup Timer	If a provisional response is not received in this amount of time, the call will auto-congest. Defaults to 64 times the default T1 timer.
Session Timers	Session-Timers feature operates in the following three modes: originate, Request and run session-timers always; accept, run session-timers only when requested by other UA; refuse, do not run session timers in any case.
Minimum Session	Minimum session refresh interval in seconds. Default is 90secs.
Maximum Session Refresh Interval	Maximum session refresh interval in seconds. Defaults to 1800s.
Session Refresher	The session refresher, uac or uas. Defaults to uas.

Table 4-1-6 Definition of Fax Options

Options	Definition
Mode	Working mode T.38 and T.30
Enabled	Enabled
Error Correction	Error Correction
Max Datagram	In some cases,T.38 endpoints will provide a T38FaxMxDatagram value (during T.38 setup) that is based on an incorrect interpretation of the T.38 recommendation, and result in failures because Asterisk does not believe it can send T.38 packets of a reasonable size to that endpoint (Cisco media gateway are one example of this situation).In these cases, during a T.38 call you will see warring messages on The console/in the logs from the Asterisk UDPTL stack complaining about lack of buffer space to send T.38FaxMaxDatagram value specified by the other end[point, and use a configured value instead.
Fax Detect	FAX detection will cause the SIP channel to jump to the 'faX' extension (if exists) based one or more events being detected. The events that can be detected are an incoming CNG tone or an incoming T.38 re-INVITE request.
Fax Activity	activate T38 fax gateway with 'timeout' seconds
Fax Timeout	activate T38 fax gateway with 'timeout' seconds

Advanced SIP Settings

Networking

Table 4-2-1 Definition of Networking Options

Options	Definition
UDP Bind Port	Choose a port on which to listen for UDP traffic.
Enable TCP	Enable server for incoming TCP connection (default is no).
TCP Bind Port	Choose a port on which to listen for TCP traffic.
TCP Authentication Timeout	The maximum number of seconds a client has to authenticate. If the client does not authenticate before this timeout expires, the client will be disconnected.(default value is: 30 seconds).
TCP Authentication Limit	The maximum number of unauthenticated sessions that will be allowed to connect at any given time (default is: 50).
Enable Hostname Lookup	Enable DNS SRV lookups on outbound calls Note: the gateway only uses the first host in SRV records Disabling DNS SRV lookups disables the ability to place SIP calls based on domain names to some other SIP users on the Internet specifying a port in a SIP peer definition or when dialing outbound calls with suppress SRV lookups for that peer or call.
Enable Internal SIP Call	Whether enable the internal SIP calls or not when you select the registration option "Endpoint registers with this gateway".

Internal SIP Call Prefix	Specify a prefix before routing the internal calls.

NAT Settings

Table 4-2-2 Definition of NAT Settings Options

Options	Definition
Local Network	Format:192.168.0.0/255.255.0.0 or 172.16.0.0./12. A list of IP address or IP ranges which are located inside a NA Ted network. This gateway will replace the internal IP address in SIP and SDP messages with the external IP address when a NAT exists between the gateway and other endpoints.
Local Network List	Local IP address list that you added.
Subscribe Network Change Event	Through the use of the test_stun_monitor module, the gateway has the ability to detect when the perceived external network address has changed. When the stun_ monitor is installed and configured, chan_sip will renew all outbound registrations when the monitor detects any sort of network change has occurred. By default this option is enabled, but only takes effect once res_stun_monitor is configured. If res_stun_monitor is enabled and you wish to not generate all outbound registrations on a network change, use the option below to disable this feature.
Match External Address Locally	Only substitute the exeternaddr or externhost setting if it matches
Dynamic Exclude Static	Disallow all dynamic hosts from registering as any IP address used for staticly defined hosts .This helps avoid the configuration error of allowing your users to register at the same address as a SIP provide.
Externally Mapped TCP Port	The externally mapped TCP port, when the gateway is behind a static NAT or PAI
External Address	The external address (and optional TCP port) of the NAT. External address=hostname [:port] specifies a static address[:port] to be used in SIP and SDP messages. Examples: External address=12.34.56.78 External address=12.34.56.78.9900
External Hostname	The external hostname (and optional TCP port) of the NAT. External Hostname=hostname[:port] is similar to “External address”. Examples: External Hostname=foo.dyndns.net
Hostname Refresh Interval	How often to perform a hostname lookup. This can be useful when your NAT device lets you choose the port mapping, but the IP address is dynamic. Beware, you might suffer from service disruption when the name server resolution fails.

RTP Settings

Table 4-2-3 Definition of RTP Settings Options

Options	Definition
Start of RTP Port Range	Start of range of port numbers to be used for RTP.
End of RTP port Range	End of range of port numbers to be used for RTP

Parsing and Compatibility

Table 4-2-4 Instruction of Parsing and Compatibility

Options	Definition
Strict RFC Interpretation	Check header tags, character conversion in URIs, and multiline headers for strict SIP compatibility(default is yes)
Send Compact Headers	Send compact SIP headers
SDP Owner	Allows you to change the username filed in the SDP owner string. This filed MUST NOT contain spaces.
Disallowed SIP Methods	When a dialog is started with another SIP endpoint, the other endpoint should include an Allow header telling us what SIP methods the endpoint implements. However, some endpoint either do not include an Allow header or lie about what methods they implement. In the former case, the gateway makes the assumption that the endpoint support all known SIP methods. If you know that your SIP endpoint does not provide support for a specific method, then you may provide a list of methods that your endpoint does not implement in the disallowed_ methods option. Note that if your endpoint is truthful with its Allow header, then there is need to set this option.
Shrink Caller ID	The shrinkcallerid function removes '(', ' ', ')', non-trailing '.', and '-' not in square brackets. For example, the caller id value 555.5555 becomes 5555555 when this option is enabled. Disabling this option results in no modification of the caller id value, which is necessary when the caller id represents something that must be preserved. By default this option is on.
Maximum Registration Expiry	Maximum allowed time of incoming registrations and subscriptions (seconds).
Minimum Registration Expiry	Minimum length of registrations/subscriptions (default 60).
Default Registration Expiry	Default length of incoming/outgoing registration.
Registration Timeout	How often, in seconds, to retry registration calls. Default 20 seconds.

Number of Registration	Number of registration attempts before we give up.0=continue forever, hammering the other server until it accepts the registration. Default is 0 tries, continue forever.

Security

Table 4-2-5 Instruction of Security

Options	Definition
Match Auth Username	If available, match user entry using the 'username' field from the authentication line instead of the 'from' field.
Realm	Realm for digest authentication. Realms MUST be globally unique according to RFC 3261. Set this to your host name or domain name.
Use Domain as Realm	Use the domain from the SIP Domains setting as the realm. In this case, the realm will be based on the request 'to' or 'from' header and should match one of the domain. Otherwise, the configured 'realm' value will be used.
Always Auth Reject	When an incoming INVITE or REGISTER is to be rejected, for any reason, always reject with an identical response equivalent to valid username and invalid password/hash instead of letting the requester know whether there was a matching user or peer for their request. This reduces the ability of an attacker to scan for valid SIP usernames. This option is set to 'yes' by default.
Authenticate Options Requests	Enabling this option will authenticate OPTIONS requests just like INVITE requests are. By default this option is disabled.
Allow Guest Calling	Allow or reject guest calls (default is yes, to allow). If your gateway is connected to the Internet and you allow guest calls, you want to check which services you offer everyone out there, by enabling them in the default context.

Media

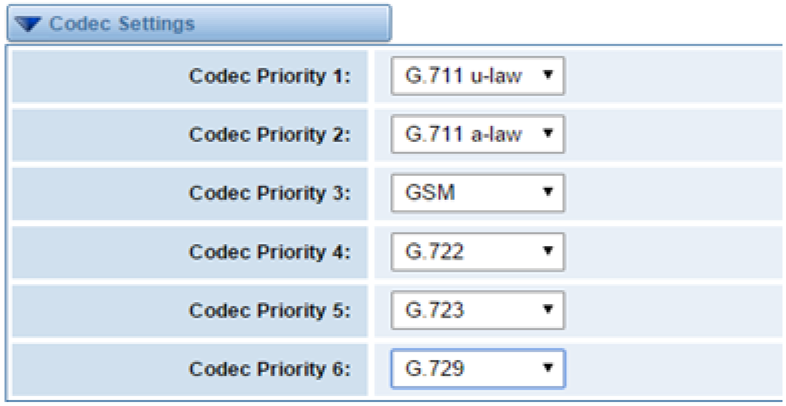
Table 4-2-6 Instruction of Media

Options	Definition
TOS for SIP Packets	Sets type of service for SIP packets
TOS for RTP Packets	Sets type of service for RTP packets

Codec Settings

Select codecs from the list below.

