DGW-L20X User Manual

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Revision History

Issue version	Issue date	Detail of change
1.0	Jul 12 2019	Initial

Content

1 Overview

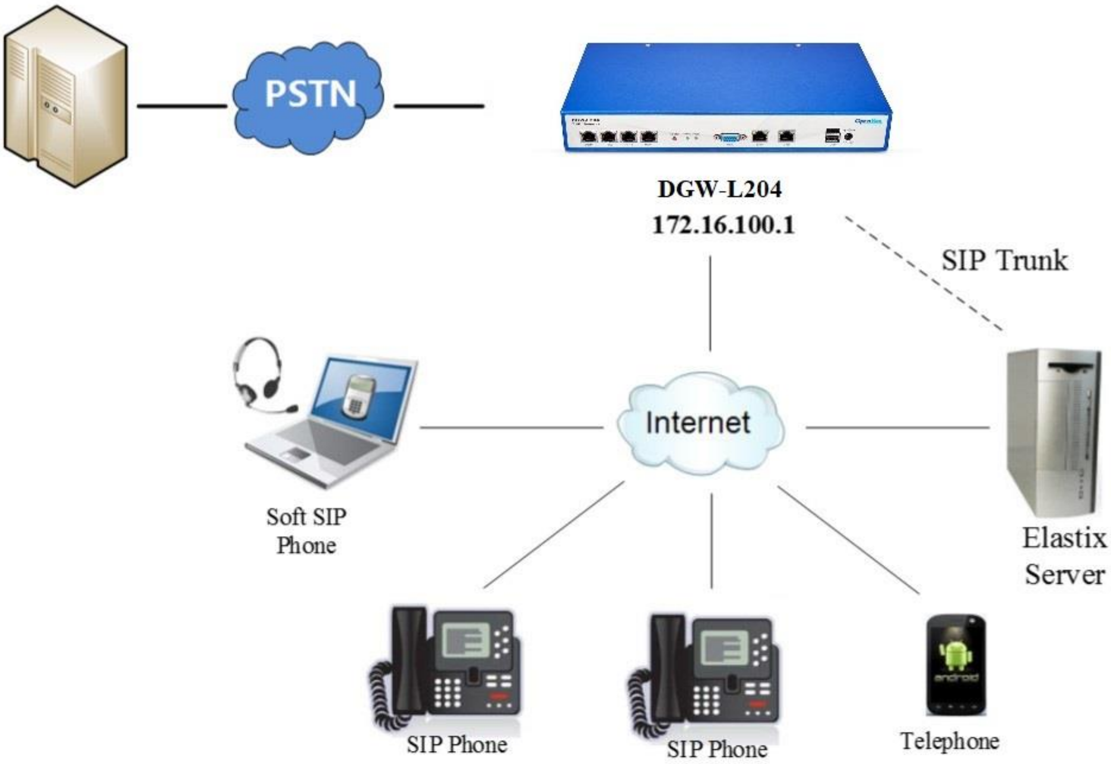
1.1 What is DGW-L20X?

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).

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, 3CX, FreeSWITCH SIP server and VOS VoIP operating platform.

1.2 Sample Application

Figure 1-2-1 Topological Graph



1.3 Product Appearance

The picture below is appearance of DGW-L20X.

Figure 1-3-1 Product Appearance

Figure 1-3-2 Front Panel

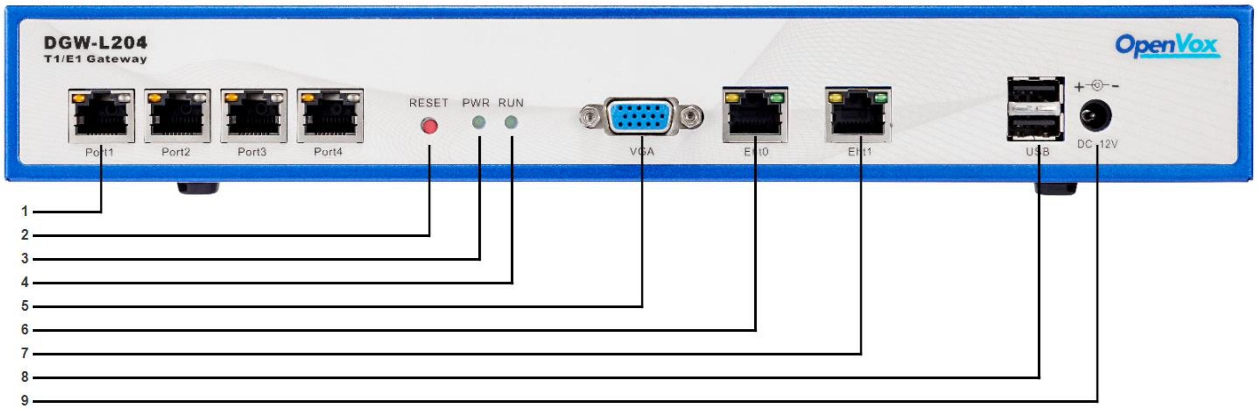


Table 1-3-1 Description of Front Panel

Interface	**Function**	**Color**	**Work Status**
1 Port 1-Port4	E1/T1 ports. 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.
			Fast blinking (Green 0.5s and Flash 0.5s): Work abnormally.
			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.

1.4 Main Features

- Based on Asterisk®
- Editable Asterisk® 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
- Support call status information
- Support T.38/Pass-through fax
- Support Auto Provision, SNMP and TR069
- 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

1.5 Physical Information

Table 1-5-1 Description of Physical Information

Weight	1352g
Size	31cm*16.25cm*4.4cm
Temperature	-40~85°C (Storage)
	0~40°C (Operation)

Operation humidity	5%~95% non-condensing
Max power	18W
LAN port	1
WAN port	1

1.6 Software

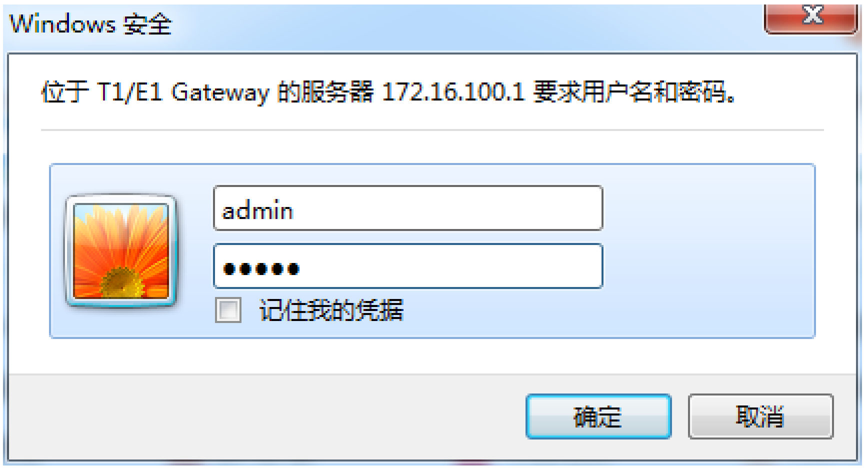
Default IP: 172.16.100.1(WAN), 192.168.100.1(LAN)

Username: admin

Password: admin

Notice: Log in

Figure 1-6-1 LOG IN Interface



2 System

2.1 Status

On the "**System Status**" page, you will find all Interface status, channels status, SIP, IAX2, Routing rules, and Network information.

Figure 2-1-1 System Status

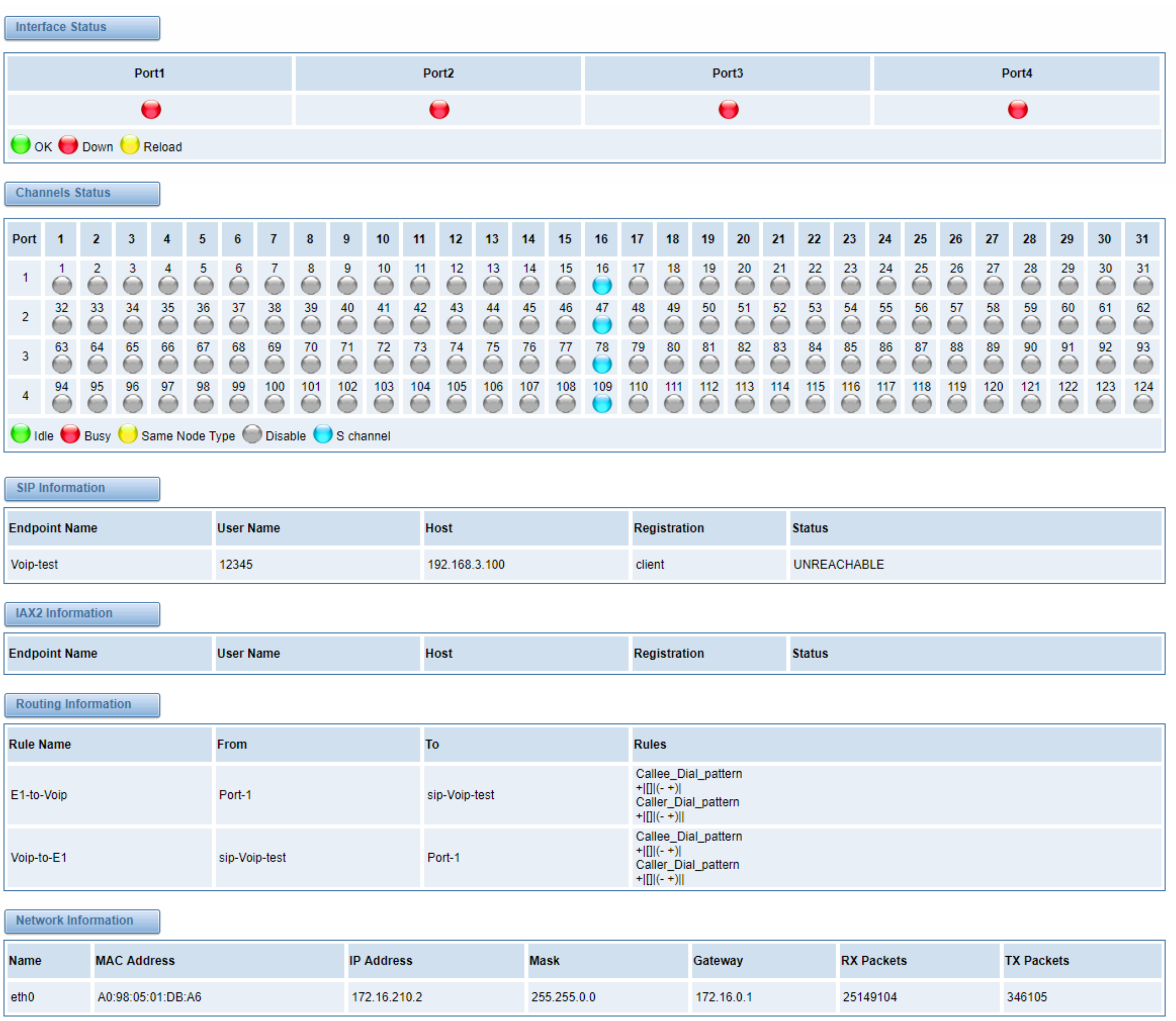


Table 2-1-1 Description of System Status

Options	**Definition**
Interface	Show the status of port, include “OK” and “Down”. “Down” means no trunk.
Channels Status	Show the Channels status of port, include "**Idle**". "**Busy**". "**Disable**" and “**S channel**”. "**Idle**" means it is available; "**Busy**" means the channel is busy.

2.2 Call Status

The verbose of the system call status will be present on the “**Call Status**” page. You can select the specified T1/E1 port which you are care for.

Figure 2-2-1 Verbose of call status

Call Status

Select Port 1

Channel	Status	Direction	CallerID	CalleeID	AnsweredTime	Duration
1	ANSWERED	IP->PSTN	2001	2001	2016-03-10 09:39:10	00: 00: 40
2	ANSWERED	IP->PSTN	2002	2002	2016-03-10 09:39:10	00: 00: 40
3	ANSWERED	IP->PSTN	2003	2003	2016-03-10 09:39:11	00: 00: 39
4	ANSWERED	IP->PSTN	2004	2004	2016-03-10 09:39:11	00: 00: 39
5	ANSWERED	IP->PSTN	2005	2005	2016-03-10 09:39:11	00: 00: 39
6	ANSWERED	IP->PSTN	2006	2006	2016-03-10 09:39:12	00: 00: 38
7	ANSWERED	IP->PSTN	2007	2007	2016-03-10 09:39:12	00: 00: 38
8	ANSWERED	IP->PSTN	2008	2008	2016-03-10 09:39:12	00: 00: 38
9	ANSWERED	IP->PSTN	2009	2009	2016-03-10 09:39:13	00: 00: 37
10	ANSWERED	IP->PSTN	2010	2010	2016-03-10 09:39:13	00: 00: 37
11	ANSWERED	IP->PSTN	2011	2011	2016-03-10 09:39:13	00: 00: 37
12	ANSWERED	IP->PSTN	2012	2012	2016-03-10 09:39:14	00: 00: 36
13	ANSWERED	IP->PSTN	2013	2013	2016-03-10 09:39:14	00: 00: 36
14	ANSWERED	IP->PSTN	2014	2014	2016-03-10 09:39:14	00: 00: 36
15	ANSWERED	IP->PSTN	2015	2015	2016-03-10 09:39:15	00: 00: 35

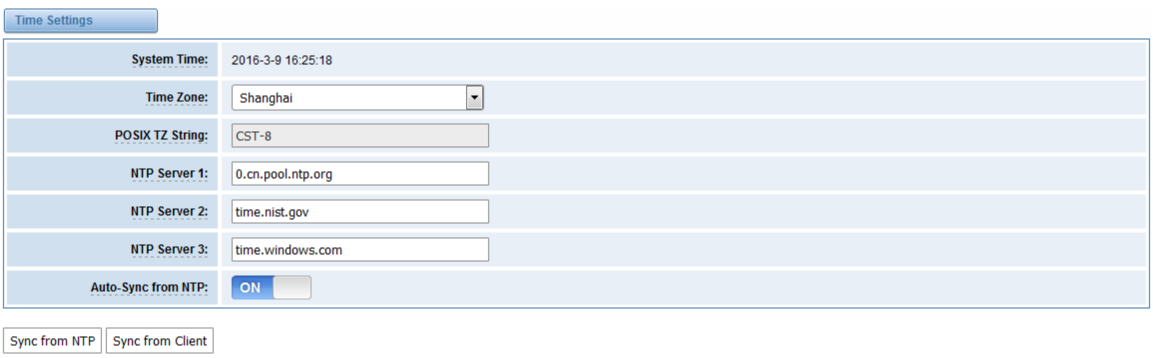
2.3 Time

Table 2-3-1 Description 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, [0.cn.pool.ntp.org].
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 2-3-1 Time Settings



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

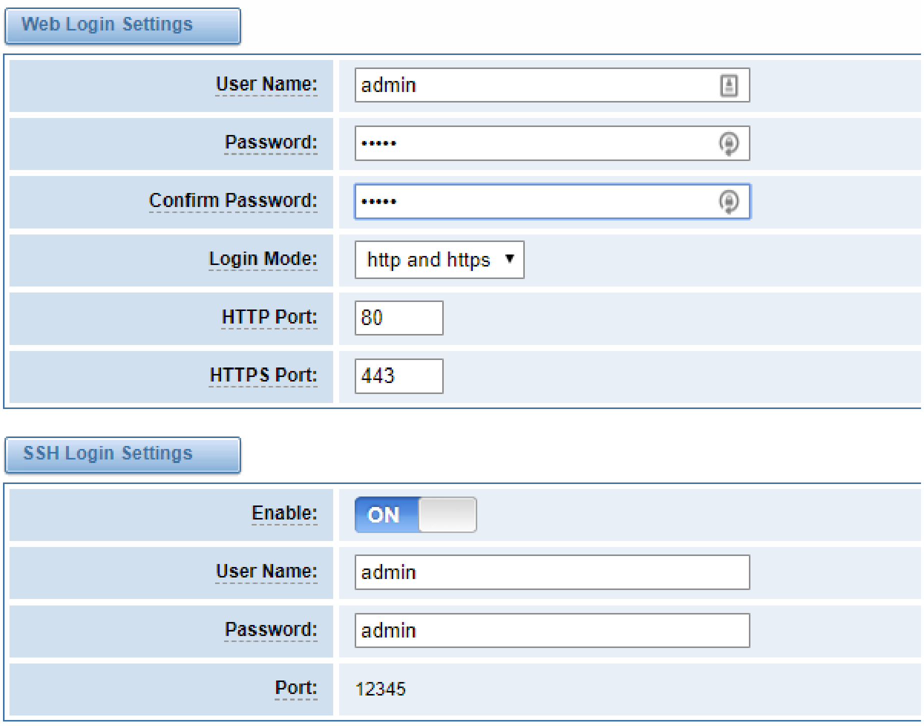
2.4 Login Settings

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 logout, just rewriting your new user name and password will be OK. Also you can specify the web server port number. Usually the web login default mode is “http and https”. For safety, you can switch to “only https” mode.

Table 2-4-1 Description of Web Login Settings

Options	**Definition**
User Name	Your gateway does not have administration role. All you can do here is defining the user name and password to manage your gateway. And it has all privileges to operate your gateway .User Name: 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.
Login Mode	Specify the web login mode: **http and https**, **only https**. Default is http and https.
Port	Specify the web server port number. Do not use port 443 which is reserved for HTTPS.

Figure 2-4-1 Login Settings



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

2.5 General

2.5.1 Language Settings

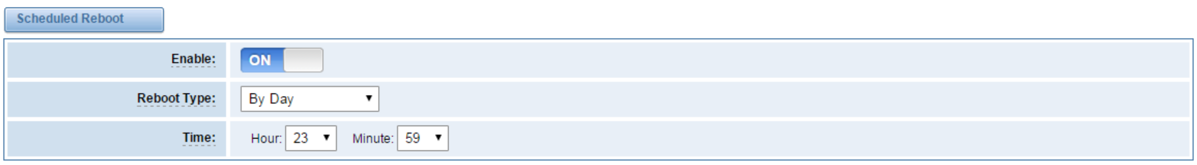
You can choose different languages for your system. If you want to change language, you can switch “Advanced” on, then “Download” your current language package. After that, you can modify the package with the language you need. Then upload your modified packages, “Choose File” and “Add”.

Figure 2-5-1 Language Settings

2.5.2 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 2-5-2 Reboot Types



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

2.6 Tools

On the “Tools” pages, there are reboot tools, update Firmware, upload Configuration, backup

Configuration and Restore Configuration toolkits.

2.6.1 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.