Figure 4-2-1 Codec Settings



Advanced IAX2 Settings

Table 4-3-1 Instruction of General

Options	Definition
Bind Port	Bind port and bindaddr may be specified
Bind Address	More than once to bind to multiple addresses, but the first will be the default.
Enable IAXCompat	Cause Asterisk to spawn a separate thread when it receive a Dialplan Request instead of blocking while for a response.
Enable No checksums	Enable No checksums.
Enable Delay Reject	You may specify a default AMA flag for iaxtel calls. It must be one of ‘default’, ‘omit’, ‘billing’, or ‘documentation’. These flags are used in the generation of call detail records.
ADSI	ADSI (Analog Display Services Interface) can be enable if you have (or may have) ADSI compatible CPE equipment.
SRV Loopup	Whether or not to perform an SRV lookup on outbound calls
AMA Flags	You may specify a global default AMA flag for iaxtel calls. These flags are used in the generation of call detail records.
autokill	If we don’t get ACK to our NEW within 2000ms,and autokill is set to yes, then we cancel the whole thing(that’s enough time for one retransmission only).This is used to keep things from stalling for a long time for a host that is not available for bad connections.
Language	You may specify a global default language for users. This can be specified also on a per-user basis. If omitted, will fallback to English(en)
Account Code	You may specify a default account for Call Detail Records (CDRs) in addition specifying on a per-user basis.

Table 4-3-2 Instruction of Music on Hold

Options	Definition
Mohsuggest	The ‘Mohsuggest’ option specifies which music on hold class to suggest to the peer channel when this channel place the peer on hold. It may be specified globally or on a per-user or per-peer basis.

Mohinterpret	You may specify a global default language for users. This can be specified also on a per-user basis. If omitted, will fallback to English(en)

Table 4-3-3 Instruction of Codec Settings

Options	Definition
Band Width	Specify bandwith of low, medium, or high to control which codes are used in general
Disallow	Fine tune codes here using “allow” and “disallow” clause with specific codes
Allow	Fine tune codes here using “allow” and “disallow” clause with specific codes
Codec Priority	Codec priority controls the codec negotiation of an inbound IAX2 call. This option is inherited to all user entity separately which will override the setting in general.

Table 4-3-4 Instruction of Jitter Buffer

Options	Definition
Jitter Buffer	Global default as to whether you want the jitter buffer at all
Force Jitter Buffer	In the ideal world, when we bridge VoIP channels we don’t want to jitter buffering on the switch, since the endpoints can each handle this. However, some endpoints may have poor jitter buffers themselves, so this option will force to always jitter buffer, even in this case.
Max Jitter Buffers	A maximum size for the jitter buffer
Resyncthreshold	When the jitter buffer notice a significant change in delay that continue over a few frames, it will resync, assuming that the change in delay was caused by a timestamping mix-up. The threshold for noticing a change in delay is measured as twice the measured jitter plus this resync threshold.
Max Jitter Interps	The maximum number of interpolation frames the jitter buffer should return in a row. Since some clients do not send CNG/DTX frames to indicate silence, the jitter buffer will assume silence has begun after returning this many interpolations. This prevents interpolating throughout a long silence.
Jitter Target Extra	Number of milliseconds by which the new jitter buffer will pad its size. The default is 40, so without modification, the new jitter buffer will set its size to the jitter value may help if your network normally has low jitter, but occasionally has spikes.

Table 4-3-5 Instruction of Misc Settings

Options	Definition
IAX Thread Count	Establishes the number of iax helper thread to handle I/O
IAX Max Thread Count	Establishes the number of extra dynamic threads that may by spawned to handle I/O
Max Call Number	limits the amount of call numbers allowed for each individual remote IP address.
MaxCallNumbers_Nonvalidated	used to set the combined number of call numbers that can be allocated for connections where call token validation has been disabled.

Table 4-3-6 Instruction of Quality of Service

Options	Definition
Tos	Type of service
Cos	Class of service

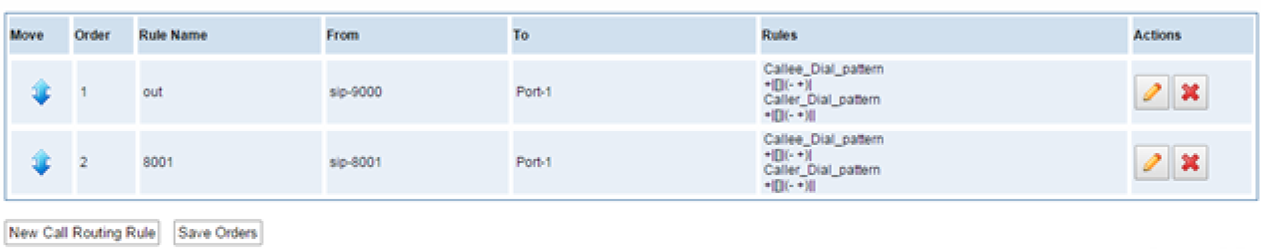
Advanced Fax Settings

Table 4-4-1 Instruction of Quality of Fax Settings

Options	Definition
UDPTL Start	DPTL start configure addresses
UDPTL End	DPTL end configure addresses
UDPTL Checksums	Whether to enable or disable UDP checksums on UDPTL traffic
UDPTL FEC Entries	The number of error correction entries in a UDPTL packet
UDPTL FEC Span	The span over which parity is calculated for FEC in a UDPTL packet
Use Even Ports	Some VoIP providers will only accept an offer with an even-numbered UDPTL port. Set this option so that Asterisk will only attempt to use even-numbered ports when negotiating T.38. Default is no.
Maximum Transmission Rate	Maximum Transmission Rate
Minimum Transmission Rate	Minimum Transmission Rate
Send Progress/Status events to manager session	Manager events with ‘call’ class permissions will receive events indicating the steps to initiate a fax session. Fax completion events are always sent to manager sessions with ‘call’ class permissions, regardless of the value of this option.
Modem Capabilities	Set this value to modify the default modem options. Defasult:v17,v27,v29
ECM	Enable/disable T.30 ECM(error correction mode) by default

5. Routing

Figure 5-1-1 Routing Rules



You are allowed to set up new routing rule by , and after setting routing rules, move rules’ order by pulling up and down, click button to edit the routing and to delete it. Finally click the button to save what you set. shows current routing rules. Otherwise you can set up unlimited routing rules.