2.6.2 Update Firmware

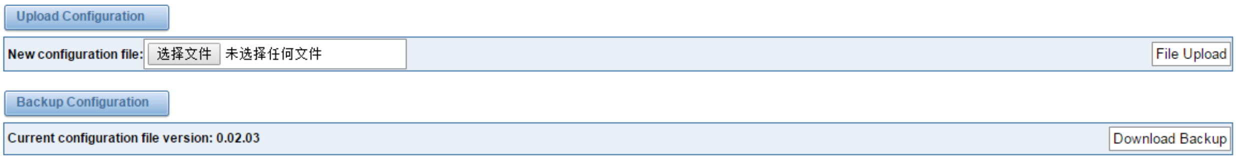
We offer two kinds of update types for you. You can choose System Update or System Online Update. System Online Update is an easier way to update your system, if you choose that, you will see some information below.

Figure 2-6-2 Prompt Information

2.6.3 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.

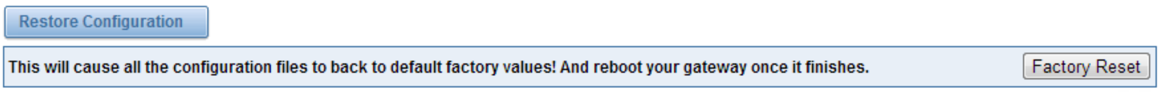
Figure 2-6-3 Upload and Backup



2.6.4 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



2.7 System 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 2-7-1 System Information

Model Name:	DGW-1004
Firmware Version:	2.5.3
Firmware Build:	1907
Hardware Version:	1.1
Port Amount:	4
Storage Usage:	100.2M/193.5M (56%)
Memory Usage:	11.107 % Memory Clean
Kernel Build Time:	2019-Jul-2-17:10:59
Contact Address:	10/F, Building 6-A, Baoneng Science and Technology Industrial Park, Longhua New District, Shenzhen, Guangdong,China
Tel:	+86-755-82535461
Fax:	+86-755-83823074
E-Mail:	support@openvox.cn
Web Site:	http://www.openvox.cn
System Time:	2019-7-11 12:22:21
System Uptime:	7 days 18:16:14

3 T1/E1

3.1 General

Figure 3-1-1 General Settings

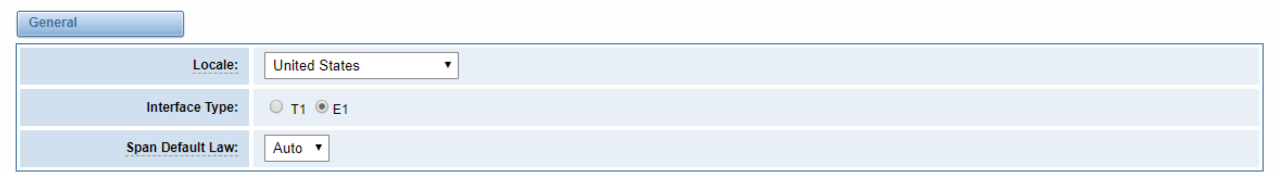


Table 3-1-1 Definition of General Settings

Options	Definition
Locale	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.
Interface Type	It shows you the current type of port. It has two type: E1 and T1.

Figure 3-1-2 Advanced interface type

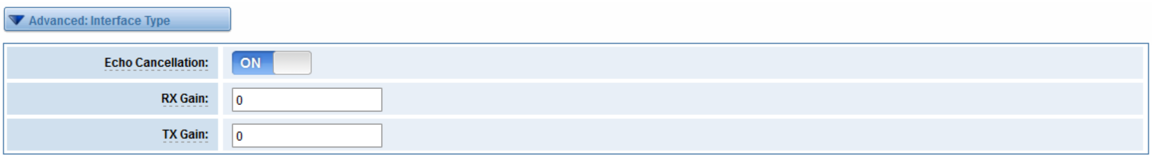


Table 3-1-2 Definition of advanced interface type

Options	Definition
RX Gain	Gain for the RX (receive -into Asterisk) channel.Default:0.0.
TX Gain	Gain for the TX (transmit -out of Asterisk Asterisk) channel.Default:0.0.

Figure 3-1-3 Port Details

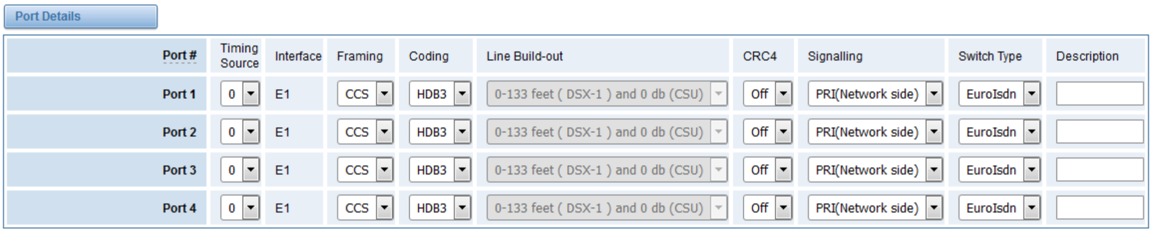


Table 3-1-3 Definition of Port Details

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 the interface.
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.

3.2 PRI

Figure 3-2-1 ISDN: Signaling

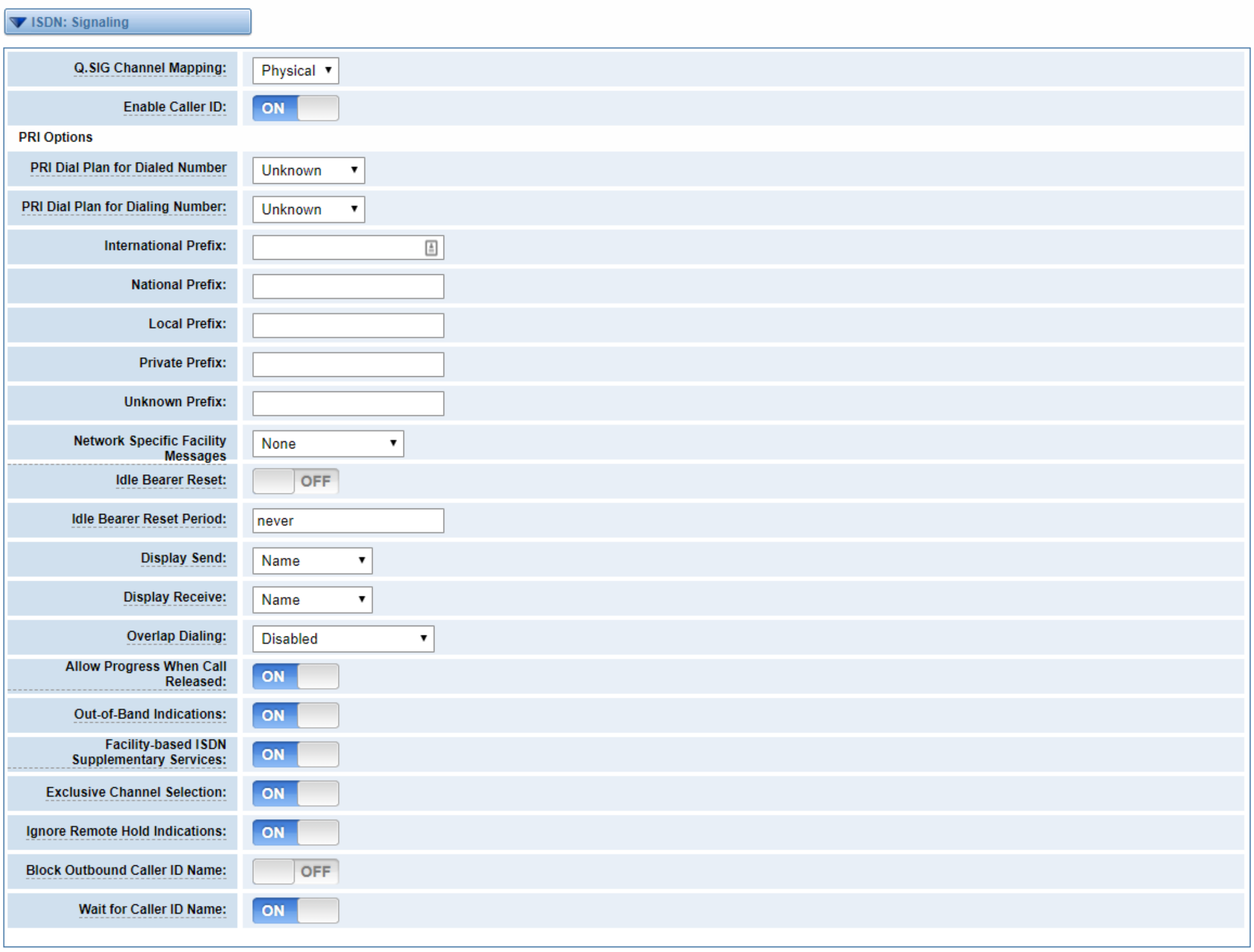


Table 3-2-1 Definition of Signaling

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 use caller ID.

PRI Dial Plan for Dialed Number	PRI Dialplan: The ISDN-level Type of Number (TON) or numbering plan, used for the dialed number. Leaving this as 'unknown' (the default) works for most cases. 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 number's numbering plan). In North America, the typical use is sending the 10 digits; caller ID 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',' tollfreemgacom',' account'.
Idle Bearer Reset	Whether or not to reset unused B channels.
Idle Bearer Reset Period	Sets the time in seconds between restart of unused B channels; defaults to 'never'.
Display Send	Send/receive ISDN display IE options, the display options are a comma separated list of the following options: **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/FACILITY) for COLP 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. Sent text goes out in default to 'name'.
Display Receive	Send/receive ISDN display IE options. The display options are a comma separated list of the following options: **block**: Do not pass display text data. **name_initial**: Use display text in SETUP/CONNECT messages as the party name. **bame_update**: Use display text in other messages (NOTIFY/FACILITY) for COLP 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. Sent text goes out in default to 'name'.
Overlap Dialing	Enable overlap dialing mode--sending overlap digits.
Allow Progress When Call Released	Allow inband audio (progress) when a call is DISCONNECT Ted by the 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 enable transmission of facility-based ISDN supplementary services (such as caller name form CPE over facility), enable this option. 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 ignore remote hold indications (and use MOH that is supplied over the B channel) enable this option.
Block Outbound Caller ID Name	Enable if you need to hide just the name and the number for legacy PBX use. Only applies to PRI channels.
Wait for Caller ID Name	Support caller ID on call waiting.

3.3 MFC/R2

3.3.1 MFC/R2 Signaling

Figure 3-3-1 MFC/R2 Signaling

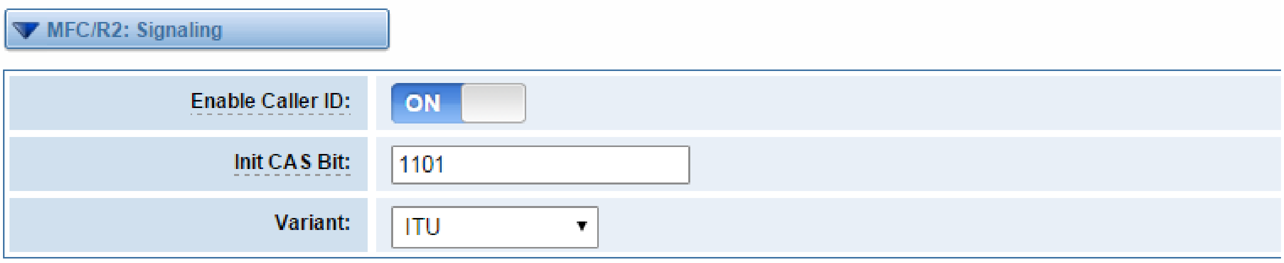
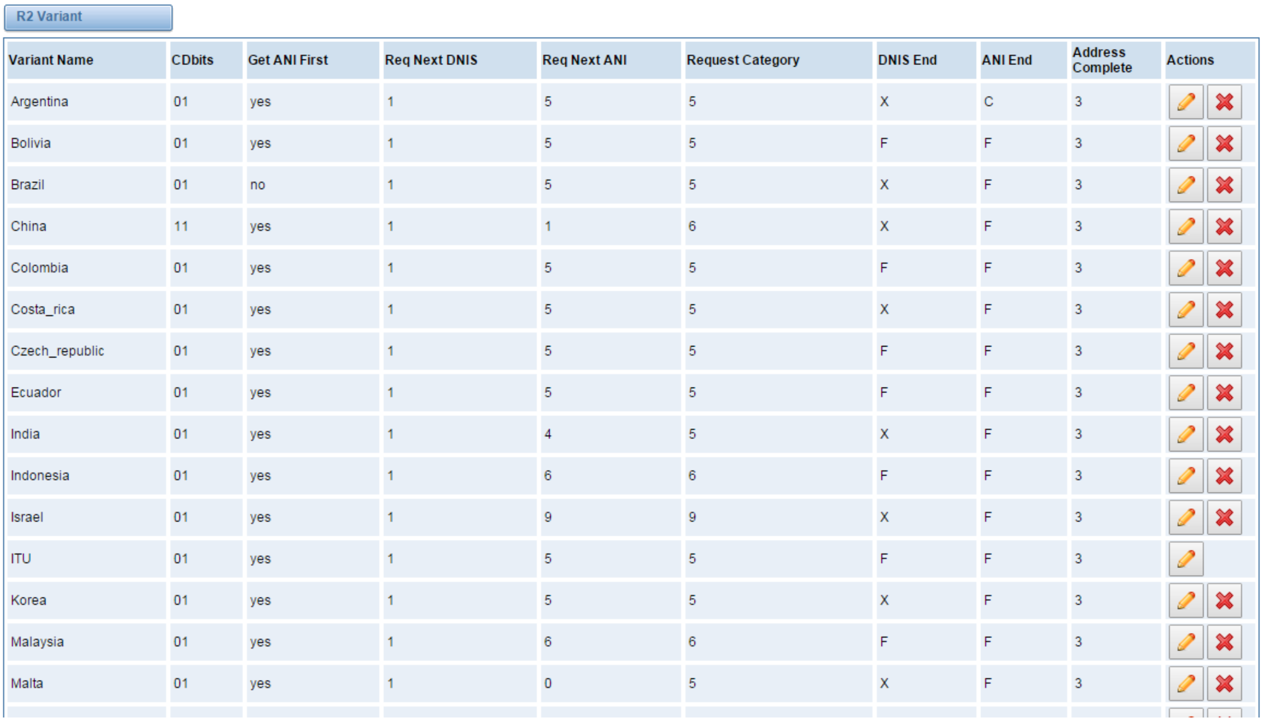


Table 3-3-1 Definition of MFC/R2 Signaling

options	**Definition**
Enable Caller ID	Whether or not to use caller ID.
Init CAS Bit	The initial position of the CAS bits.
Variant	The standard of MFCR2: ITU, ANSI and China.

3.3.2 Modify R2 variant

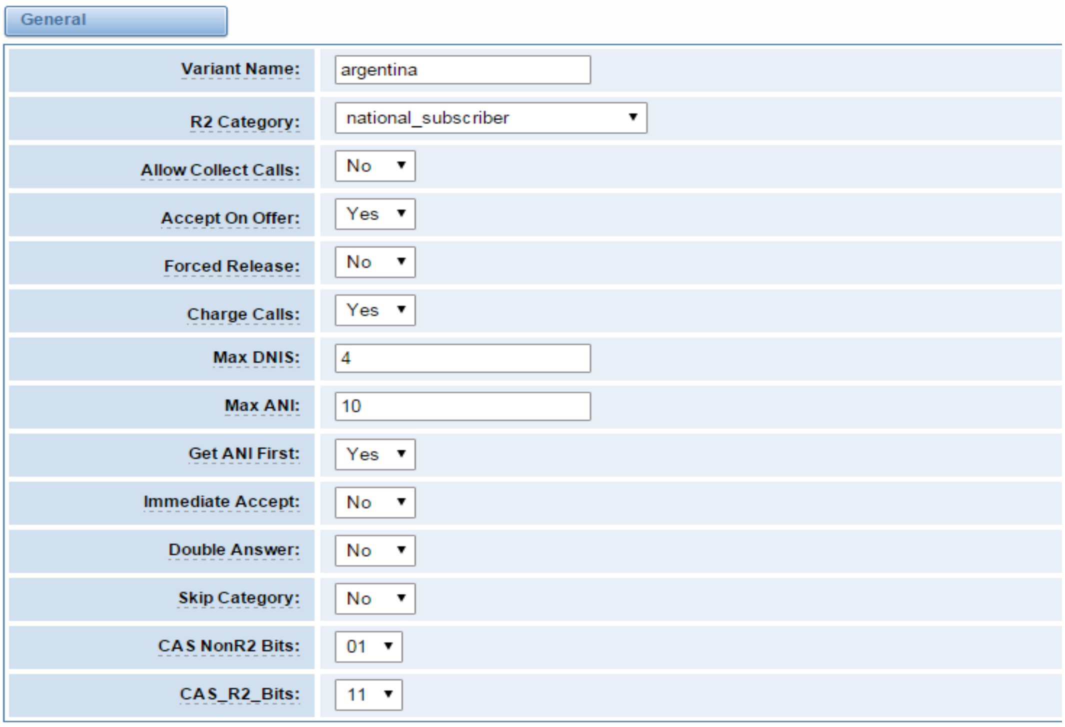
Figure 3-3-2 R2 Variant


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.

Figure 3-3-3 General


General

Variant Name:	argentina
R2 Category:	national_subscriber
Allow Collect Calls:	No
Accept On Offer:	Yes
Forced Release:	No
Charge Calls:	Yes
Max DNIS:	4
Max ANI:	10
Get ANI First:	Yes
Immediate Accept:	No
Double Answer:	No
Skip Category:	No
CAS NonR2 Bits:	01
CAS_R2_Bits:	11

Table 3-3-2 Definition of General

Options	**Definition**
Variant Name	The variant name.
R2 Category	national subscriber works just fine usually.
Allow Collect Calls	Default is to block collect calls.
Accept On Offer	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.
Forced Release	Brazil use a special signal to force the release of the line instead of the normal clear back signal.
Charge Calls	Whether or not report to the other end 'accept call with charge', when interconnecting with old PBXs this may be useful.
Max DNIS	Max amount of DNIS to ask for.
Max ANI	Max amount of ANI to ask for.
Get ANI First	Whether or not get the ANI before getting DNIS.
Immediate Accept	This feature allows to skip the use of Group B/II signals and go directly to the accepted state for incoming calls.
Double Answer	This will cause that every answer signal is changed by answer->clear back->answer, sort of flash.
Skip Category	Skip request of calling party category and ANI.
CASNonR2 Bits	Which bits are never used.
CAS_R2_Bits	Which bits will be used.