Call Routing Rule

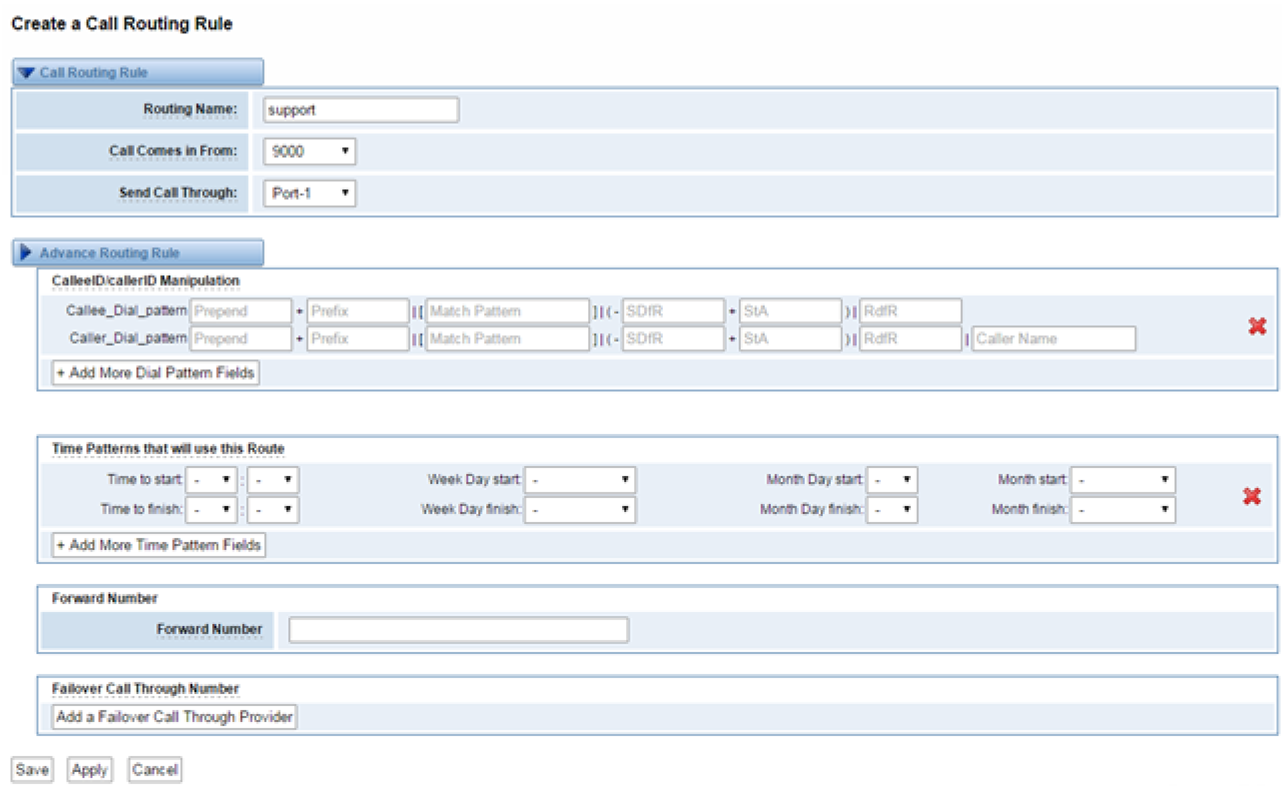
There is an example for Routing rules number conversion, it transform calling, called number at the same time. Suppose you want eleven numbers start at 159 to call the eleven numbers of start at 136. Calling transform delete the three numbers from left, then writing number 086 as prefix, delete the last four numbers, and then add number 0755 at the end, it will show caller name is OpenVox. Called transform adds 086 as prefix , and Change the last two number to 88.

Figure 5-1-2

processing rules	prepend	prefix	Match pattern	SdfR	StA	RdfR	Caller Name
Calling Transformation	086	159	××××××××	4	0755		OpenVox
Called transformation	086	136	××××××××	2	88		N/A

You can click button to set up your routings.

Figure 5-1-3 Example of Set Up Routing Rule



The figure above realizes that calls from “support” SIP endpoint switch you have registered will be transferred to Port-1. When “Call Comes in From” is 9000, “prepend”, “prefix” and “match pattern” in “Advanced Routing Rule” are ineffective, and just “CallerID” option is available.

Table 5-1-1 Definition of Routing Options

Options	Definition
Routing Name	The name of this route. Should be used to describe what types of calls this route matches (for example, 'SIP2GSM' or 'GSM2SIP').
Call Comes in From	The launching point of incoming calls.
Send call Through	The destination to receive the incoming calls.

Table 5-1-2 Description of Advanced Routing Rule

Options	Definition

Dial Patterns that will use this Route	A Dial Pattern is a unique set of digits that will select this route and send the call to the designated trunks. If a dialed pattern matches this route, no subsequent routes will be tried. If Time Groups are enabled, subsequent routes will be checked for matches outside of the designated time(s). Rules: **X** matches any digit from 0-9 **Z** matches any digit from 1-9 **N** matches any digit from 2-9 [1237-9] matches any digit in the brackets (example: 1,2,3,7,8,9) . matches one or more dialed digits: matches one or more dialed digits. **prepend**: Digits to prepend to a successful match. If the dialed number matches the patterns specified by the subsequent columns, then this will be prepended before sending to the trunks. **prefix**: Prefix to remove on a successful match. The dialed number is compared to this and the subsequent columns for a match. Upon a match, this prefix is removed from the dialed number before sending it to the trunks. **match pattern**: The dialed number will be compared against the prefix + this match pattern. Upon a match, the match pattern portion of the dialed number will be sent to the trunks **SDfR**(Stripped Digits from Right): The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. **RDfR**(Reserved Digits from Right) :Designated information to be added to the right end of the current number. **StA**(Suffix to Add):Designated information to be added to the right end of the current number. **Caller Name**: What caller name would you like to set before sending this call to the endpoint. Native language charset is allowable, e.g. Chinese charset, Latin charset.
Forward Number	What destination number will you dial? This is very useful when you have a transfer call.
Failover Call Through Number	The gateway will attempt to send the call out each of these in the order you specify.

You can create various time routes and use these time conditions to limit some specific calls.

Figure 5-1-4 Advance Routing Rule

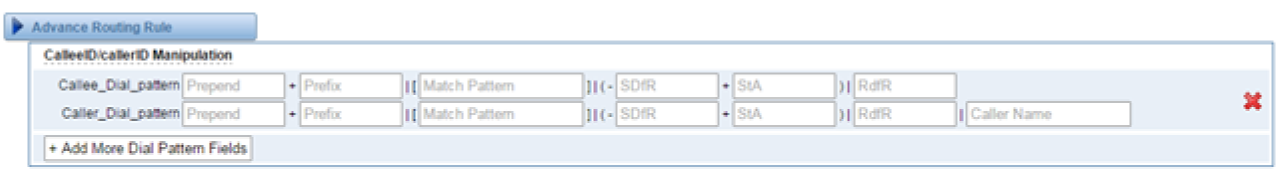
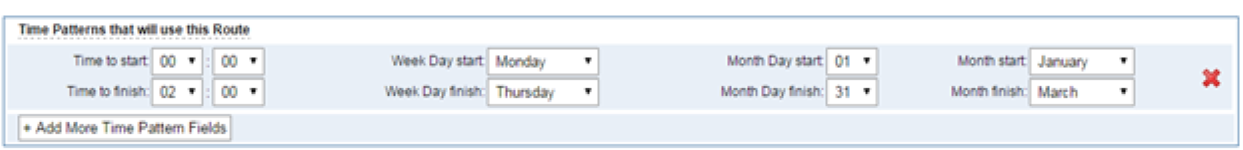
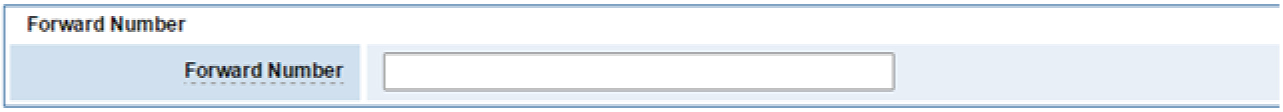


Figure 5-1-5 Time Patterns that will use this Route



If you configure like this, then from January to March, from the first day to the last day of these months, from Monday to Thursday, from 00:00 to 02:00, during this time (meet all above time conditions), all calls will follow this route. And the time will synchronize with your Sever time.

Figure 5-1-6 Change Rules



You can configure forward number when you have a transfer call.

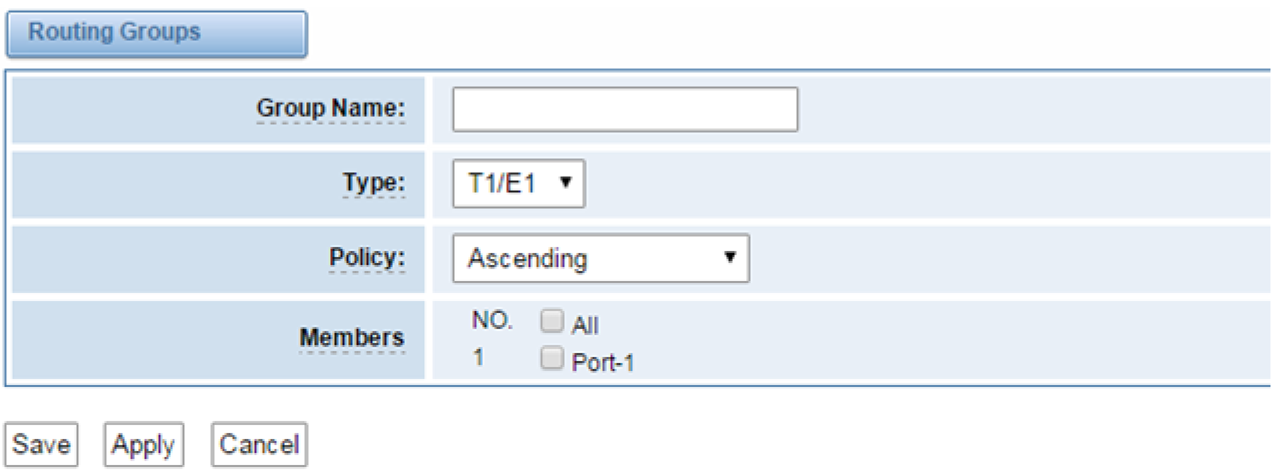
Figure 5-1-7 Failover Call Through Number

You can add one or more “Failover Call Through Numbers”.

Groups

Sometimes you want to make a call through one port, but you don’t know if it is available, so you have to check which port is free. That would be troublesome. But with our product, you don’t need to worry about it. You can combine many Ports or SIP to groups. Then if you want to make a call, it will find available port automatically.

Figure 5-2-1 Establish Group



6. Network

- On “Network” page, there are three sub-pages, “WAN Settings”, “DDNS Settings”, “Toolkit” .

WAN/LAN Settings

There are two types of WAN port IP, Static and DHCP. Static is the default type, and it is 172.16.100.1. The LAN port is a fixed IP and it is 192.168.100.1.

Figure 6-1-1 WAN/LAN Settings Interface

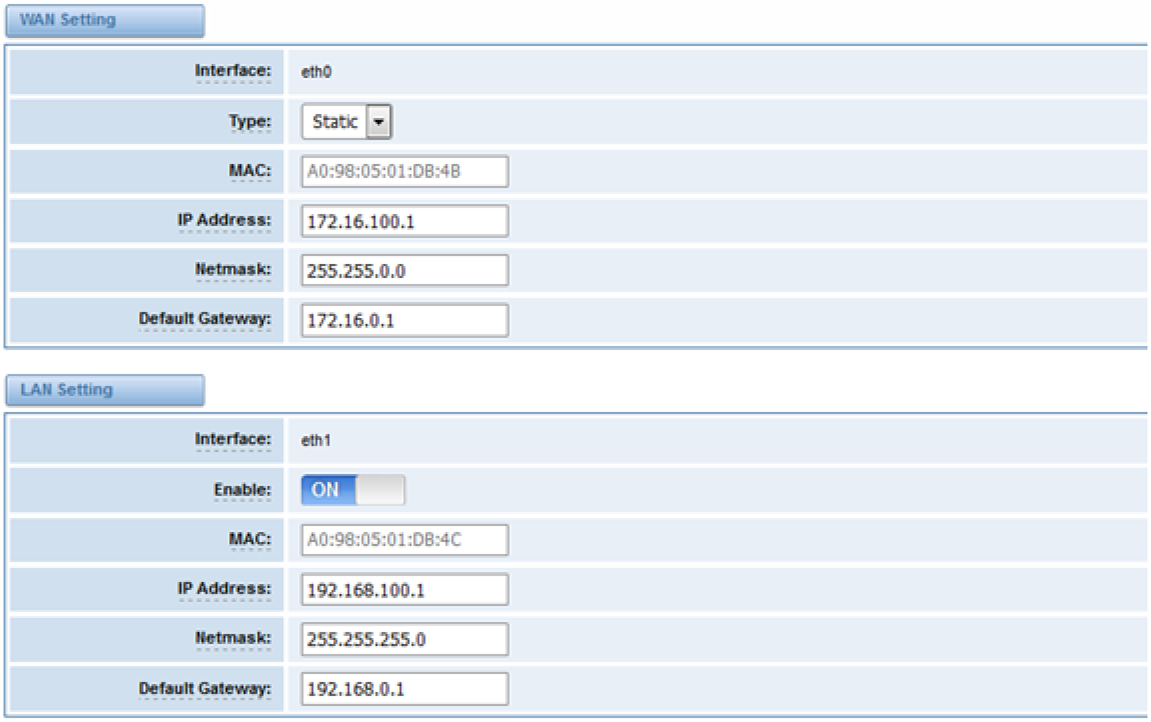
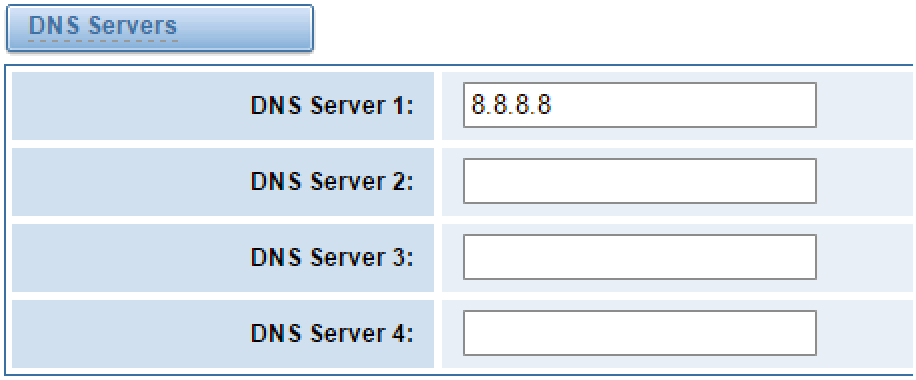


Table 6-1-1Definition of WAN/LAN Settings

Options	Definition
Interface	Specify which interface to capture packets from. 'All' means capture packets from all interfaces.
Type	The method to get IP. Static: manually set up your gateway IP. DHCP: automatically get IP from your local LAN.
MAC	Physical address of your network interface.
Address	The IP address of your gateway.
Network	The subnet mask of your gateway.
Default Gateway	Default getaway IP address.

Basically this info is from your local network service provider, and you can fill in four DNS servers.

Figure 6-1-2 DNS Interface



DNS Servers: A list of DNS IP address. Basically this info is from your local network service provider.

DDNS Settings

You can enable or disable DDNS (dynamic domain name server).