Figure 3-3-4 Timer

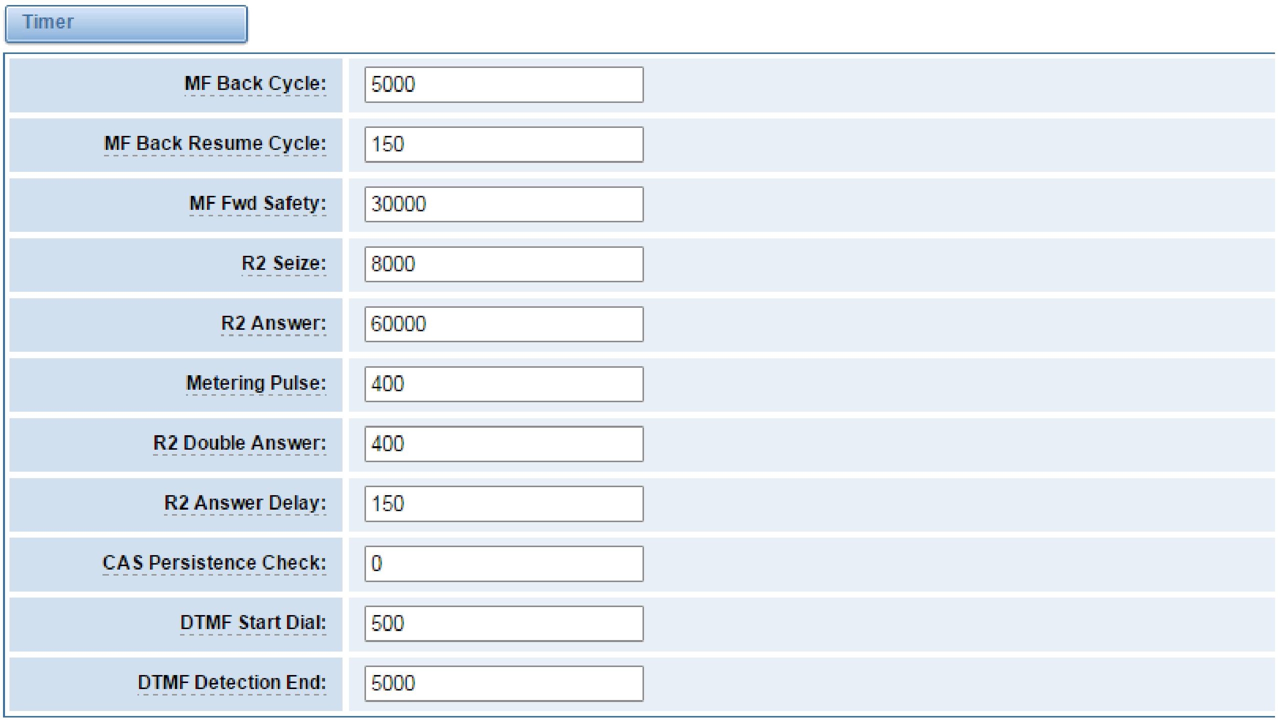


Table 3-3-3 Definition of Timer

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 3-3-5 Group A

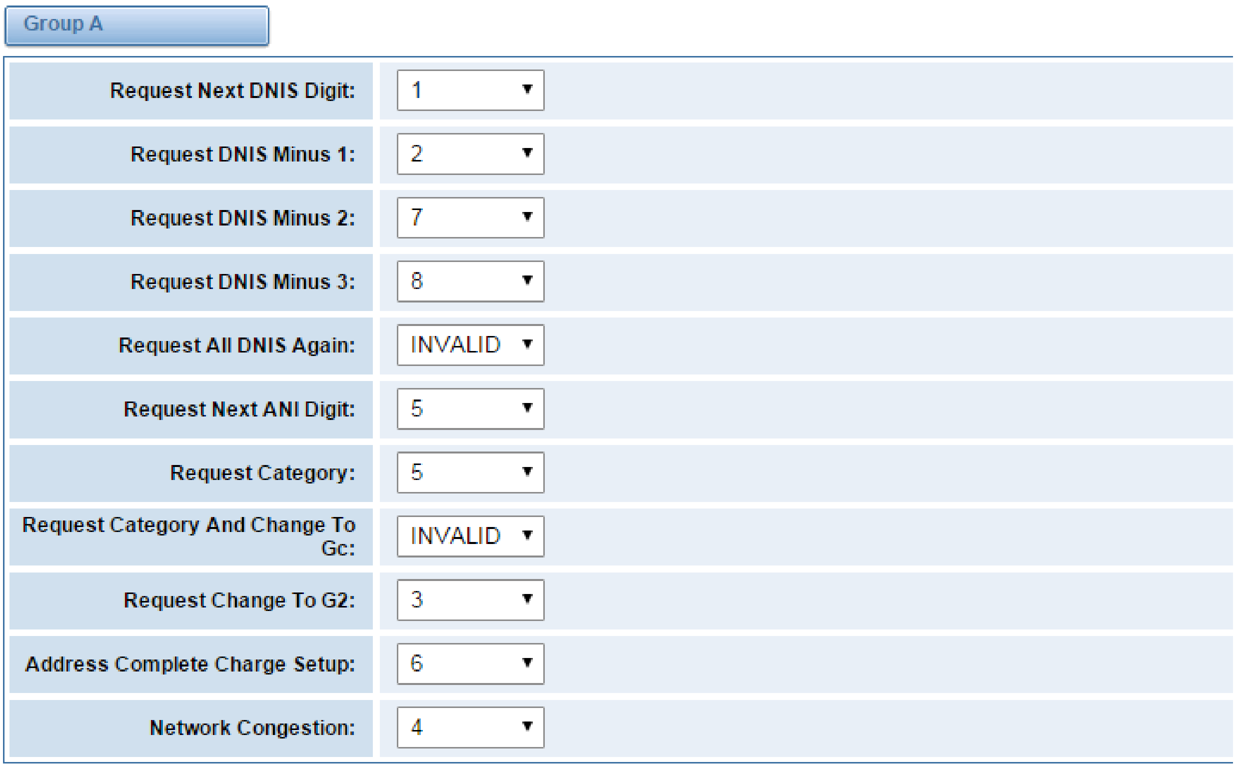


Figure 3-3-6 Group B

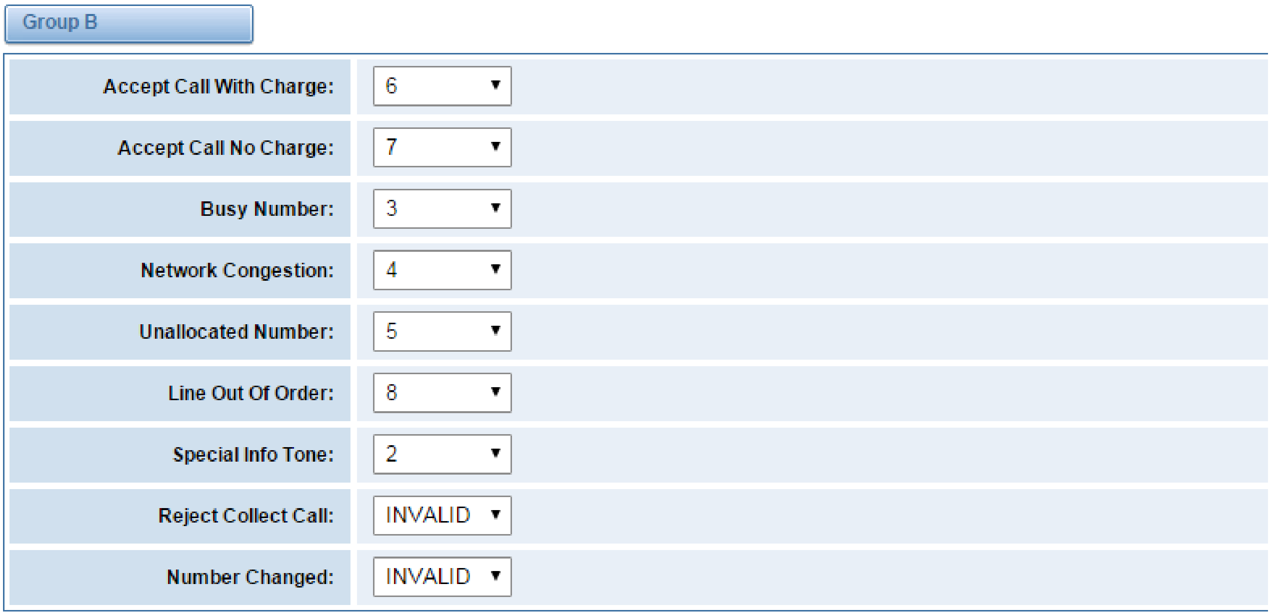


Figure 3-3-7 Group C

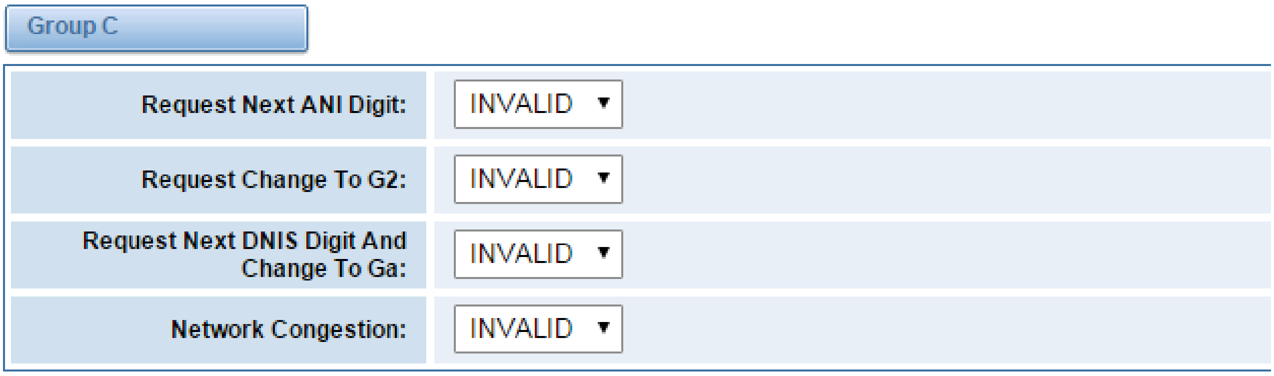


Figure 3-3-8 Group 1

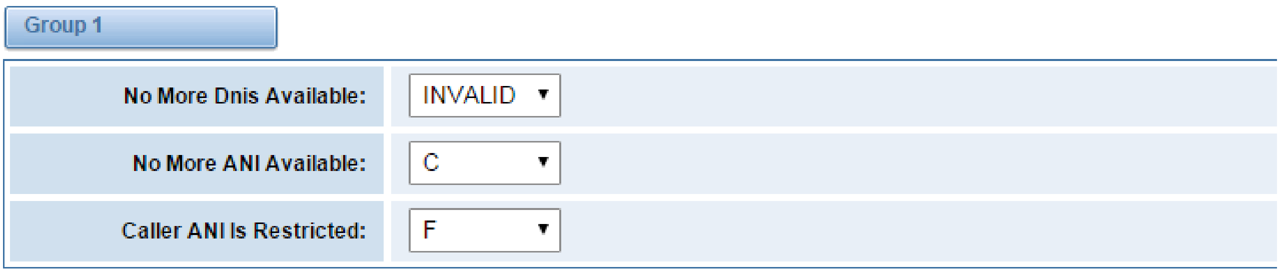
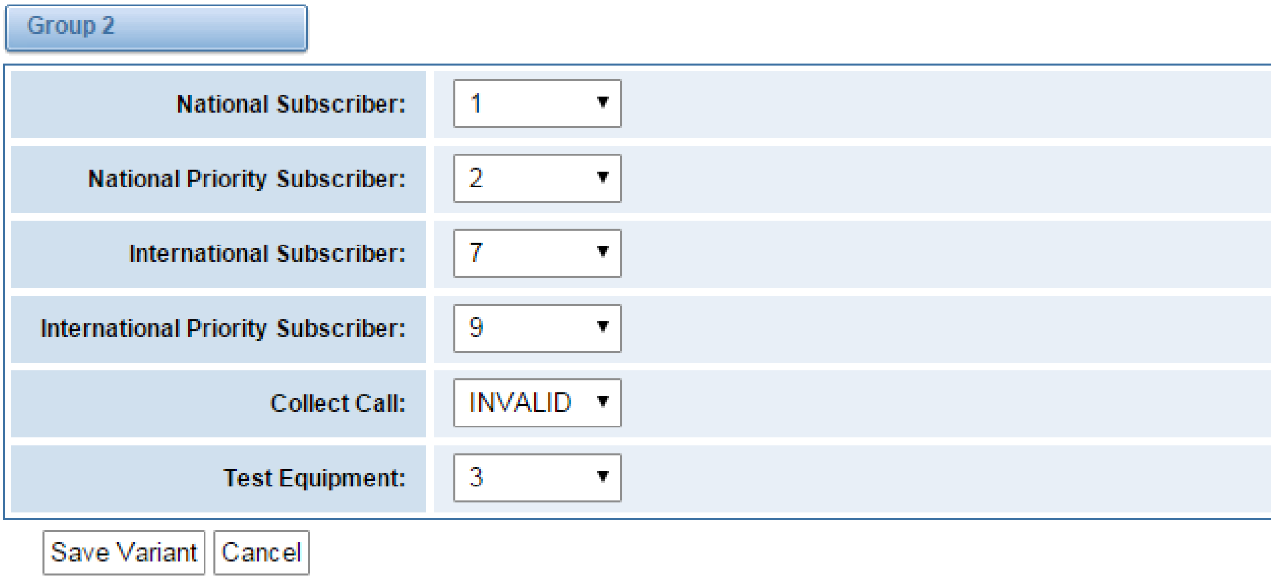


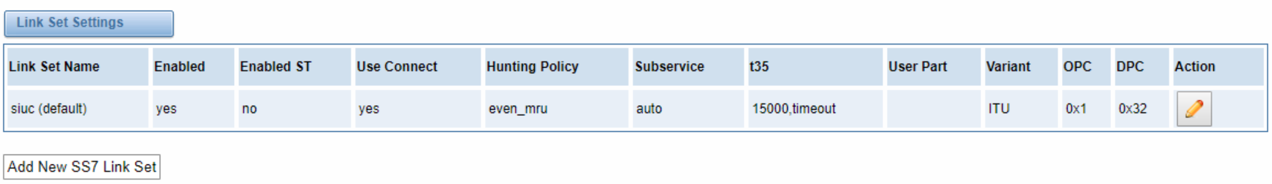
Figure 3-3-9 Group 2



3.4 Chan-SS7

3.4.1 Link Set Settings

Figure 3-4-1 Link Set Settings


Link Set Settings

Link Set Name	Enabled	Enabled ST	Use Connect	Hunting Policy	Subservice	t35	User Part	Variant	OPC	DPC	Action
siuc (default)	yes	no	yes	even_mru	auto	15000,timeout		ITU	0x1	0x32	

Add New SS7 Link Set

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

Figure 3-4-2 Chan-SS7 Link Set Settings

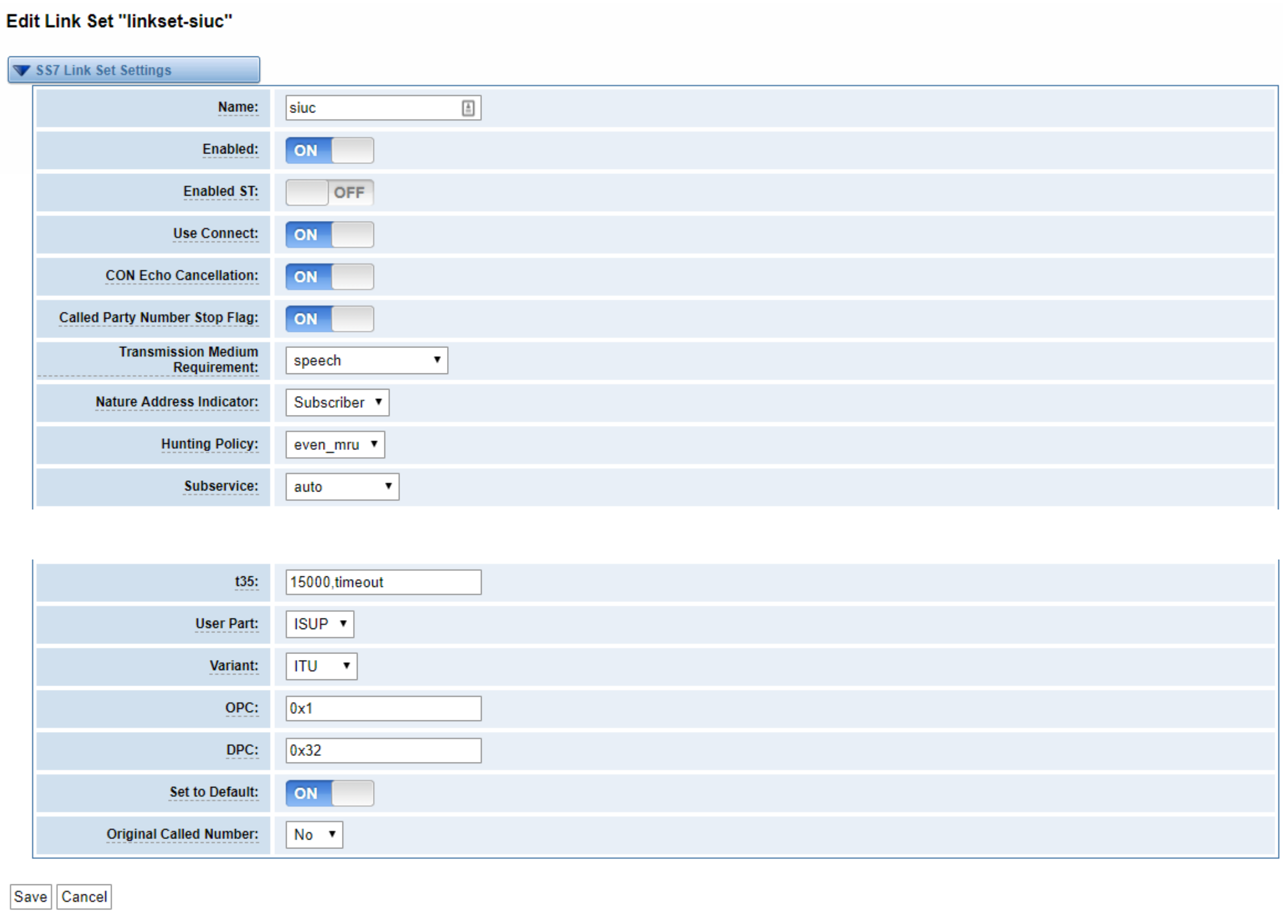


Table 3-4-1 Definition of SS7 Link Set Settings

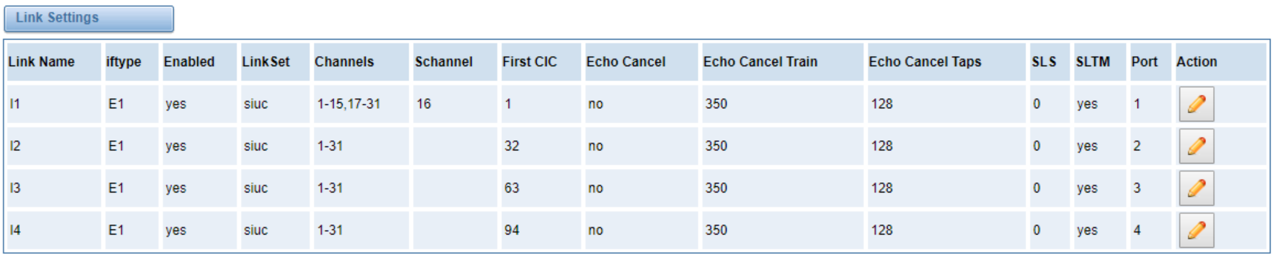
options	**Definition**
Name	The link set name.
Type	SS7 variant.
Enabled ST	This is used to decide whether end-of-pulsing is not used to determine when incoming address is complete.
Use Connect	This setting specifies whether to reply incoming call with CON rather than ACM and ANM.
CON Echo Cancellation	This setting will enable Echo Cancellation when 'Use Connect' is enabled.