Figure 6-2-1 DDNS Interface

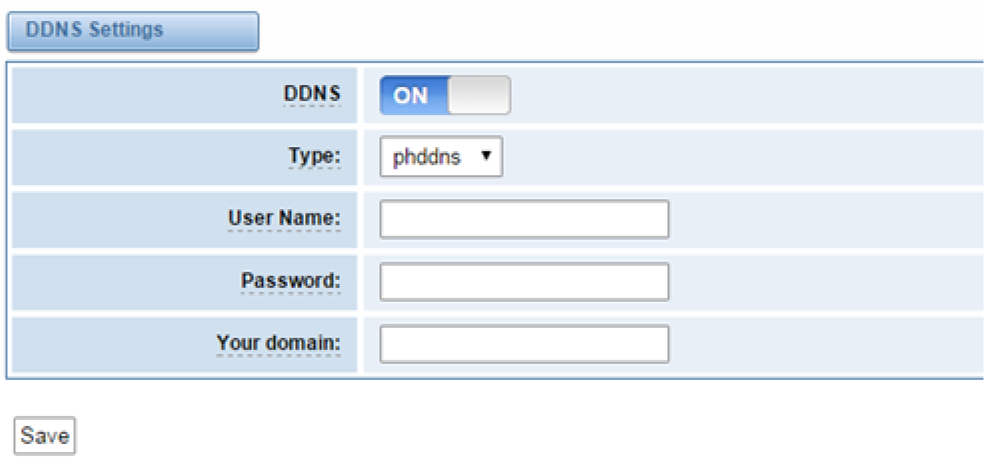


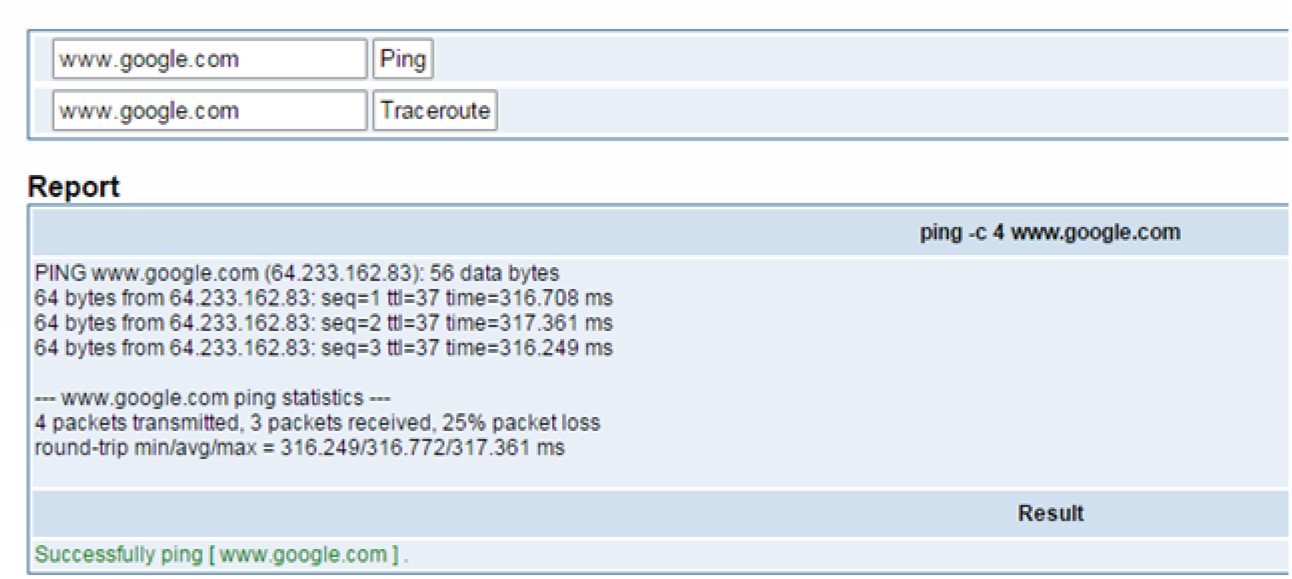
Table 6-2-1 Definition of DDNS Settings

Options	Definition	
DDNS	Enable/Disable DDNS(dynamic domain name server)	
Type	Set the type of DDNS server.	
Username	Your DDNS account’s login name.	
Password	Your DDNS account’s password.	
Your domain	The domain to which your web server will belong.	

Toolkit

It is used to check network connectivity. Support Ping command on web GUI.

Figure 6-3-1 Network Connectivity Checking



7. Advanced

Asterisk API

When you make “Enable” switch to “ON”, this page is available.

Figure 7-1-1 API Interface

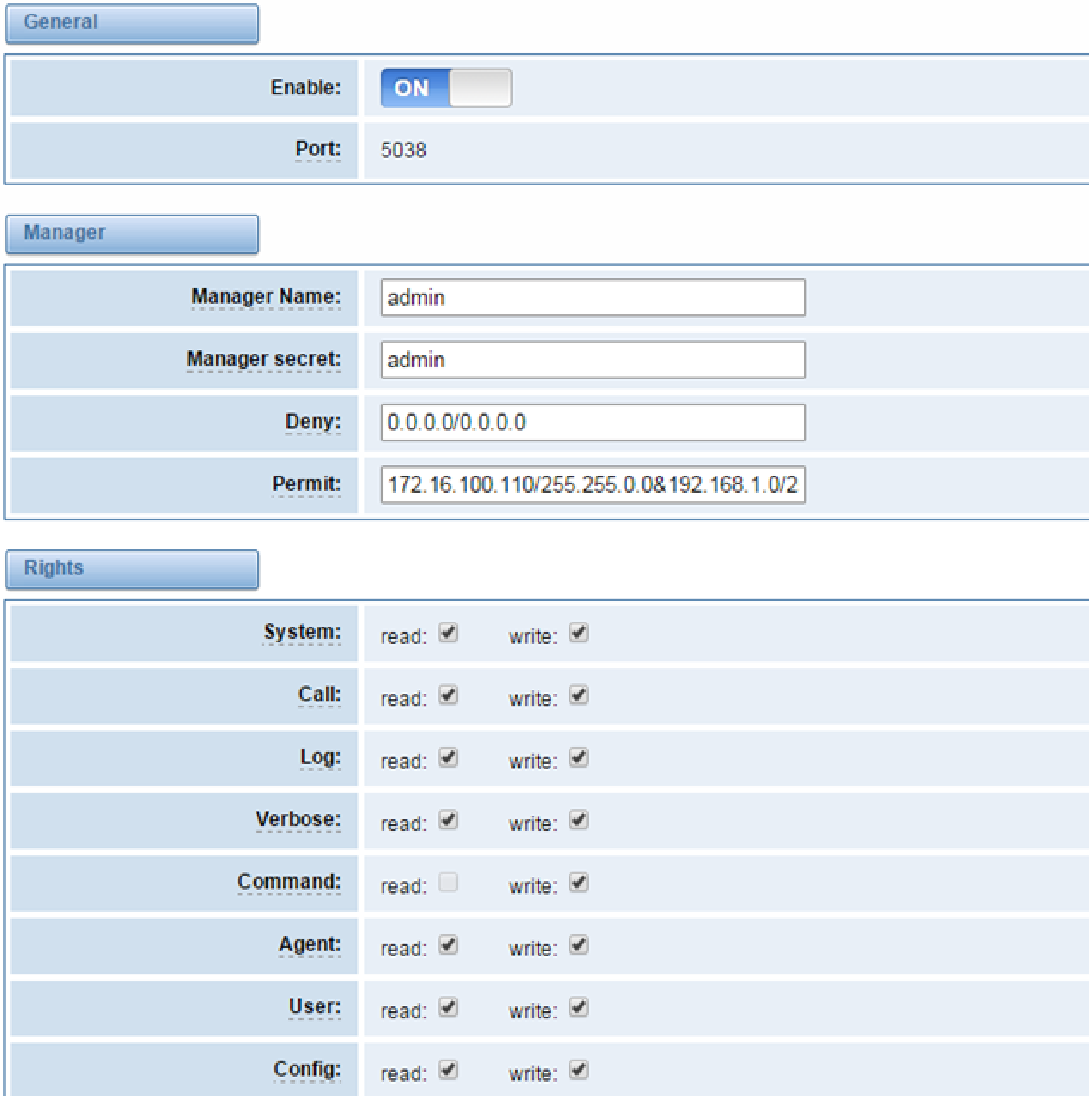


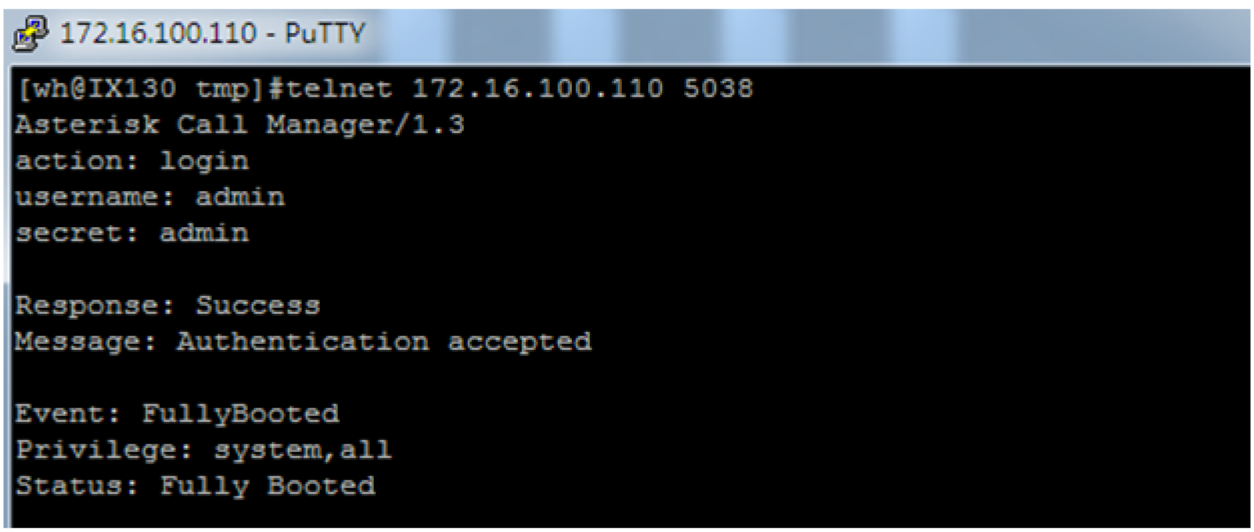
Table 7-1-1 Definition of Asterisk API

Options	Definition	
Port	Network port number	
Manager Name	Name of the manager without space	
Manager secret	Password for the manager. Characters: Allowed characters “-_+.<>&0-9a-zA-Z”. Length:4-32 characters.	
Deny	If you want to deny many hosts or networks, use char & as separator. Example: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0	
Permit	If you want to permit many hosts or network, use char & as separator. Example: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0	

System	General information about the system and ability to run system management commands, such as Shutdown, Restart, and Reload.	
Call	Information about channels and ability to set information in a running channel.	
Log	Logging information. Read-only. (Defined but not yet used.)	
Verbose	Verbose information. Read-only. (Defined but not yet used.)	
Command	Permission to run CLI commands. Write-only.	
Agent	Information about queues and agents and ability to add queue members to a queue.	
User	Permission to send and receive UserEvent.	
Config	Ability to read and write configuration files.	
DTMF	Receive DTMF events. Read-only.	
Reporting	Ability to get information about the system.	
Dialplan	Receive NewExten and Var Set events. Read-only.	
Originate	Permission to originate new calls. Write-only.	
All	Select all or deselect all.	

Once you set like the above figure, the host 172.16.100.110/255.255.0.0 is allowed to access the gateway API. Please refer to the following figure to access the gateway API by putty. 172.16.100.110 is the gateway's IP, and 5038 is its API port.

Figure 7-1-2 Putty Access



Asterisk CLI

In this page, you are allowed to run Asterisk commands.

Figure 7-2-1 Asterisk Command Interface

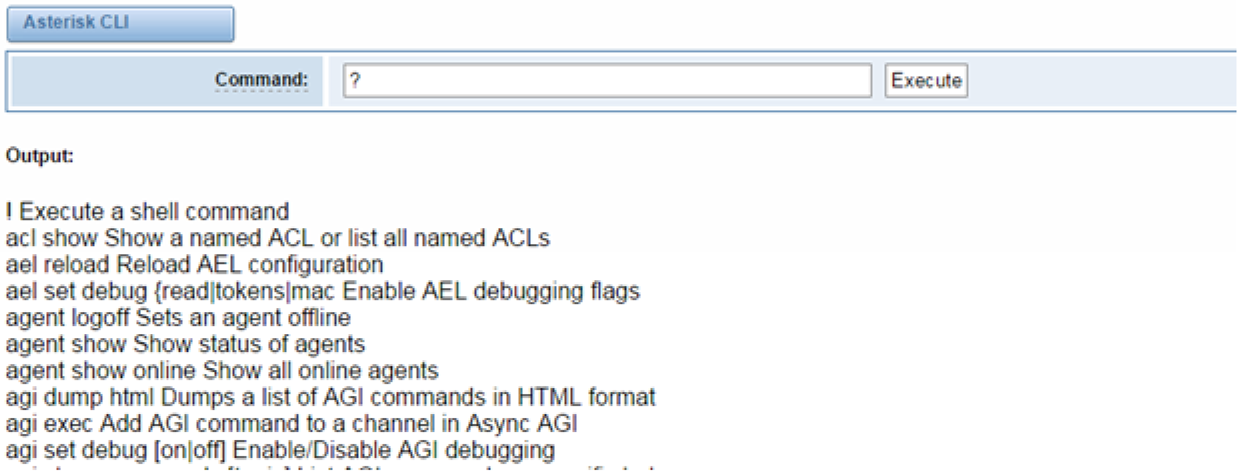


Table 7-2-1 Definition of Asterisk CLI

Options	Definition
Command	Type your Asterisk CLI commands here to check or debug your gateway. e.g, type “help” or “?” you will get all help information.

If you type “help” or “?” and execute it, the page will show you the executable commands.

Asterisk File Editor

On this page, you are allowed to edit and create configuration files.

Click the file to edit.

Figure 7-3-1 Configuration Files List

Prime Config. Files

File Name	File Size
system.conf	92
sip.conf	105
sip_endpoints.conf	1683
logger.conf	4765
extensions.conf	122
sip_general.conf	575
extensions_macro.conf	2298
extensions_routing.conf	1111
dahdi-channels.conf	184
chan_dahdi.conf	871
ss7.conf	391

Config. Files List

File Name	File Size
acl.conf	2817
agents.conf	2531
alarmreceiver.conf	2084
amd.conf	767
asterisk.conf	4237
calendar.conf	5171
ccss.conf	8827
cdr.conf	133
cdr_custom.conf	1617
cdr_manager.conf	418

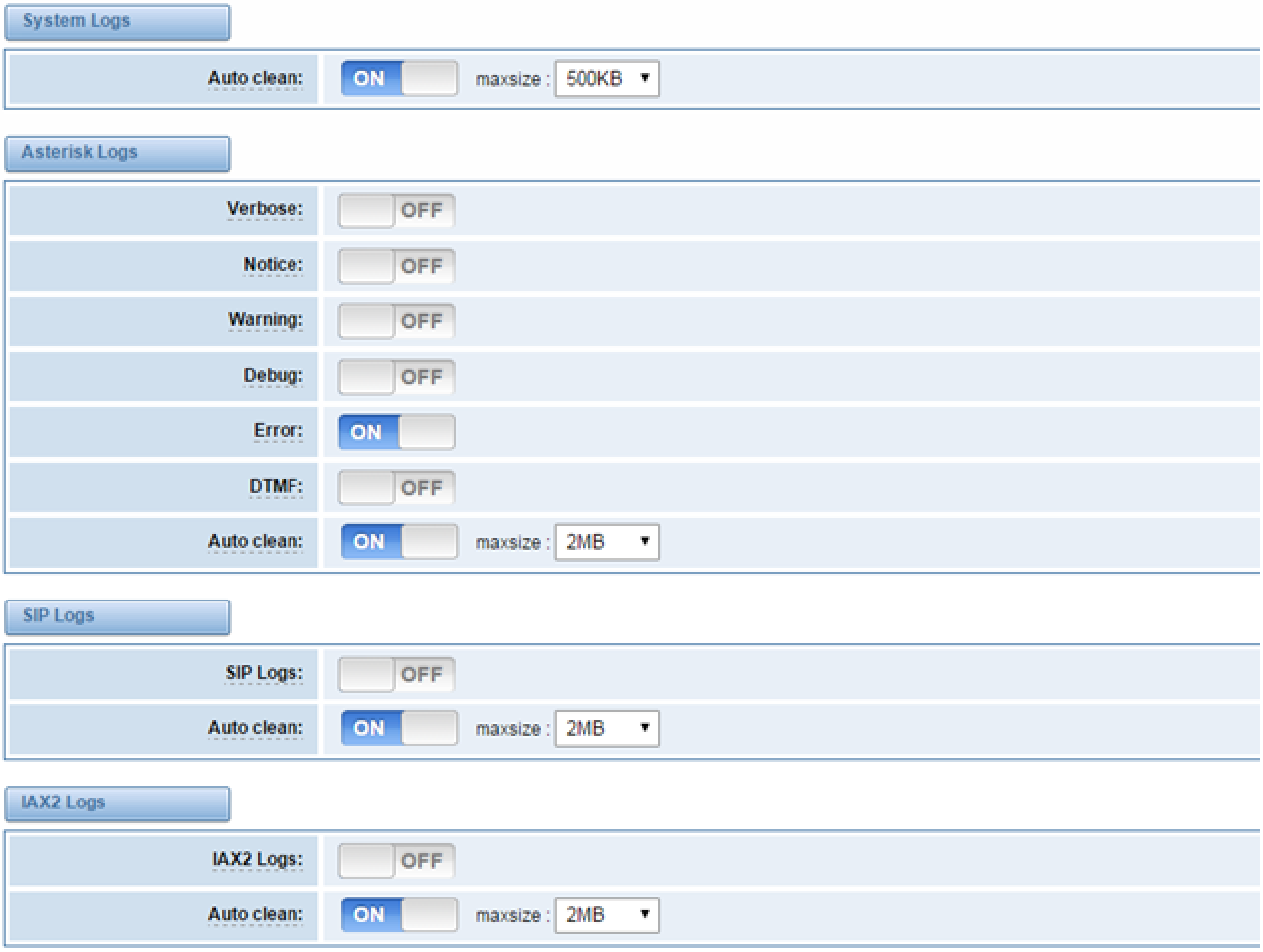
Click “New Configuration File” to create a new configuration file. After editing or creating, please reload Asterisk.

8. Logs

On the “Log Settings” page, you should set the related logs on to scan the responding logs page. For example, set “System Logs” on like the following, then you can turn to “System” page for system logs, otherwise, system logs is unavailable. And the same with other log pages.