Called Party Number Stop Flag	Add a stop flag 'F' before called number send.
Transmission Medium Requirement	Specify the bearer circuit capabilities (speech, 3.1-kHz audio, 64-Kb unrestricted, and so forth) that are needed for the call being set up.
Nature Address Indicator	the nature of address indicator field is national or subscriber(default).
Hunting Policy	This sets the hunting policy, ie. the algorithm used to allocate a circuit for outgoing calls. This should be configured appropriately at each end of the SS7 link to minimize the risk of call collision, both ends try to make an outgoing call on the same circuit at the same time.
Subservice	The subservice field: national, international, 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 dialling dial plan, you might choose:4000,st.
User Part	The type of User Part.
Variant	running under SS7 standard.
OPC	The point code for this SS7 signalling point.
DPC	The destination point code.

Set to Default	Set the linkset as the default linkset.
Original Called Number	Information sent in the forward direction when a call is redirected and identifies the original called party.

3.4.2 Link Settings

Figure 3-4-3 Link Settings

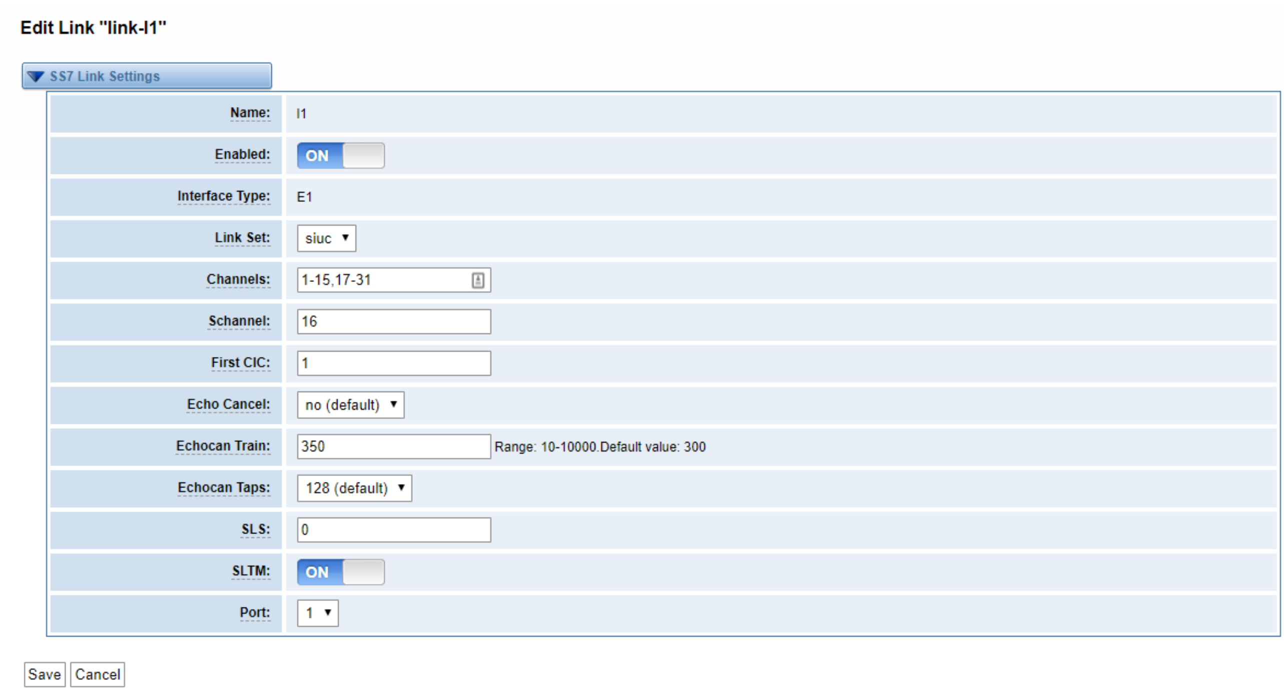


Link Settings

Link Name	iftype	Enabled	LinkSet	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	0	yes	1	
l2	E1	yes	siuc	1-31		32	no	350	128	0	yes	2	
l3	E1	yes	siuc	1-31		63	no	350	128	0	yes	3	
l4	E1	yes	siuc	1-31		94	no	350	128	0	yes	4	

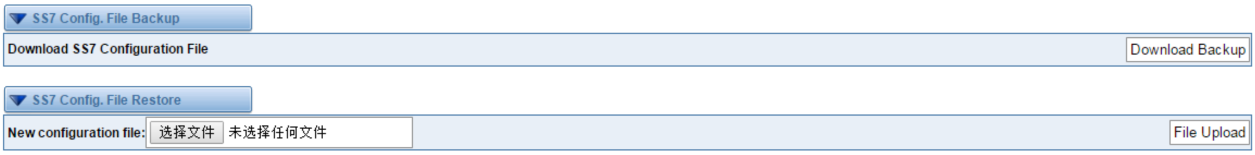
You can click button as shown below.

Figure 3-4-4 SS7 Edit Link Settings



3.4.3 SS7 Configuration file backup and restore

Figure 3-4-5 Configuration file backup and restore



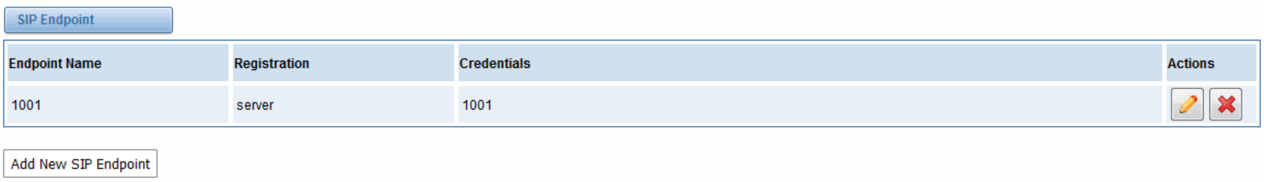
4 VOIP

4.1 VOIP Endpoints

4.1.1 SIP Endpoints

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

Figure 4-1-1 SIP Status



4.1.2 Main Endpoint Settings

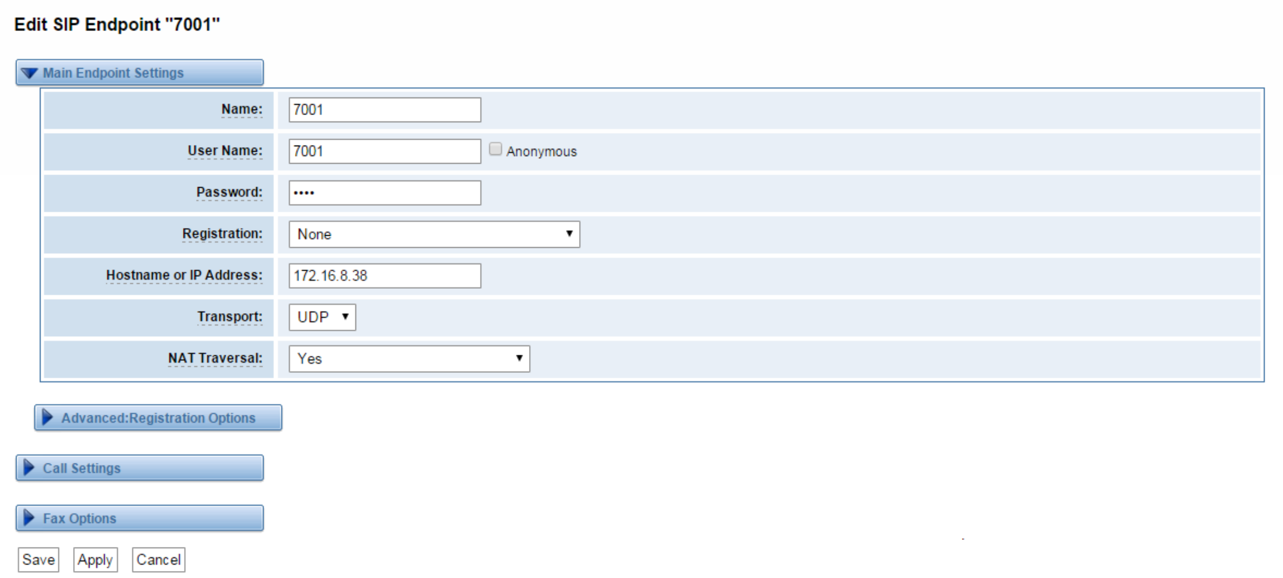
You can click Add New SIP Endpoint button to add a new SIP endpoint, and if you want to modify existed endpoints, you can click button.

There are three 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:

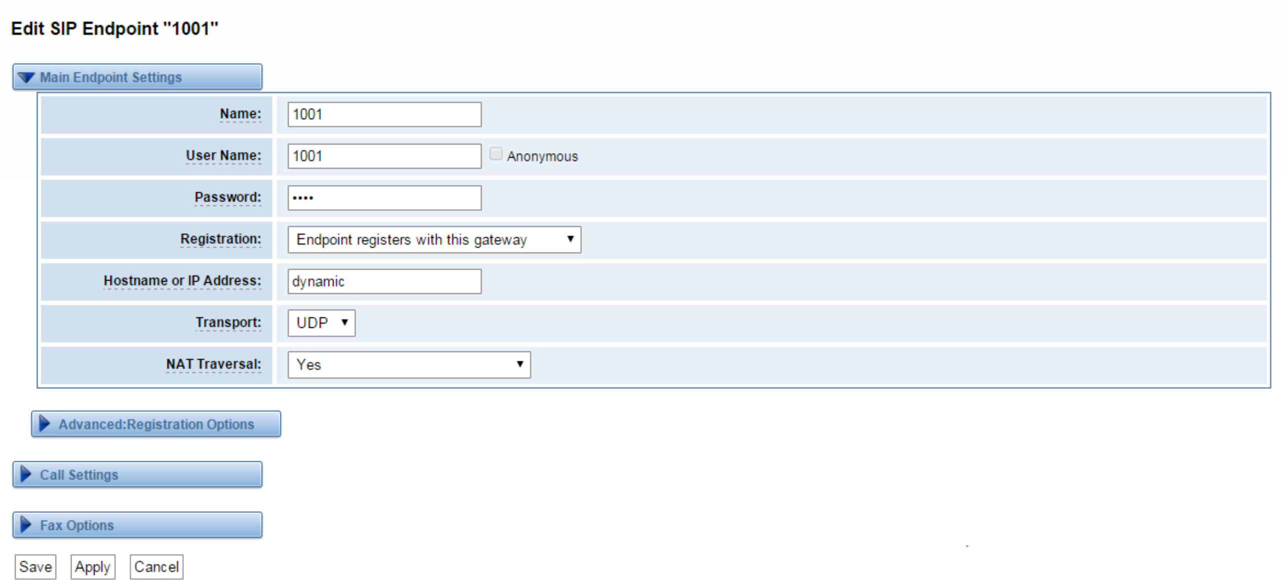
If you set up a SIP endpoint by registration "None" to a server, then you can't register other SIP endpoints to this server. (If you add other SIP endpoints, this will cause Out-band Routes and Trunks confused.)

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 works 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 register 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.

4.1.3 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.

4.1.4 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	Whether or not the Remote-Party-ID header should be trusted.

Remote-Party-ID	
Send Remote-Party-ID	Whether or not to send the Remote-Party-ID header.
Caller ID Presentation	Whether or not to display Caller ID.

4.1.5 Advanced Signaling Settings

Table 4-1-4 Definition of Signaling Options

Options	**Definition**
Progress Inband	If we should generate in-band ringing. Always use 'never' to never use in-band signaling, 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.

4.1.6 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 Refresh Interval	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.

4.1.7 Fax Options

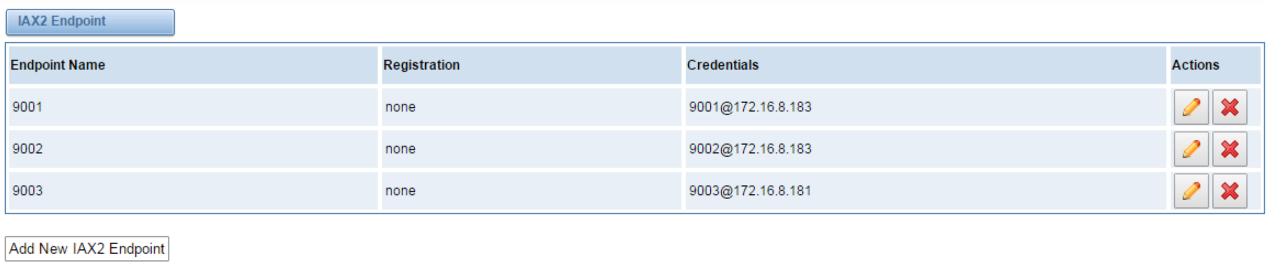
Table 4-1-6 Definition of Fax Options

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.

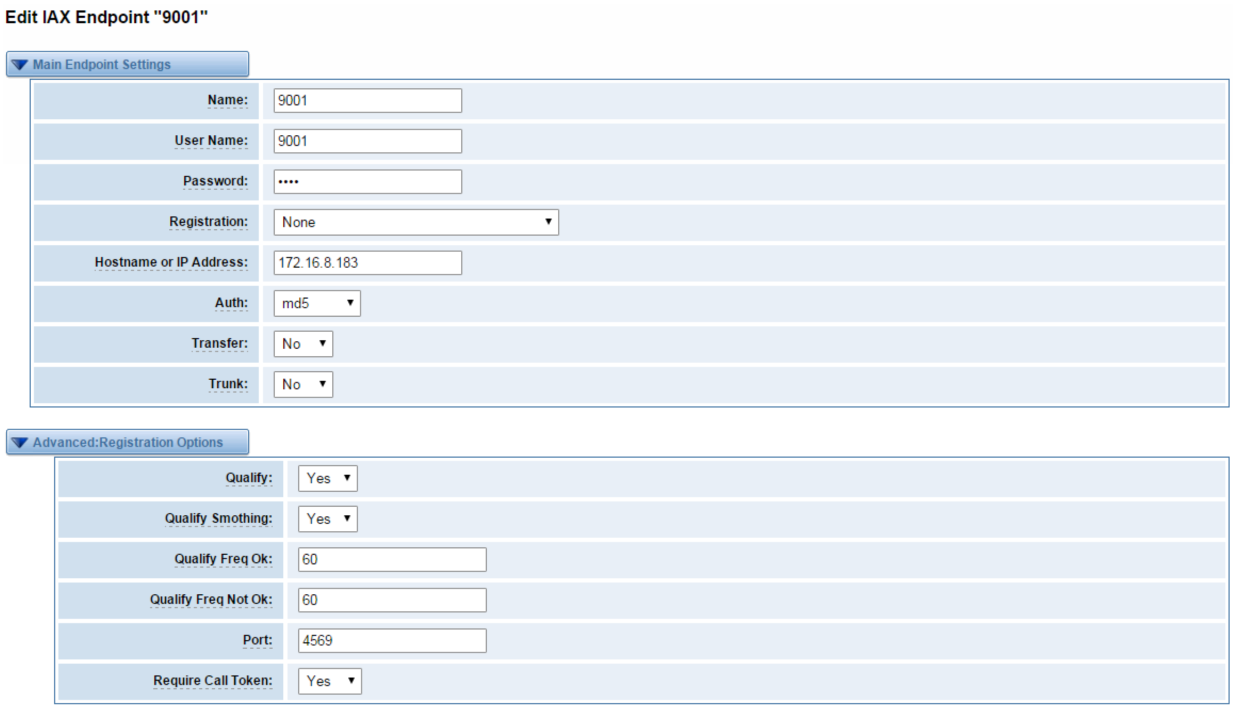
4.2 IAX2 Endpoint

Figure 4-2-1 IAX2 Endpoint



You can click  button as shown below

Figure 4-2-2 Edit IAX Endpoint "9001"



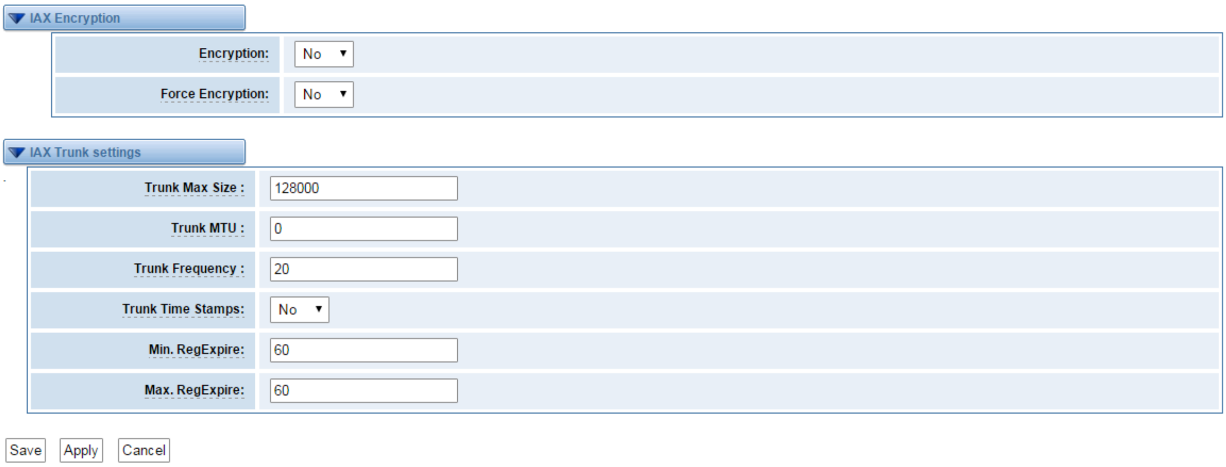


Table 4-2-1 Definition of IAX2 Endpoint

Options	**Definition**
Name	A name which is able to read by human. And it's only used for user's reference.
User name	User name the endpoint will use to authenticate with the gateway.
Password	Password the endpoint will use to authenticate with gateway. Allowed characters.
Registration	Whether this endpoint will register to this gateway or this gateway to the endpoint.
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.
Auth	Authentication method for connections.
Transfer	Disable or not IAX2 native transfer.
Trunk	Use IAX2 trunking with this host.
Qualify	Whether or not to check the endpoint's connection status.