Log Settings

Figure 8-1-1 Logs Settings


System Logs
Auto clean:
ON
maxsize :
500KB
Asterisk Logs
Verbose:
OFF
Notice:
OFF
Warning:
OFF
Debug:
OFF
Error:
ON
DTMF:
OFF
Auto clean:
ON
maxsize :
2MB
SIP Logs
SIP Logs:
OFF
Auto clean:
ON
maxsize :
2MB
IAX2 Logs
IAX2 Logs:
OFF
Auto clean:
ON
maxsize :
2MB

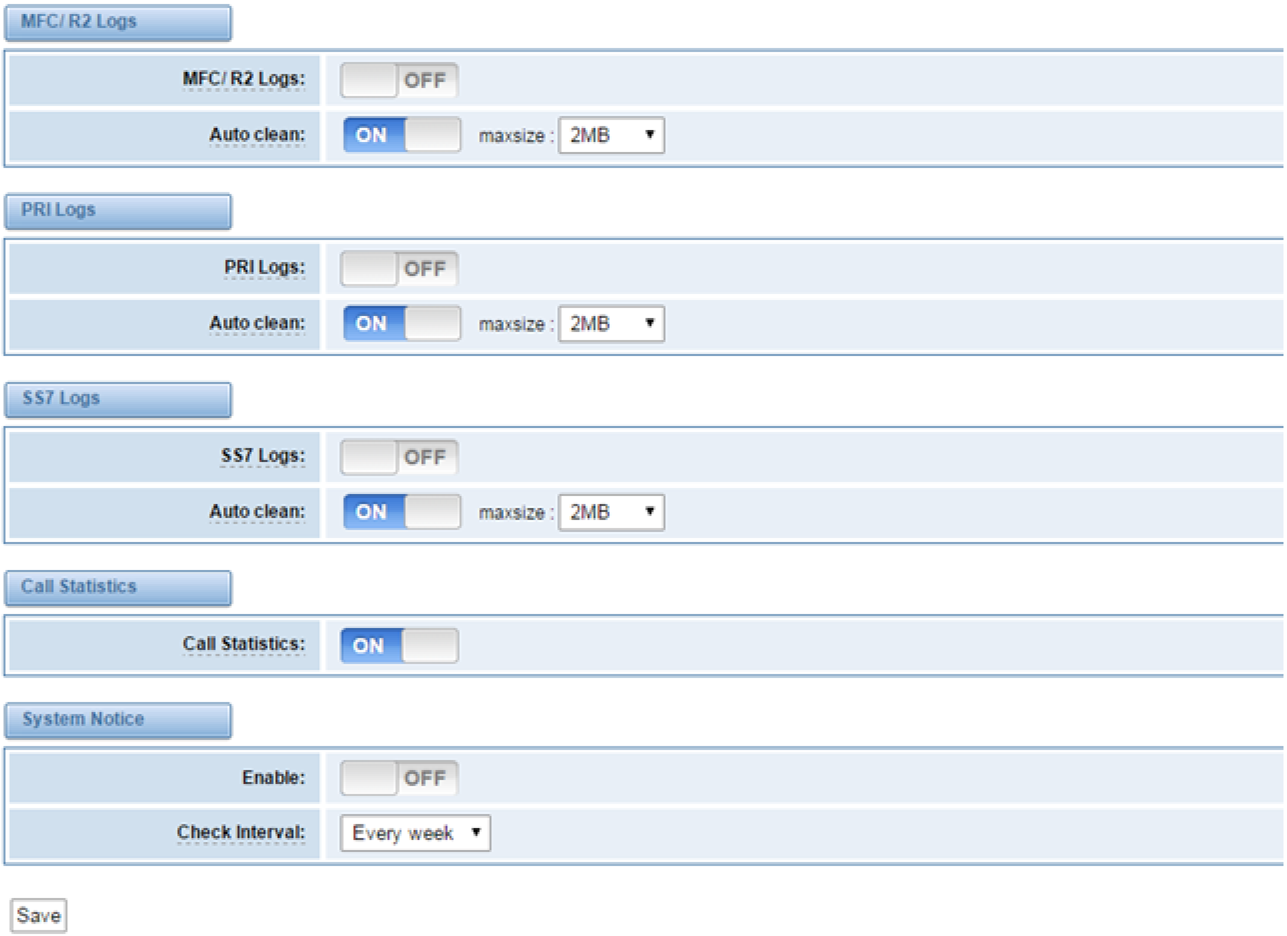


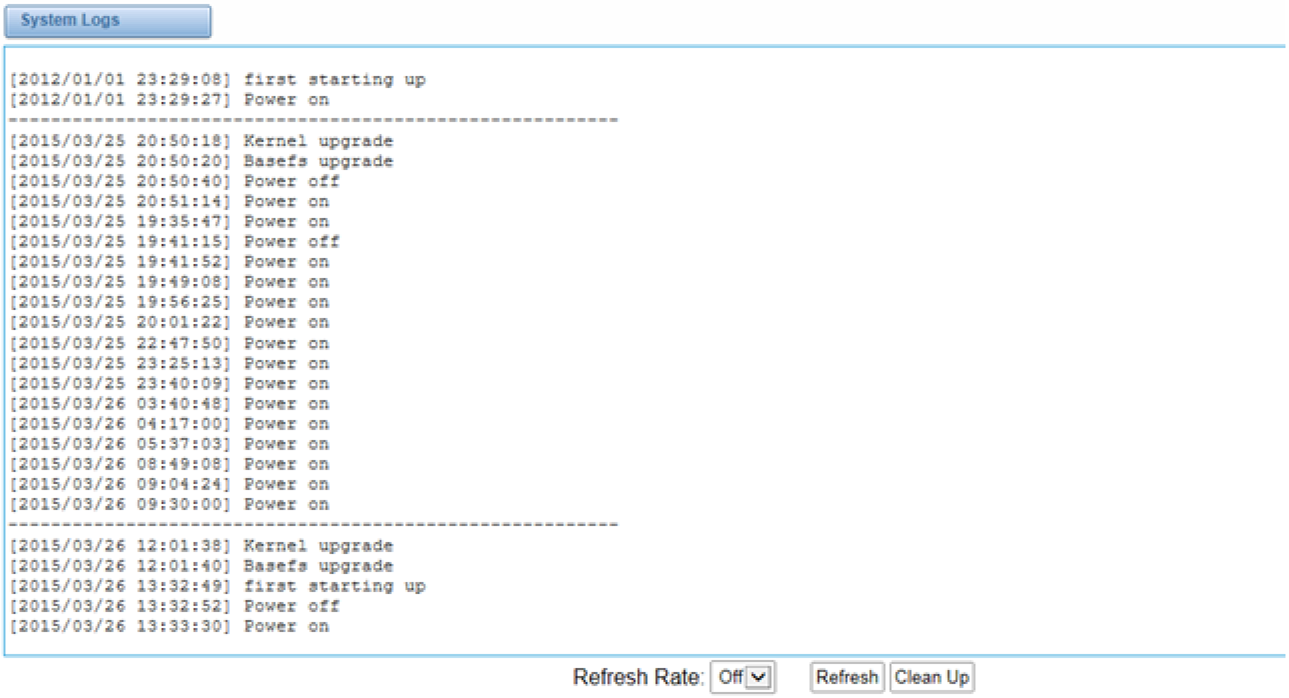
Table 8-1-1 Definition of Logs

Options	Definition
Auto clean: (System Logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **switch off** : logs will remain, and the file size will increase gradually. default on, default size=1MB
Verbose:	Asterisk console verbose message switch.
Notice:	Asterisk console notice message switch.
Warning:	Asterisk console warning message switch.
Debug:	Asterisk console debug message switch.
Error:	Asterisk console error message switch.
DTMF:	Asterisk console DTMF info switch.
Auto clean: (asterisk logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **Switch off**: logs will remain, and the file size will increase gradually. **default on**, default size=100KB

SIP Logs:	Whether enable or disable SIP log.
Auto clean: (SIP logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **Switch off**: logs will remain, and the file size will increase gradually. **default on**, default size=2MB
IAX2 Logs	Whether enable or disable IAX log
Auto clean	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **Switch off**: logs will remain, and the file size will increase gradually. **default on**, default size=2MB
MFC/ R2 Logs	Whether enable or disable MFC/ R2 Logs log.
Auto clean	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **Switch off**: logs will remain, and the file size will increase gradually. **default on**, default size=100KB
PRI Logs	PRI port logs. You can choose one or more ports. If you choose "All", the "PRI" page will show you the logs about all the ports.
Auto clean (PRI logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **Switch off**: logs will remain, and the file size will increase gradually. **default on**, default size=2MB
.SS7 Logs	Whether enable or disable SS7 log
Auto clean	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **Switch off**: logs will remain, and the file size will increase gradually. **default on**, default size=100KB
Call Statistics	Whether enable or disable Call Statistics.