Qualify Smothing	Use an average of the last two PONG result to reduce falsely detected LAGGED host. The default is 'no'.
Qualify Freq Ok	How frequently to ping the peer when everything seems to be OK, in milliseconds.
Qualify Freq not Ok	How frequently to ping the peer when it's either; LAGGED or UNAVAILABLE, in milliseconds.
Port	The port number the gateway will connect to at this endpoint.
Encryption	Enable IAX2 encryption. The default is no.
Force Encryption	Force encryption insures no connection is established unless both sides support encryption. By turning this option on, encryption is automatically; turned on as well. The default is no.
Trunk Max Size	Defaults to 128000 bytes, which supports up to 800; calls of ulaw at 20ms a frame.
Trunk MTU	With a large amount of traffic on IAX2 trunk, there is a risk of bad voice quality when allowing the Linux system to handle fragmentation of UDP packets. Depending on the side of each payload, allowing the OS to handle fragmentation may not be very efficient. This setting sets the maximum transmission unit for AIX2 UDP trunking. The default is 1240 bytes which means if a trunk's payload is over 1240 bytes for every 20ms it will be broken into multiple 1240 bytes messages. Zero disables this functionality and let's the OS handle fragmentation.
Trunk Frequency	How frequently to send trunk msgs (in ms). This is 20ms by default.

Trunk Time Stamps	Should we send timestamps for the individual sub_frames within trunk frames? There is a small bandwith use for these (less than 1kbps/call), but they ensure that frame timestamps get sent end-to-end properly. If both ends of all your trunks go directly to TDM, _and_your trunkfreq equals the frame length for your codecs, you can probably suppress these. The receiver must also need to have it enabled.
Min. RegExpire	Minimum amounts of time that IAX2 peers can request as a registration interval (in seconds).
Max. RegExpire	Maximum amounts of time that IAX2 peers can request as a registration expiration interval (in seconds).

4.3 Advanced SIP Settings

4.3.1 Networking

Table 4-3-1 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.

4.3.2 Advanced NAT Settings

Table 4-3-2 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 NATed 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.

4.3.3 Advanced RTP Settings

Table 4-3-3 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.

4.3.4 Parsing and Compatibility

Table 4-3-4 Instruction of Parsing and Compatibility

Options	**Definition**
Strict RFC	Check header tags, character conversion in URIs, and multiline headers for strict SIP compatibility (default is yes).
Send Compac	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	Maximum allowed time of incoming registrations and subscriptions (seconds).
Minimum Registration Expiry	Minimum length of registrations/subscriptions (default 60).
Default	Default length of incoming/outgoing registration.
Registration Timeout	How often, in seconds, to retry registration calls. Default 20 seconds.
Number of Registration	Number of registrations attempts before we give up.0=continue forever, hammering the other server until it accepts the registration. Default is 0 tries, continue forever.

4.3.5 Security

Table 4-3-5 Instruction of Security

Option	**Definition**
Match Auth	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.

4.3.6 Media

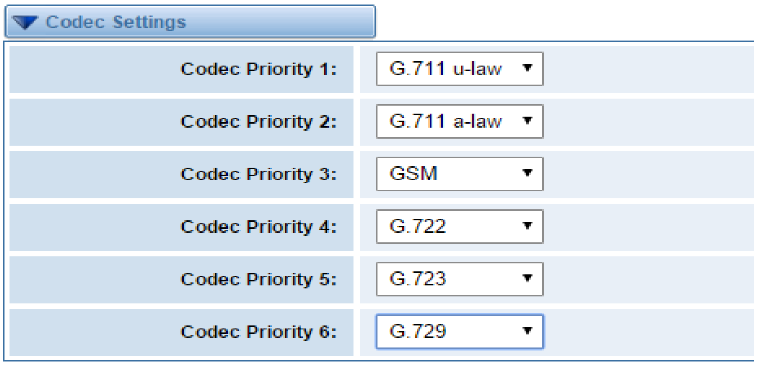
Table 4-3-6 Instruction of Media

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

4.3.7 Codec Settings

Select codecs from the list below.

Figure 4-3-1 Codec Settings



4.4 Advanced IAX2 Settings

Table 4-4-1 Instruction of General

Options	**Definition**
Bind Port	Bind port and bindaddr may be specified.
Enable IAXCompat	More than once to bind to multiple addresses, but the first will be the default.
Enable No Checksums	Set iaxcompat to yes if you plan to use layered switches or some other scenario which may cause some delay when doing a lookup in the dialplan. It incurs a small performance hit to enable it. This option cause Asterisk to spawn a separate thread when it receives an IAX DPREQ (Dialplan Request) instead of blocking while it waits for a response.
Enable Delay Reject	Disable UDP checksums (if no checksums is set, then no checksums will be calculated/checked on system supporting the feature).
ADSI	ADSI (Analog Display Services Interface) can be enable if you have (or may

	have) ADSI compatible CPE equipment.
SRV Lookup	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.
Auto Kill	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 4-4-2 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 fall back to English(en).

Table 4-4-3 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 4-4-4 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 notices 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 interpolations 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 these 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 4-4-5 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	The 'maxcallnumbers' option limits the amount of call numbers allowed for each individual remote IP address. Once an IP address reaches its call number limit, no more new connections are allowed until the previous ones close. This option can be used in a peer definition as well, but only takes effect for the IP of a dynamic peer after it completes registration.
MaxCallNumbers_ Nonvalidated	The 'maxcallnumbers-nonvalidated' is used to set the combined number of call numbers that can be allocated for connections where call token validation has been disabled. Unlike the 'maxcallnumbers' option, this limit is not separate for each individual IP address. Any connection resulting in a non-call token validated call number being allocated contributes to this limit. For use cases, see the call should be sufficient in most cases.

Table 4-4-6 Instruction of Quality of Service

Options	Definition
Tos	Type of service
Cos	Class of service

4.5 Advanced fax setting

Table 4-5-1 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	Manager events with 'call' class permissions will receive events indicating

events to manager session	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

The gateway embraces the flexible and friendly routing settings for user. It supports up to 512 routing rules and about 100 pairs of calleeID/callerID manipulations can be set in a rule. It supports DID function (The usage of DID function: [How to use DID function with OpenVox T1/E1 Gateway](#)). The gateway support trunk group and trunk priority management.

5.1 Call Routing Rule

Figure 5-1-1 Routing Rules

<table>
<tr><th>Move</th><th>Order</th><th>Rule Name</th><th>From</th><th>To</th><th>Rules</th><th>Actions</th></tr>
<tr><td></td><td>1</td><td>6001to540</td><td>iax-6001</td><td>sip-540</td><td>Callee_Dial_pattern
+|[]|(- +)|
Caller_Dial_pattern
+|[]|(- +)|</td><td></td></tr>
<tr><td></td><td>2</td><td>iaxtoports</td><td>iax-6001</td><td>grp-ports</td><td>Callee_Dial_pattern
+|[]|(- +)|
Caller_Dial_pattern
+|[]|(- +)|</td><td></td></tr>
<tr><td></td><td>3</td><td>6001toport</td><td>sip-7001</td><td>grp-ports</td><td>Callee_Dial_pattern
+|[]|(- +)|
Caller_Dial_pattern
+|[]|(- +)|</td><td></td></tr>
</table>

New Call Routing Rule | Save Orders

You are allowed to set up new routing rule by New Call Routing Rule, 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 Save Orders button to save what you set. **Rules** will show current routing rules. Otherwise you can set up unlimited routing rules.

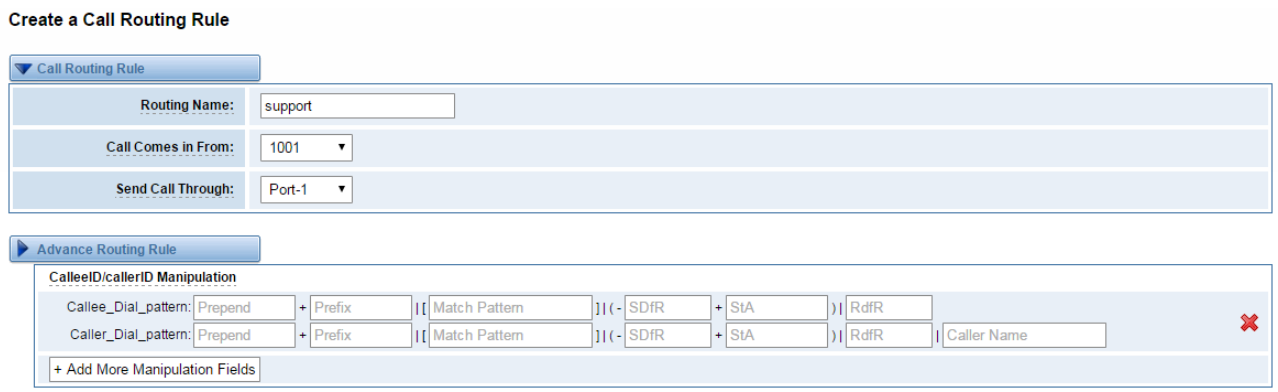
There is an example for routing rules number conversion, it transforms calling, called number at the same time. Suppose you want eleven numbers start at 159 to call the eleven numbers of start at 136. Calling transform delete the three numbers from left, then writing number 086 as prefix, delete the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

Table 5-1-1 Example for routing rules number conversion

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

You can click New Call Routing Rule button to set up your routings.

Figure 5-1-2 Example of Setup 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 1001, "prepend", "prefix" and "match pattern" in "Advanced Routing Rule" are ineffective, and just "CallerID" option is available.

Table 5-1-2 Definition of Routing Options

Options	Definition
Routing Name	The name of this route. Should be used to describe what types of calls.
Call Comes in From	The launching point of incoming calls.
Send call Through	The destination to receive the incoming calls.

Table 5-1-3 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) **.** wildcard, 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 number 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. **StA (Suffix to Add)**: Designated information to be added to the right end of the

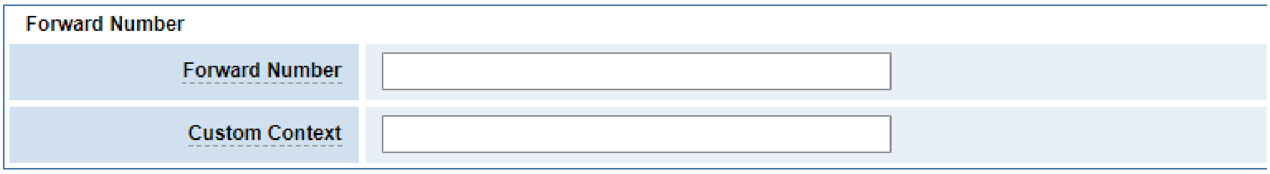
	current number. **RDfR (Reserved Digits from Right)**: The number of digits to be reserved from the right end of the number. If the value of this item under the length of the current number, the whole number will be reserved. **Caller Name**: What caller name would you like to set before sending this call to the endpoint. Sets Caller*ID data on the channel: **SIP_FROM(name),SIP_FROM(number)** : defalut:Sip From and Number eg: When selecting SIP From, Name is Peter and Number is 402. The From mode is: \"Peter\"<sip:402@172.16.6.239;transport=UDP>;tag=bd481672 **SIP_TO(name),SIP_TO(number)** : defalut: EXTEN eg: When selecting SIP To, Name is Jason and Number is 401. To mode is: \"Jason\"<sip:401@172.16.6.239;transport=UDP> **None**: Set Caller*ID number which is to be send on the channel from PBX(manipulated if it sets).
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 5-1-3 Time Patterns that will use this Route

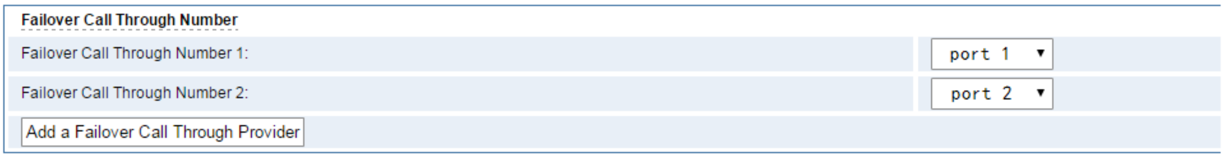
If you configure like this, then from January to March, from the first day to the last day of these months, from Monday to Thursday, from 00:00 to 02:00, during this time (meet all above time conditions), all calls will follow this route. And the time will synchronize with your Sever time.

Figure 5-1-4 Forward Number



You can configure forward number when you have a transfer call. You can also set your custom context as you like before sending the call to the endpoint.

Figure 5-1-5 Failover Call Through Number

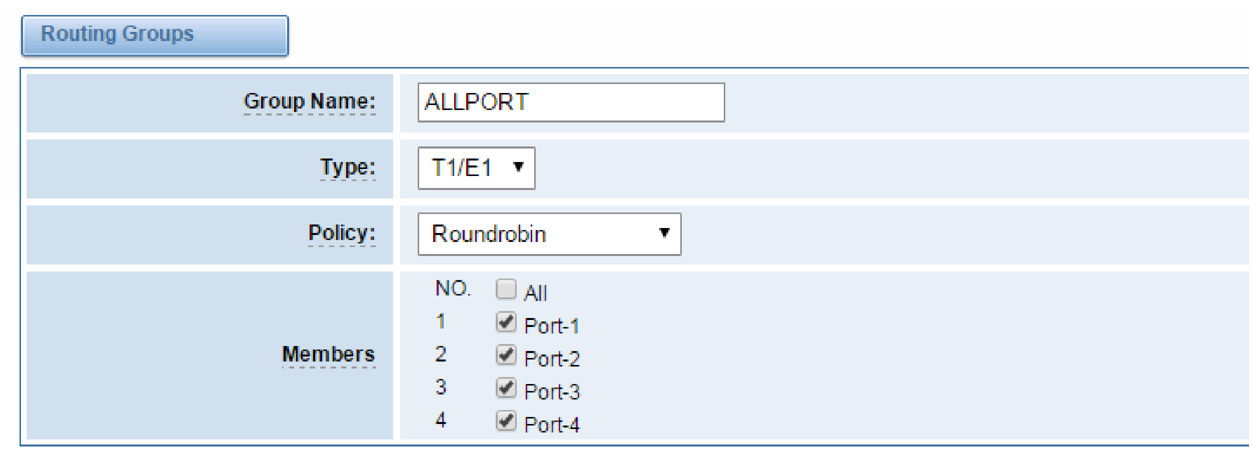


You can add one or more “Failover Call Through Numbers”.

5.2 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, “Network Settings”, “DDNS Settings”, “Toolkit” and “Static Route Settings”.

6.1 Network 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

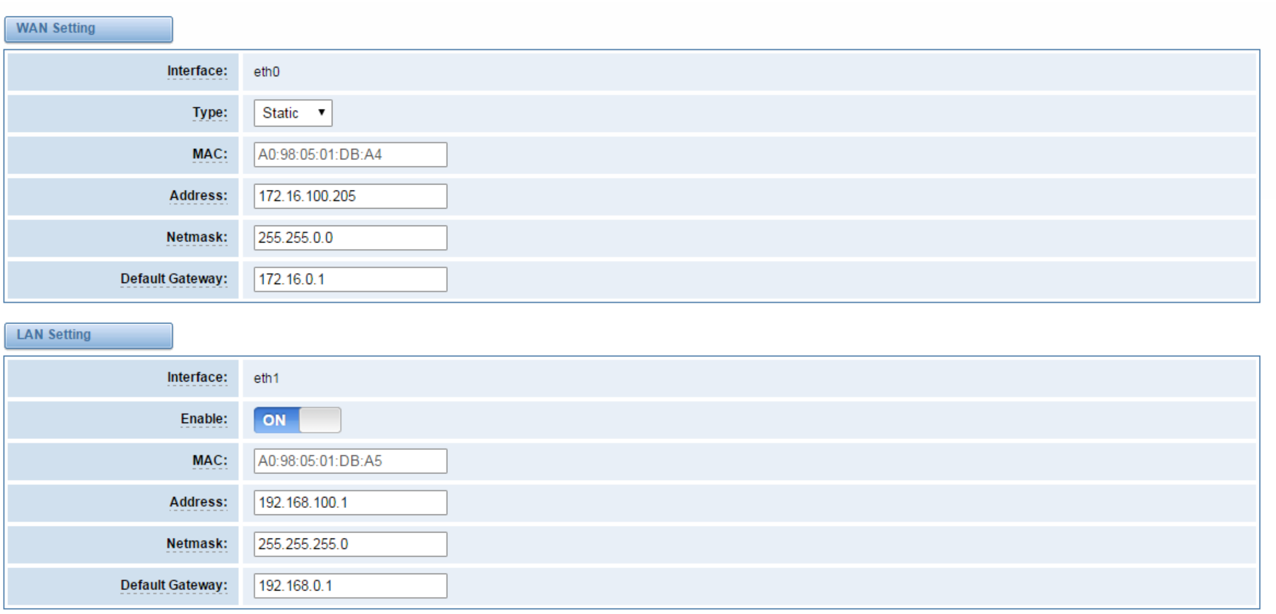


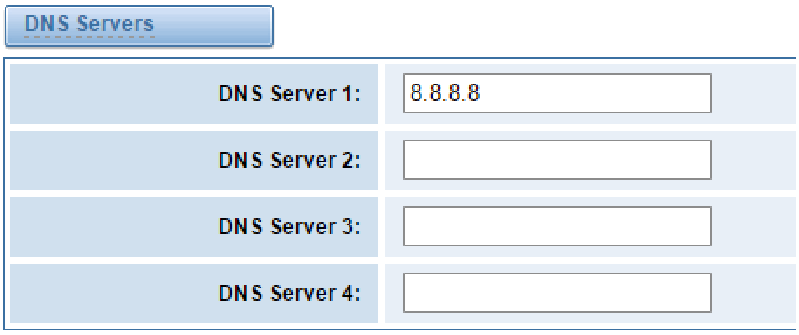
Table 6-1-1 Definition of WAN/LAN Settings

Options	**Definition**
Interface	The name of network interface.
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.