System

Figure 8-2-1System Logs Output



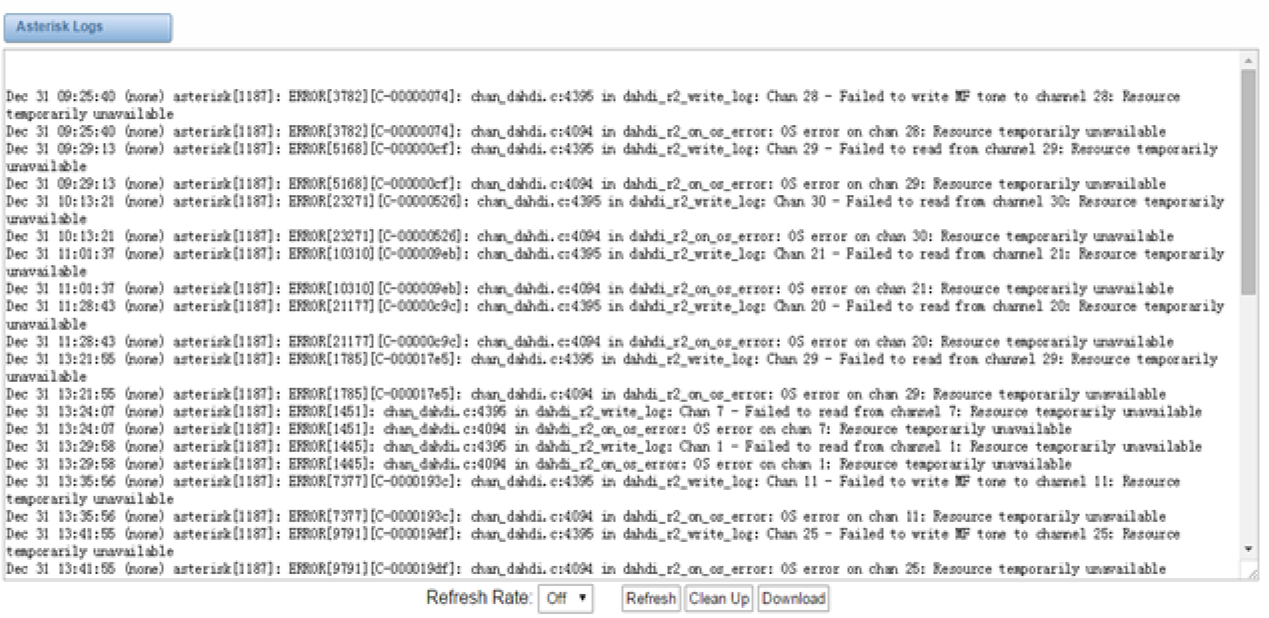
System Logs

```
[2012/01/01 23:29:08] first starting up
[2012/01/01 23:29:27] Power on
---------------------------------------------------------
[2015/03/25 20:50:18] Kernel upgrade
[2015/03/25 20:50:20] Basefs upgrade
[2015/03/25 20:50:40] Power off
[2015/03/25 20:51:14] Power on
[2015/03/25 19:35:47] Power on
[2015/03/25 19:41:15] Power off
[2015/03/25 19:41:52] Power on
[2015/03/25 19:49:08] Power on
[2015/03/25 19:56:25] Power on
[2015/03/25 20:01:22] Power on
[2015/03/25 22:47:50] Power on
[2015/03/25 23:25:13] Power on
[2015/03/25 23:40:09] Power on
[2015/03/26 03:40:48] Power on
[2015/03/26 04:17:00] Power on
[2015/03/26 05:37:03] Power on
[2015/03/26 08:49:08] Power on
[2015/03/26 09:04:24] Power on
[2015/03/26 09:30:00] Power on
---------------------------------------------------------
[2015/03/26 12:01:38] Kernel upgrade
[2015/03/26 12:01:40] Basefs upgrade
[2015/03/26 13:32:49] first starting up
[2015/03/26 13:32:52] Power off
[2015/03/26 13:33:30] Power on
```

Refresh Rate: Off | Refresh | Clean Up

Asterisk

Figure 8-3-1 Asterisk Logs



Asterisk Logs

```
Dec 31 09:25:40 (none) asterisk[1187]: ERROR[3782][C-00000074]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 28 - Failed to write MF tone to channel 28: Resource temporarily unavailable
Dec 31 09:25:40 (none) asterisk[1187]: ERROR[3782][C-00000074]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 28: Resource temporarily unavailable
Dec 31 09:29:13 (none) asterisk[1187]: ERROR[5168][C-000000cf]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 29 - Failed to read from channel 29: Resource temporarily unavailable
Dec 31 09:29:13 (none) asterisk[1187]: ERROR[5168][C-000000cf]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 29: Resource temporarily unavailable
Dec 31 10:13:21 (none) asterisk[1187]: ERROR[23271][C-00000526]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 30 - Failed to read from channel 30: Resource temporarily unavailable
Dec 31 10:13:21 (none) asterisk[1187]: ERROR[23271][C-00000526]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 30: Resource temporarily unavailable
Dec 31 11:01:37 (none) asterisk[1187]: ERROR[10310][C-000009eb]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 21 - Failed to read from channel 21: Resource temporarily unavailable
Dec 31 11:01:37 (none) asterisk[1187]: ERROR[10310][C-000009eb]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 21: Resource temporarily unavailable
Dec 31 11:28:43 (none) asterisk[1187]: ERROR[21177][C-00000c9c]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 20 - Failed to read from channel 20: Resource temporarily unavailable
Dec 31 11:28:43 (none) asterisk[1187]: ERROR[21177][C-00000c9c]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 20: Resource temporarily unavailable
Dec 31 13:21:55 (none) asterisk[1187]: ERROR[1785][C-000017e5]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 29 - Failed to read from channel 29: Resource temporarily unavailable
Dec 31 13:21:55 (none) asterisk[1187]: ERROR[1785][C-000017e5]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 29: Resource temporarily unavailable
Dec 31 13:24:07 (none) asterisk[1187]: ERROR[1451]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 7 - Failed to read from channel 7: Resource temporarily unavailable
Dec 31 13:24:07 (none) asterisk[1187]: ERROR[1451]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 7: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[1187]: ERROR[1445]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 1 - Failed to read from channel 1: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[1187]: ERROR[1445]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 1: Resource temporarily unavailable
Dec 31 13:35:56 (none) asterisk[1187]: ERROR[7377][C-0000193c]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 11 - Failed to write MF tone to channel 11: Resource temporarily unavailable
Dec 31 13:35:56 (none) asterisk[1187]: ERROR[7377][C-0000193c]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 11: Resource temporarily unavailable
Dec 31 13:41:55 (none) asterisk[1187]: ERROR[9791][C-000019df]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 25 - Failed to write MF tone to channel 25: Resource temporarily unavailable
Dec 31 13:41:55 (none) asterisk[1187]: ERROR[9791][C-000019df]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on chan 25: Resource temporarily unavailable
```

Refresh Rate: Off | Refresh | Clean Up | Download

On the pages of “system”, ”Asterisk”, “SIP”, “IAX2”, “SS7”, and “MFC/R2”, there are some functions: Displays the log by port, refresh regularly and log download.

Statistics

Figure 8-9-1 Call Statistics

Statistics

Answered	Congestion	Call Busy	Call Failed	No Answer	Unknown	Current calls	Accumulated Calls	Calls duration	ASR
46031	0	0	113	0	0	0	46144	2761938	99.76%

Refresh | Reset Statistics

The figure of call statistics, you'll find "Answered" "Congestion" "Call Busy" "Call Failed" "No Answer" "Current Calls" "Unknown" "Current calls" "Accumulated Calls" "Calls duration" and "ASR".