DNS Servers: A list of DNS IP address. Basically, this info is from your local network service provider. Note that please restart the gateway if you changed the DNS server.

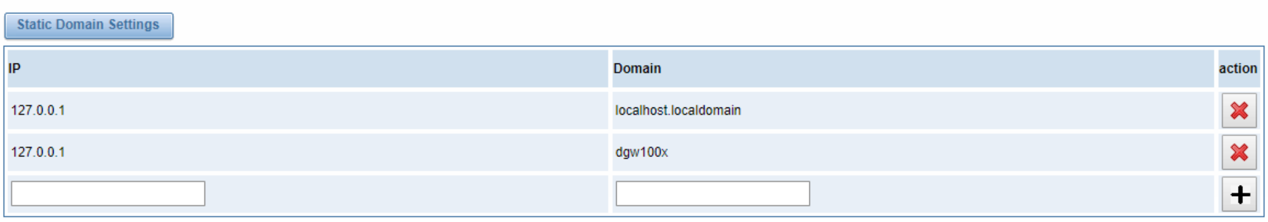
Figure 6-1-2 DNS Interface



6.2 DDNS Settings

You can set some static domain.

Figure 6-2-1 Static Domain Interface



Also, you can enable or disable DDNS (dynamic domain name server).

Figure 6-2-2 DDNS Interface

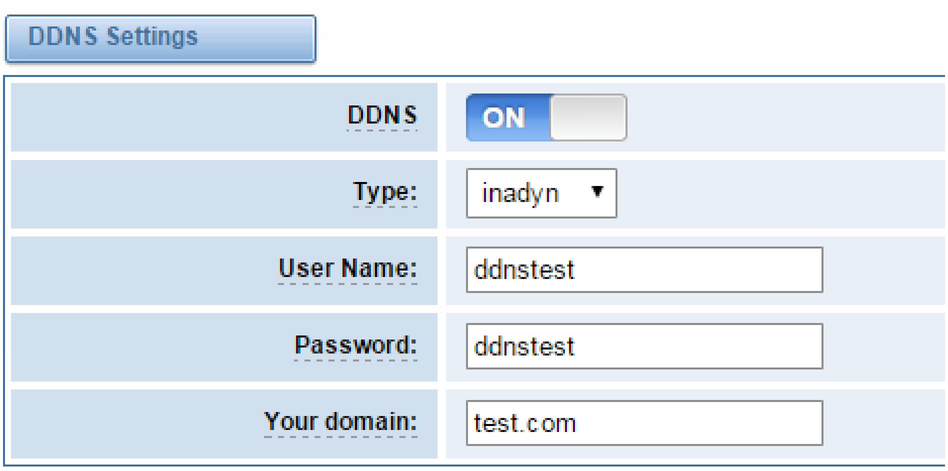


Table 6-2-1 Definition of DDNS Settings

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.

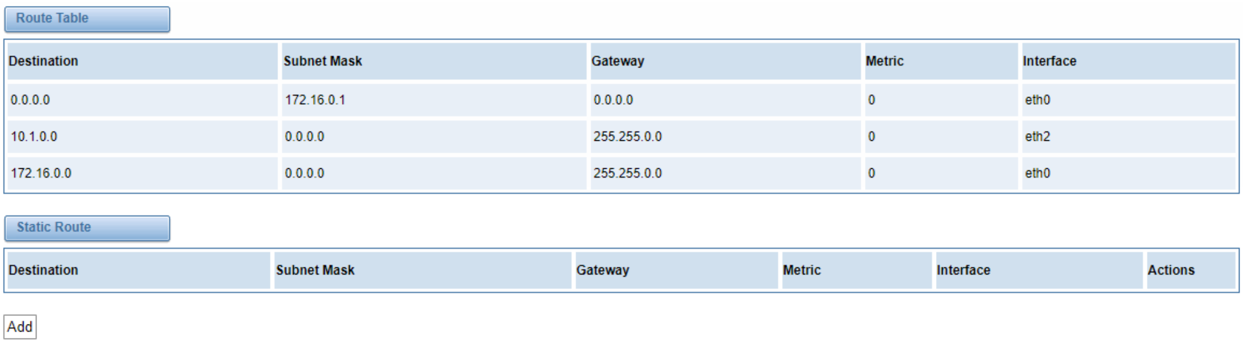
6.3 Toolkit

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

Figure 6-3-1 Network Connectivity Checking

6.4 Static Route Settings

Figure 6-4-1 Static Route Settings



Route Table

Destination	Subnet Mask	Gateway	Metric	Interface
0.0.0.0	172.16.0.1	0.0.0.0	0	eth0
10.1.0.0	0.0.0.0	255.255.0.0	0	eth2
172.16.0.0	0.0.0.0	255.255.0.0	0	eth0

Static Route

Destination	Subnet Mask	Gateway	Metric	Interface	Actions

Add

7 Advanced

7.1 Asterisk API

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

Figure 7-1-1 API Interface

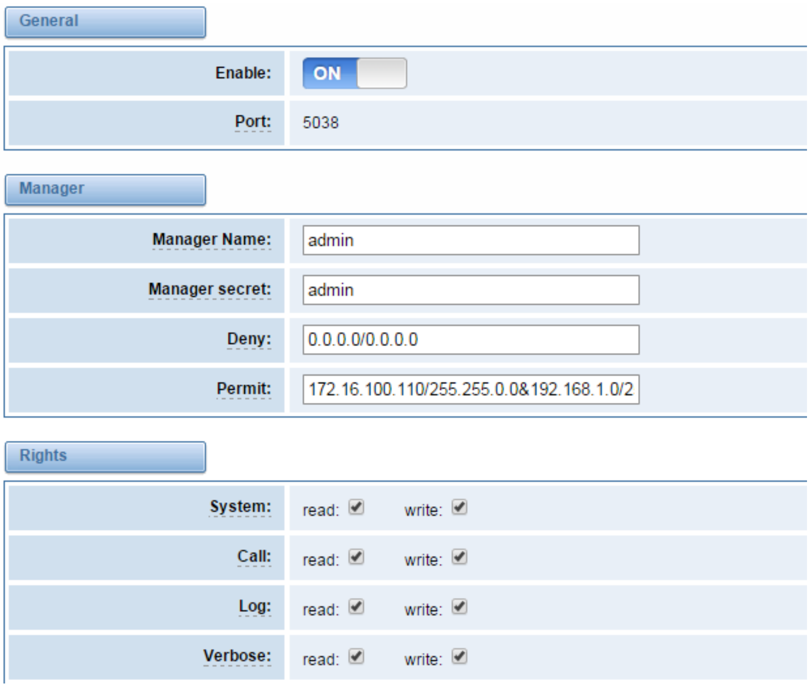


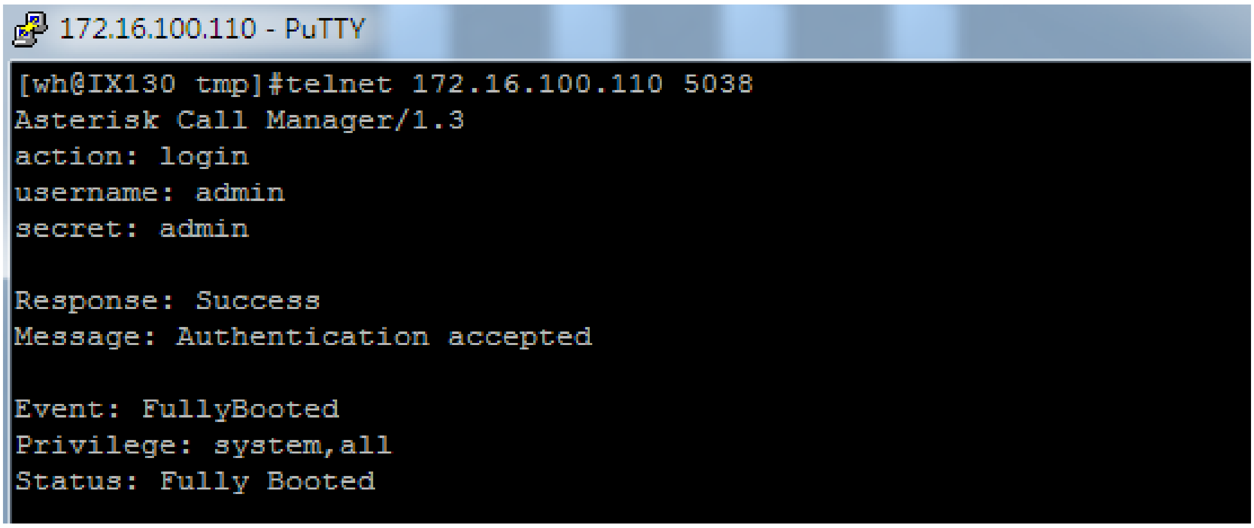
Table 7-1-1 Definition of Asterisk API

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



7.2 Asterisk CLI

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

Figure 7-2-1 Asterisk Command Interface

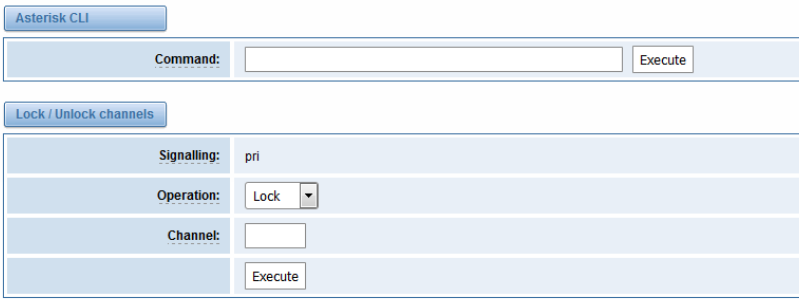


Table 7-2-1 Definition of Asterisk CLI

Options	Definition
Command	Type your Asterisk CLI commands here to check or debug your gateway.

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

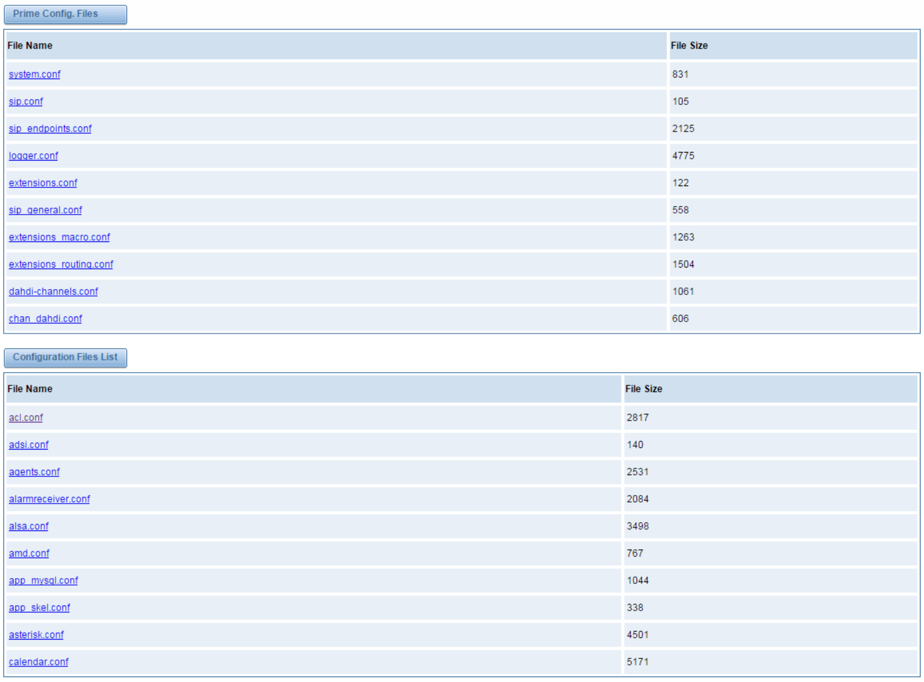
Table 7-2-2 Definition of Lock/unlock channels

Options	Definition
Signaling	Current signaling in use.
Operation	The advanced operations for lock and unlock channels.
Channel:	The channel to be lock or unlock.

7.3 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	831
sip.conf	105
sip_endpoints.conf	2125
logger.conf	4775
extensions.conf	122
sip_general.conf	558
extensions_macro.conf	1263
extensions_routing.conf	1504
dahdi-channels.conf	1061
chan_dahdi.conf	606

Configuration Files List

File Name	File Size
acl.conf	2817
adsi.conf	140
agents.conf	2531
alarmreceiver.conf	2084
alsa.conf	3498
amd.conf	767
app_mysql.conf	1044
app_skel.conf	338
asterisk.conf	4501
calendar.conf	5171

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

7.4 Auto Provisioning

Auto provisioning or auto-configuration is an easy, flexible and time-saving way to upgrade firmware and configurations for E1 gateways in mass deployment. With auto provisioning, all user information can be entered via the central ACS (Auto Configuration Server). ACS can be DHCP server or TFTP, HTTP and FTP server. It will not take effects immediately but in the next time system is power on. It could be postponed the execution of restart system also.

Note that system will not be upgrade the firmware and update configurations if the connection between ACS and gateway is disconnect.

7.4.1 Preparation

The following should be prepared before auto provisioning being applied.

- Enable the auto provisioning in gateway
- The ACS has been prepared
- The network between gateway and ACS is connected

7.4.2 Configuring gateway

Usually, the feature is disabled before being on sale. To activate the auto provisioning function, please follow the procedures as below.

Step 1 On the **ADVANCED**-> **Auto Provision interface**

Step 2 Enable the 'Enabled' option and select ACS. DHCP option 66 can be enabled if ACS has been working as DHCP server, otherwise please select protocol of provisioning and fill the value of '***Auto Config Server URL***'. Username and password may need to be filled in FTP/HTTP for the purpose of system safety. Do not forget to select Firmware upgrade, upgrade mode and fill the value of timeout, and click '***Save***'.

Step 3 Set interval of checking in **LOGS**->**System notice** then enable it, and click '***Save***'.

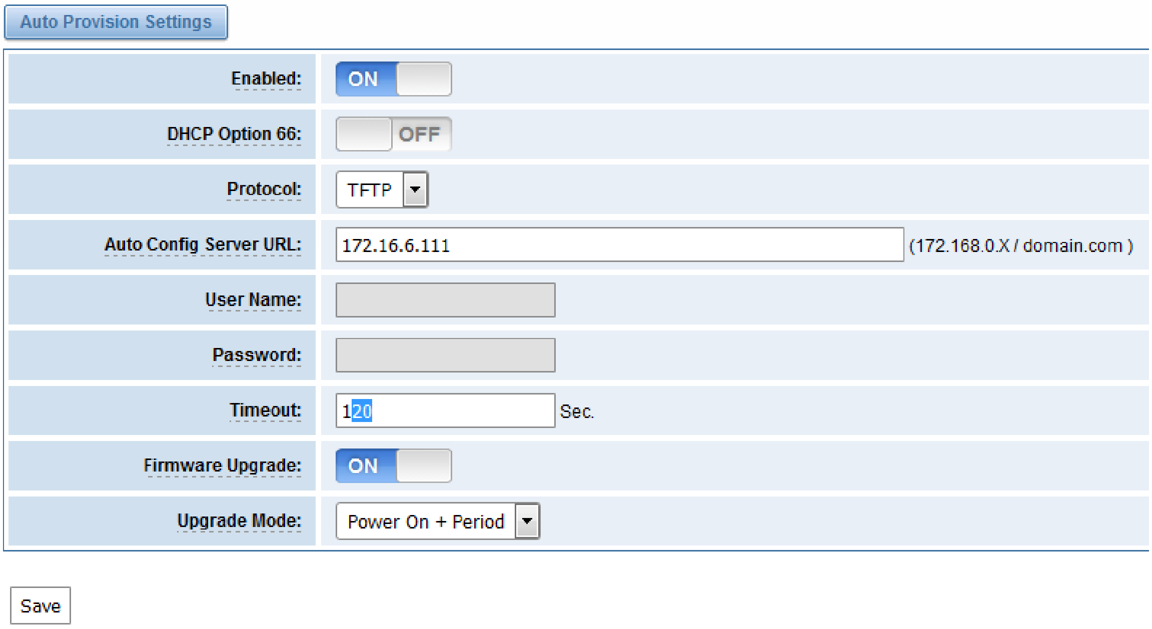
Table 7-4-1 Definition of Auto Provision

Options	**Definition**
Enabled	Whether to enable or disable Auto Provision.
DHCP Option 66	Get ACS server address from Option 66 via DHCP.
Protocol	Set protocol of connection.
Auto Config Server URL	The config server domain or IP address.
User Name	The account of downloading from ACS.
Password	The password of downloading from ACS.
Timeout	The max limit time for downloading firmware.
Firmware Upgrade	Enable/disable the mode of downloading firmware.
Upgrade Mode	Select upgrade time. Power: start upgrade configuration when Power on. Power + Period: Set the frequency of checking the latest configuration when gateway running.

Table 7-4-2 Definition of system notice

Options	**Definition**
Enable	Whether to enable or disable system notice.
Check Interval	When Upgrade Mode is set, this parameter specifies the interval of Checking.

Figure 7-4-1 Auto Provision interface



7.4.3 Configuring ACS

The Auto Configuration Server can be the one of TFTP, FTP and HTTP server. The ACS is used to store the firmware release and configurations files of the devices under management.

List the primary files in ACS download directory as table 7-4-3:

Table 7-4-3 Definition of ACS files

Options	**Definition**
DGW100x-current.bin	The firmware image.
common.conf	The wildcard configuration file for the whole gateway.
defconfig.tar.gz	The default(factory) configuration file.
EPC-{mac}.conf	The private configuration file for the specified gateway. Naming rules: “EPC-“ + “mac” +”.conf”. The naming prefix of “EPC-” stands for the private configuration file, “mac” is the physical address of network interface card but removed semicolon

	and ".conf" is the suffix. For example, the EPC-a0980501dbca.conf, 'a0980501dbca' is the MAC address (A0:98:05:01:DB:CA).

The format of common.conf , EPC-{mac}.conf and defconfig.tar.gz:

(1). Common.conf

[firmware]

FW_NAME=DGW100x-current.bin //Firmware image name

FW_MD5=b3603f3c3b5e7eb6326498640f151c79 //The md5 of firmware image

FW_VERSION=1.1.2 //Firmware version

[configs]

CONFIG_NAME=defconfig.tar.gz // default configuration file(compressed)

CONFIG_MD5KEY=2cd2dfbe52482405350816e3698cb530 // the md5 of default configuration file

(2).EPC-{mac}.conf

[dns]

DNS_SERVER1=8.8.8.8

DNS_SERVER2=8.8.4.4

DNS_SERVER3=

DNS_SERVER4=

[ntp]

NTP_SERVER1= 0.cn.pool.ntp.org

NTP_SERVER2= time.nist.gov

NTP_SERVER3= time.windows.com

[eth0]

ENABLE=yes

TYPE=static

DHCP=no

IPADDRESS=172.16.100.223

NETMASK=255.255.0.0

GATEWAY=172.16.0.1

[eth1]

ENABLE=yes

TYPE=static

DHCP=no

IPADDRESS=192.168.100.223

NETMASK=255.255.0.0

GATEWAY=192.168.0.1

[web_login]

username=admin

password=admin

(3). Defconfig.tar.gz

Figure 7-4-2 the overview of defconfig.tar.gz

```
[root@dgw100x /defconfig]#ls
config.info    group-         passwd         resolv.conf    sysconfig
fstab          hosts          passwd-        shadow         tmp
group          nsswitch.conf  profile        shadow-
[root@dgw100x /defconfig]#ls sysconfig/
NTP             hostname        nsswitch.conf   simple.script
asterisk        lighttpd        ntp.conf        syslog.conf
cron            logrotate.conf  php.ini         udhcpd.conf
dahdi           logrotate.d     redis.conf      zoneinfo
dnsmasq         network         services
[root@dgw100x /defconfig]#
```

7.4.4 Provisioning example

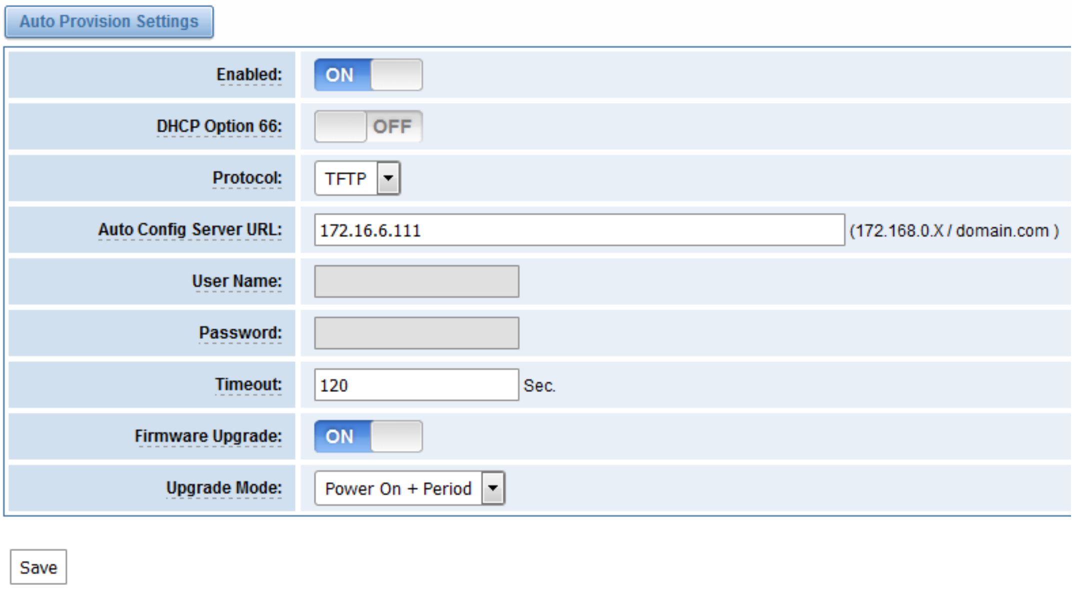
After auto provisioning is enabled, the gateway will visit the Auto Configuration Server and download the updated files periodically based on the timer ***Check Interval*** (**LOGS**->**System notice**). By default, the timer is set as every hour. System will receive a message from ACS, like figure 7-4-3, and the message will be display in the system notice (**LOGS**->**System Notice**).

Auto provisioning will not take effects immediately but in the next time system is power on. It could be postponed the execution of restart system also.

Now, an example of using Auto Provisioning will be given in the following.

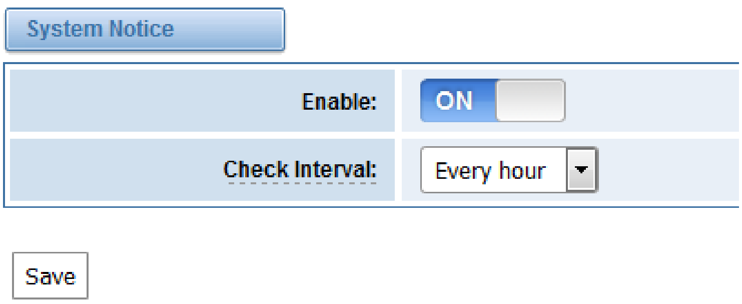
1. Activate the auto provision (TFTP) in **ADVANCED**-> **Auto Provision** like figure 7-4-4.

Figure 7-4-3 Auto provision settings



2. Enable the check interval in **LOGS**->**Log settings**->**System Notice** like figure 7-4-5.

Figure 7-4-4 Check interval setting



3. Configuring the ACS (Generate the md5 of firmware and defconfig.tar.gz)

- Copy the firmware, defconfig.tar.gz, common.conf and EPC-{mac}.conf to the working directory of TFTP server.

Figure 7-4-5 The working directory of TFTP server

Notice: The demo of E1 gateway mac address is A0:98:05:01:DB:CA (eth0), therefore the private configuration file is EPC-a0980501dbca.conf.

- Generate the md5 of firmware and defconfig.tar.gz. Then fill in common.conf and EPC-{mac}.config.

Figure 7-4-6 Generate the md5 of firmware and configuration

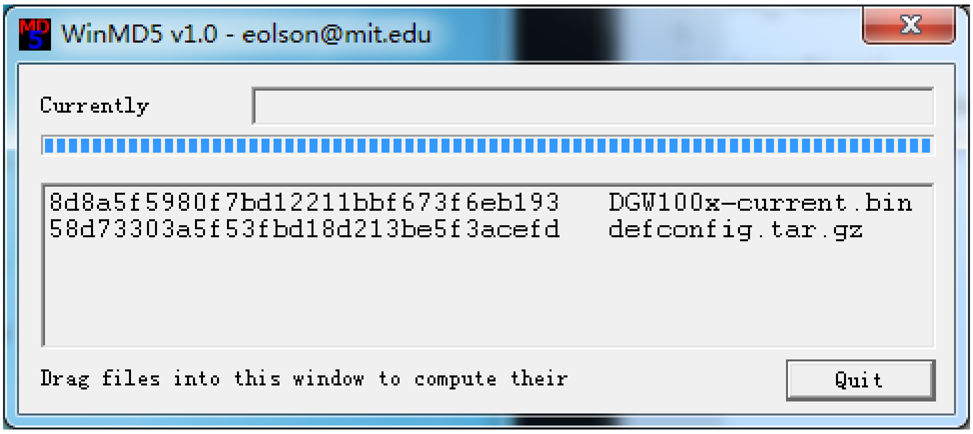


Figure 7-4-7 Common.conf

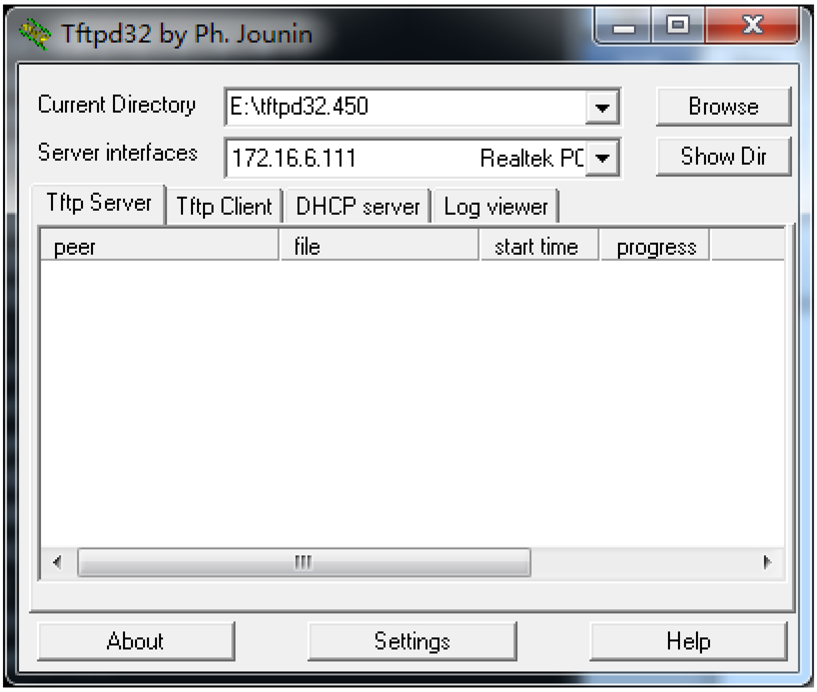
```
[root@localhost build]# cat common.conf
[firmware]
FW_NAME=DGW100x-current.bin
FW_MD5=8d8a5f5980f7bd12211bbf673f6eb193
FW_VERSION=1.1.2

[configs]
CONFIG_NAME=defconfig.tar.gz
CONFIG_MD5KEY=58d73303a5f53fbd18d213be5f3acefd
[root@localhost build]#
```

Figure 7-4-8 EPC- a0980501dbca.conf

```
[root@localhost build]# cat EPC-a0980501dbca.conf
[dns]
DNS_SERVER1=8.8.8.8
DNS_SERVER2=8.8.4.4
DNS_SERVER3=
DNS_SERVER4=
[ntp]
NTP_SERVER1= 0.cn.pool.ntp.org
NTP_SERVER2= time.nist.gov
NTP_SERVER3= time.windows.com
[eth0]
ENABLE=yes
TYPE=static
DHCP=no
IPADDRESS=172.16.100.223
NETMASK=255.255.0.0
GATEWAY=172.16.0.1
[eth1]
ENABLE=yes
TYPE=static
DHCP=no
IPADDRESS=192.168.100.223
NETMASK=255.255.0.0
GATEWAY=192.168.0.1
[web_login]
username=admin
password=admin
[root@localhost build]#
```

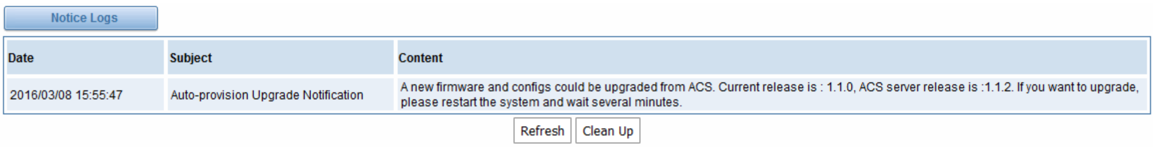
- Start TFTP service. Tftpd32.exe is a useful TFTP tools in windows7, then make sure TFTP server is select.

Figure 7-4-9 A demo TFTP server



4. The system will receive an auto provision message in web GUI.

Figure 7-4-10 System notice logs

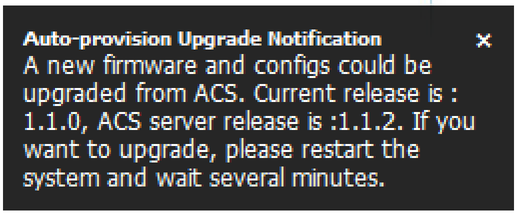


Notice Logs

Date	Subject	Content
2016/03/08 15:55:47	Auto-provision Upgrade Notification	A new firmware and configs could be upgraded from ACS. Current release is : 1.1.0, ACS server release is :1.1.2. If you want to upgrade, please restart the system and wait several minutes.

Refresh Clean Up

Figure 7-4-11 Auto provision upgrade notification



5. Restart the system. It will take about 3 minutes almost to download, upgrade Firmware and update configurations.

Figure 7-4-12 Downloading the firmware and configs

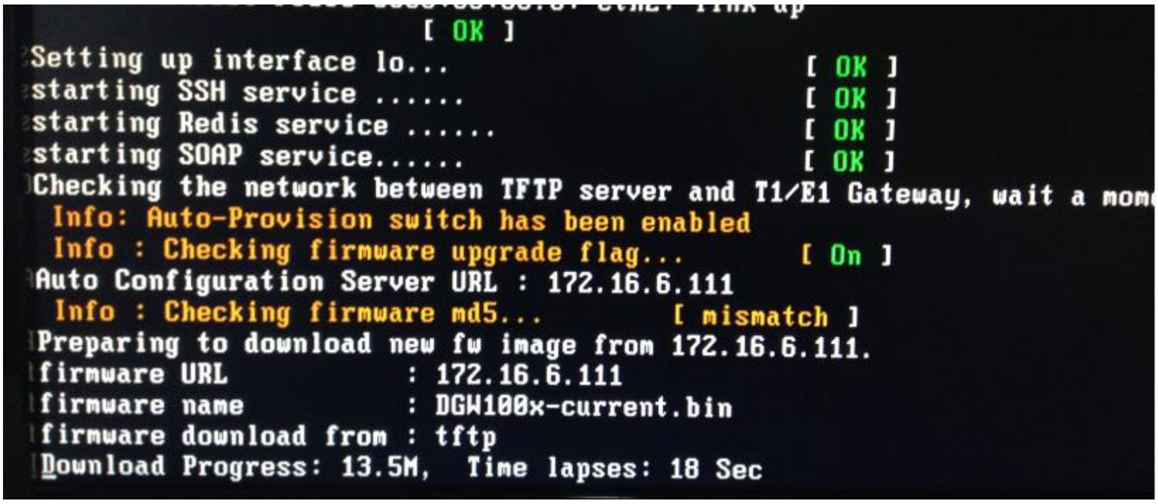
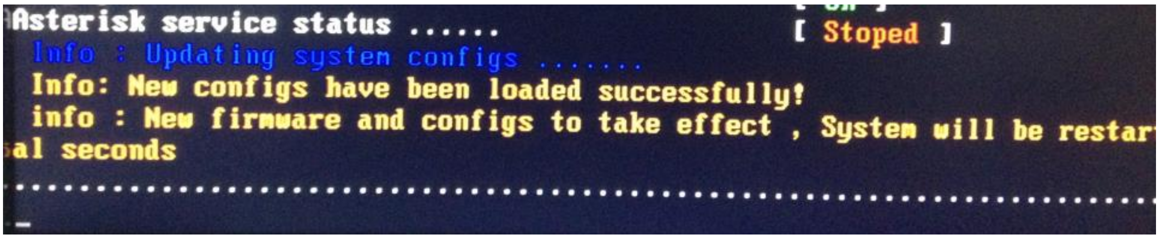


Figure 7-4-13 Applying the firmware and configs



7.5 SNMP

Simple Network Management Protocol (SNMP) is an application–layer protocol, which is used to manage and monitor network elements and exchange management information between network devices. By default, SNMP uses port 161 for communication.

Since the inception SNMP, it embraces three versions: v1, v2c and v3. V1 and v2c are the most implemented version of SNMP; v3 is target at the high security when compare to its older versions. The gateway support private SNMP MIBs (private enterprise number) to access.

7.5.1 Parameters in SNMP setting

Table 7-5-1 Definition of SNMP setting

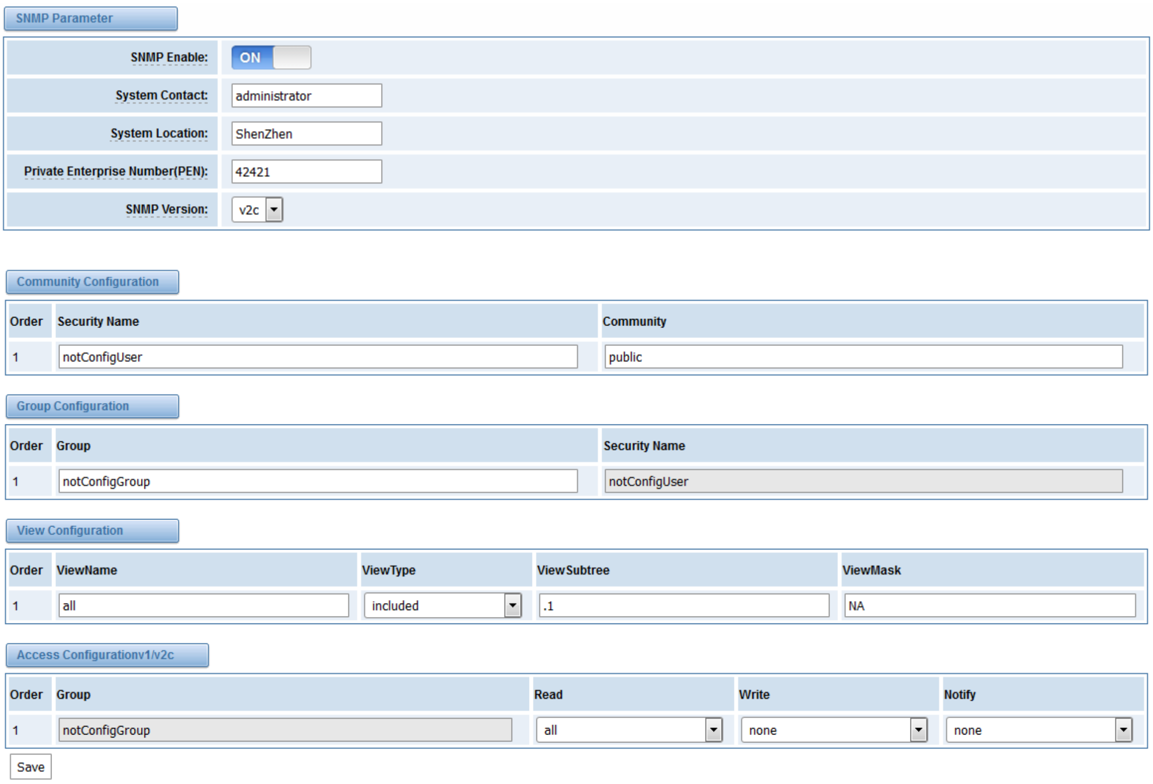
Options	**Definition**
SNMP Enable	Whether to enable SNMP.
System Contact	System contact information(optional).
System Location	The locale of system contact(optional).
Private Enterprise Number	The number is used for defining private SNMP MIBs which is assigned by Internet Assigned Numbers Authority (IANA). For more information, please access: http://pen.iana.org/pen/PenApplication.page.
SNMP Version	Select version of SNMP.
Community Configuration	Define a community name to security name.
Group Configuration	Define the security name to a group.
View Configuration	Set a view to let the group have rights to do.
Access Configuration	Grant the group can access to the view(read/write/notify).
User Configuration	Only exist in v3. Add a v3 account to SNMP. Notice that the length of auth password and privacy password are more than 8.

7.5.2 Activating SNMP

Usually, the feature is disabled by default. To activate the SNMP feature, please follow the Figure 7-5-1.

The Interface is in the **ADVANCED->SNMP**. System contact, location and private enterprise number are optional. Figure 7-5-1 is the SNMP setting interface.

Figure 7-5-1 Activating the SNMP



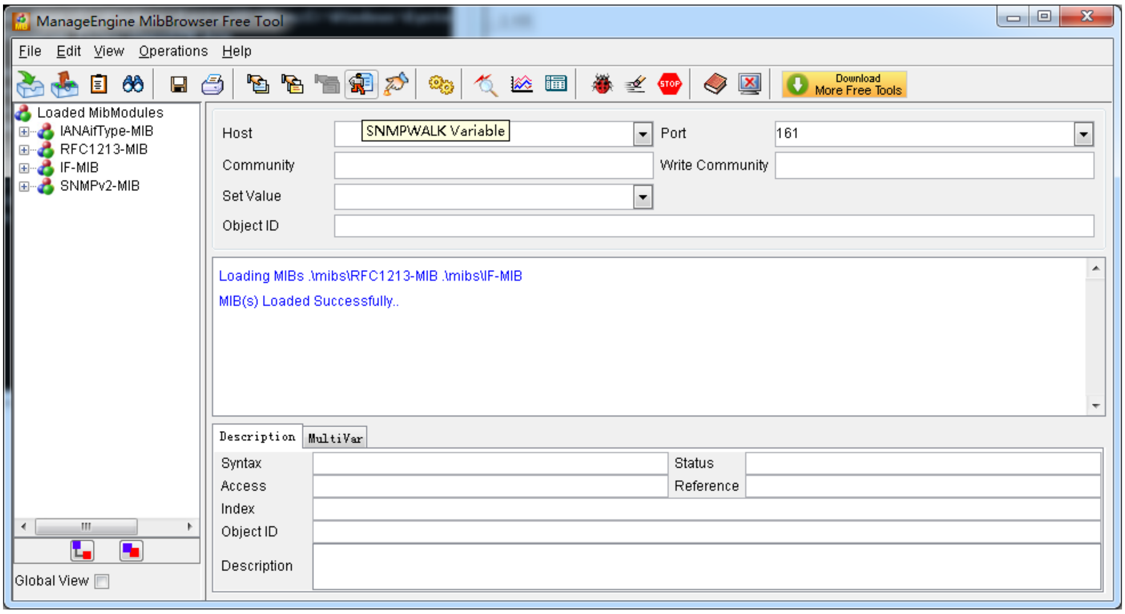
Note: Do not forget to click '***Save***' to take effect. After configuration, The SNMP feature is activated immediately.

7.5.3 Verify SNMP

A powerful, indispensable and easy-to-use MIB browser is convenient for engineer/manager to manage SNMP enabled network devices and applications. In this session, Manage Engine MIB browser is selected. It allows user to issue SNMP requests to retrieve agent's data, or make changes to the agent. It is free tool for Windows, Mac and Linux.

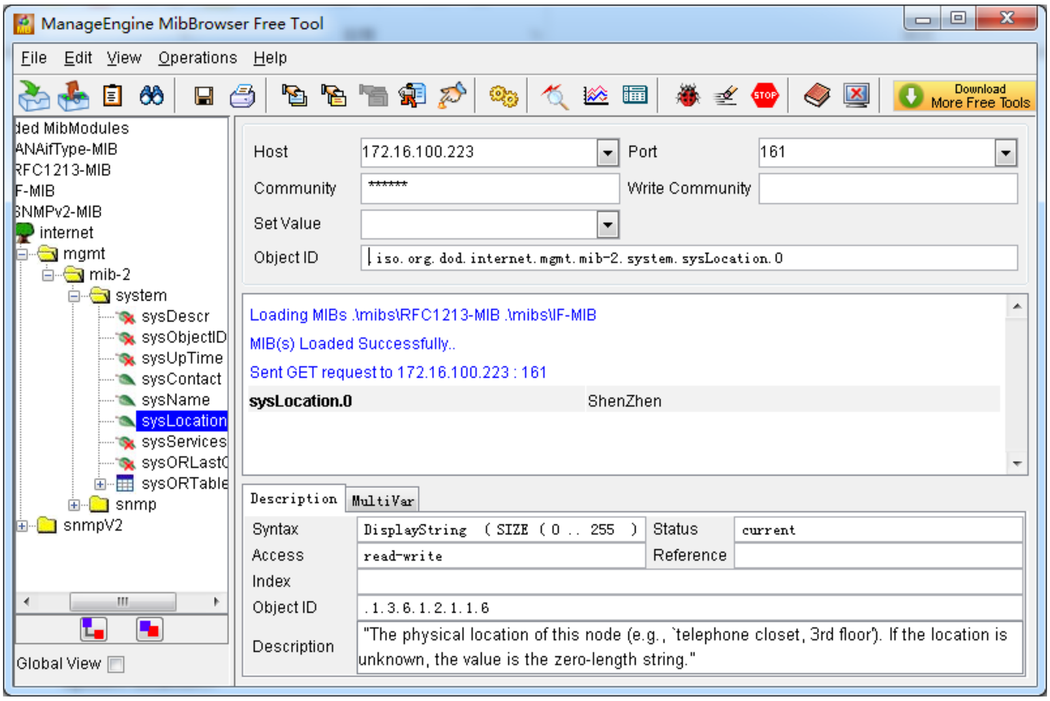
(1). Get SNMP parameters via SNMP MIB browser. It's supposed that Manage Engine MIB browser is installed perfectly. Figure 7-5-2 is the main interface of Manage Engine MIB browser.

Figure 7-5-2 Manage Engine MIB browser



And the field of ***Host***, ***Port*** and ***Community*** are filled with **172.16.100.223**, **161** and **public** respectively. Object ID is the node of SNMP MIBs, e.g. ".1.3.6.1.2.1.1.6.0" is system location and ".1.3.6.1.2.1.1.1.0" is system description.

Figure 7-5-3 Get system location

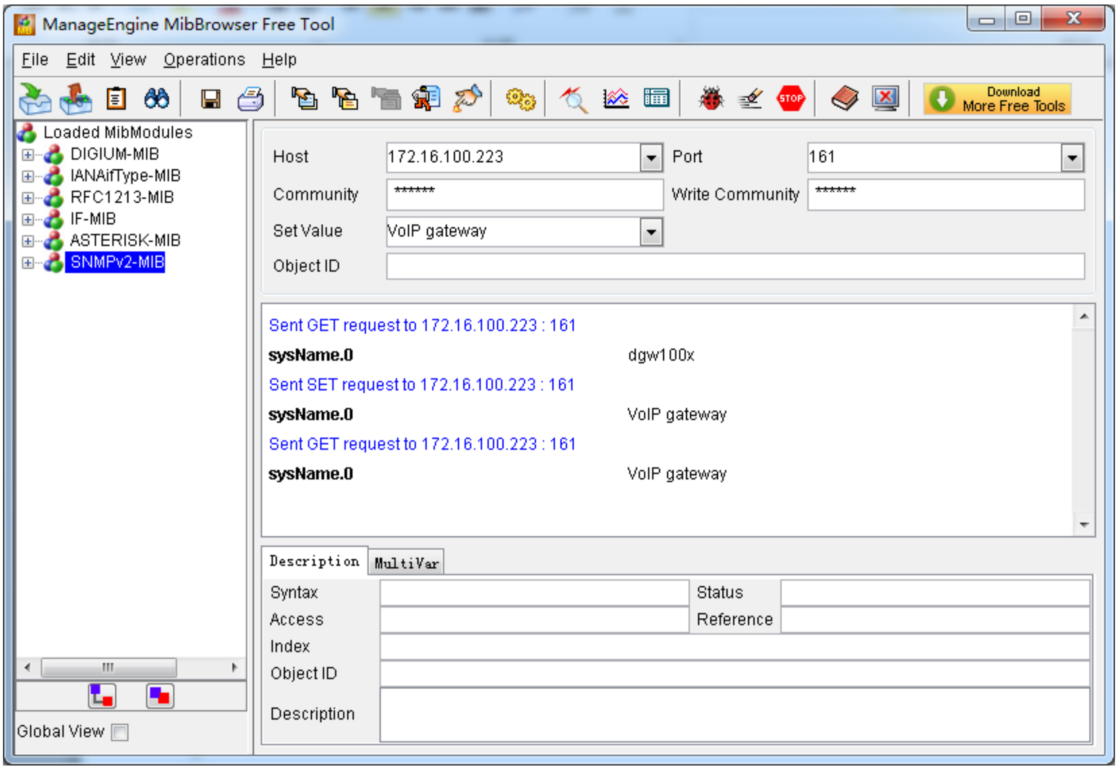


After the rest field has been filled, then verify it. Click **Operations**->**GET** to get the value of system location and it returns the value which we just set.

(2). Set SNMP parameters via SNMP MIB browser. For example, set the system name. system name is “dgw100x” by default, then set it as “VoIP gateway”. See figure 7-5-4.

- Click **Operations**->**GET** to attain the current system name.
- Fill the field of ***Set Value*** with “VoIP gateway”.
- Click **Operations**->**SET** to set the system name.
- Click **Operations**->**GET** to attain the modified system name.

Figure 7-5-4 Set system name



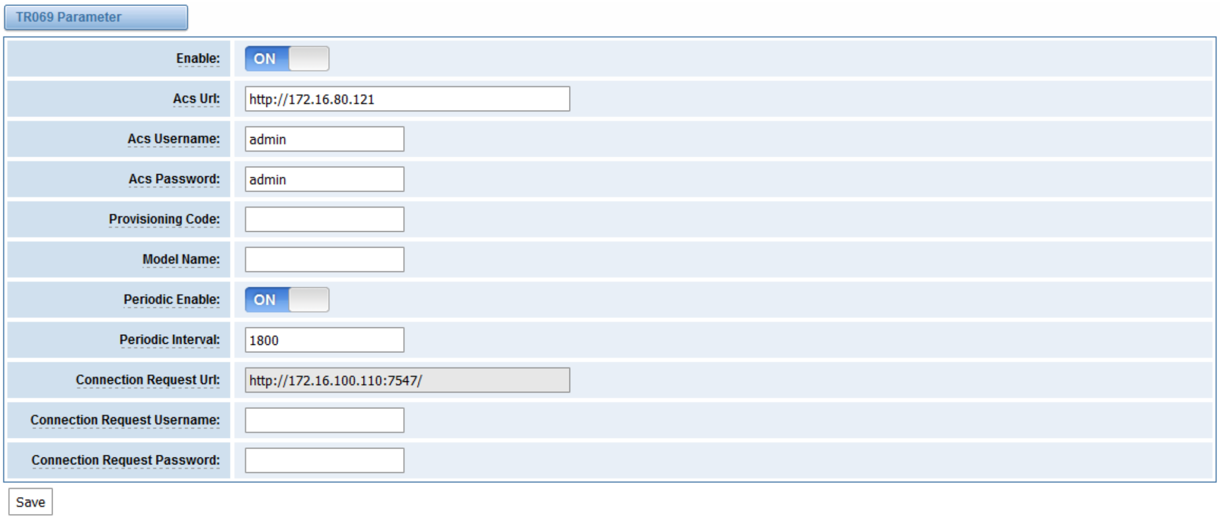
7.6 TR069

TR069 is a remote management solution which offers a single interface to manage the ACS and automate the deployment and support of data, voice and video services, thereby reducing operation and support costs, while enhancing customer satisfaction. Its user-friendly interface covers the entire service lifecycle, from centralized remote provisioning of services, to inventory management, group updates, monitoring, event triggering, and support automation. Figure 7-6-1 is TR069 configuration interface and table 7-6-1 is its definition.

Table 7-6-1 Definition of TR069 configuration interface

Options	**Definition**
Acs Url	Specify the URL of the ACS
Acs Username	Specify the user name to be used by the device to authenticate with the ACS.
Acs Password	Specify the password to be used by the device to authenticate with the file server
Provisioning Code	Information of the device vendor, which may be used to indicate the primary service provider and other provisioning information to the ACS. It can be numbers or English letters.
Model Name	A brief description of the interface type or name. It is a string of characters.
Periodic Enable	Used to specify whether to periodically report to the ACS.
Periodic Interval	The interval for reporting to the ACS.
Connection Request Url	The address used for the ACS to connect back to the device.
Connection Request Username	The account used for the ACS to connect back to the device, for example, admin.
Connection Request Password	The password used for the network management server to connect back to the device.

Figure 7-6-1 TR069 configuration interface



7.7 Network Capture

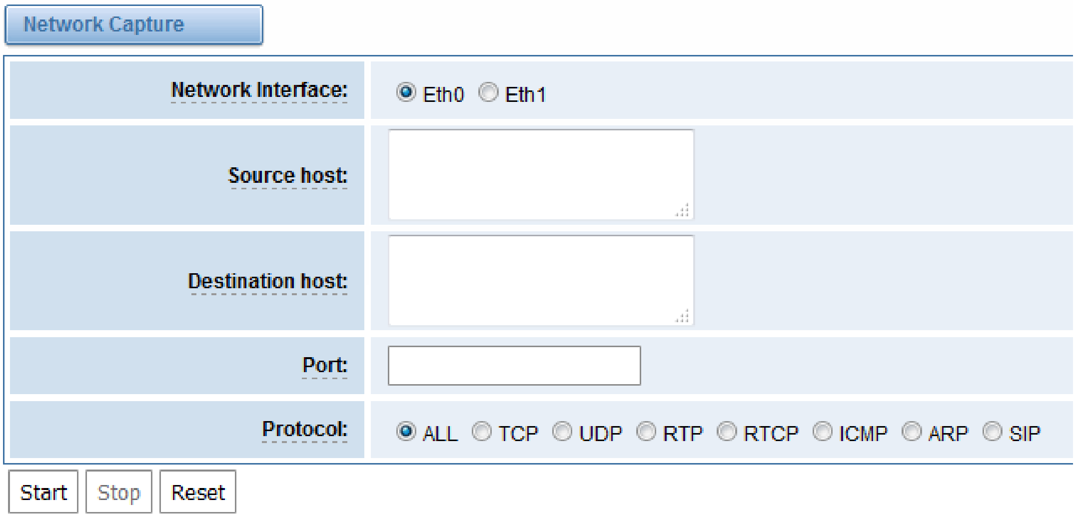
The gateway has been supplied a network packets capture in the web for ease of user to analysis, capture and monitor the gateway's network status, RTP flows, protocol analysis and so on.

Table 7-7-1 Definition of Network capture

Options	**Definition**
Network Interface	Specify which interface to be capture packets from. 'All' means capture packets from all interfaces.
Source host	Specify which source host IP address to listen for.
Destination host	Specify which destination host IP address to listen for.
Port	To specify a port that is either source or destination direction.
Protocol	To specify which protocol to be captured, 'All' stands for capture multi-protocols, the SIP default port is 5060, If you are using a different port, please amend it.

The interface is in **ADVANCED->Network Capture**.

Figure 7-7-1 Network capture interface



Moreover, user can capture SS7 signal and record port.

Figure 7-7-2 Signal Capture interface

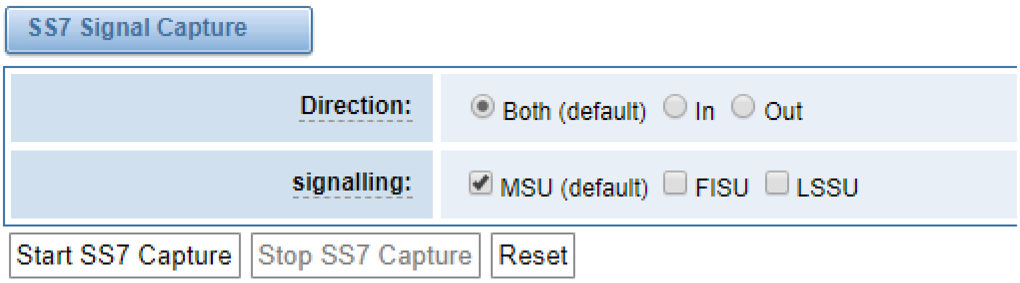
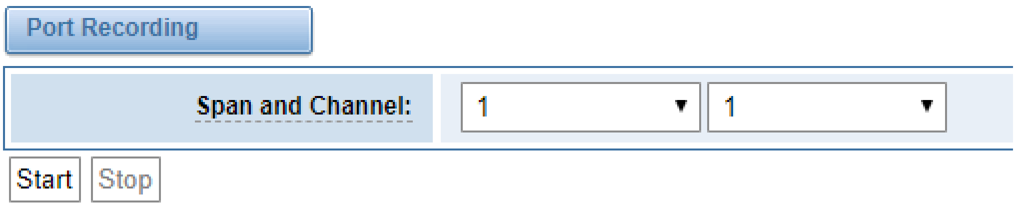


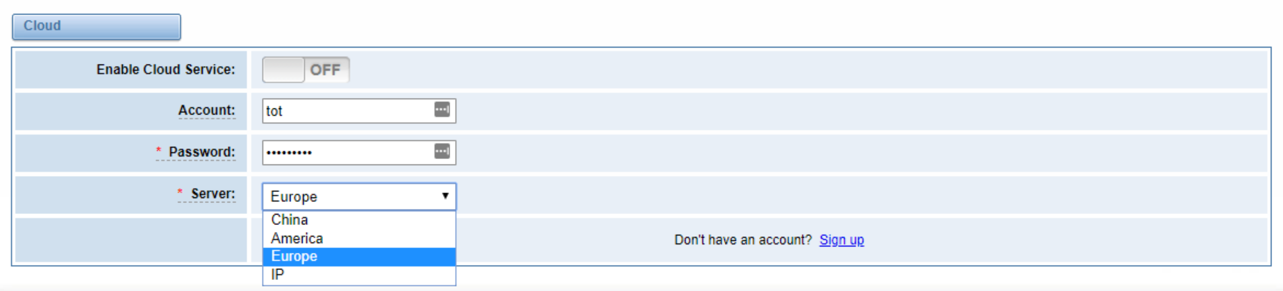
Figure 7-7-3 Port Recording interface



7.8 Cloud

OpenVox E1/T1 gateways support OpenVox Cloud Management.

Figure 7-8-1 Cloud interface



If your device is connected to the cloud management, the SSH and the web pages of the gateway can be accessed through the cloud management, and it can be monitored whether the device is connected to the cloud management platform. On the cloud management platform, you can also count your device model, quantity, distribution area, and so on which can provide you with efficient and excellent service and experience.

Table 7-8-1 Definition of Cloud Management

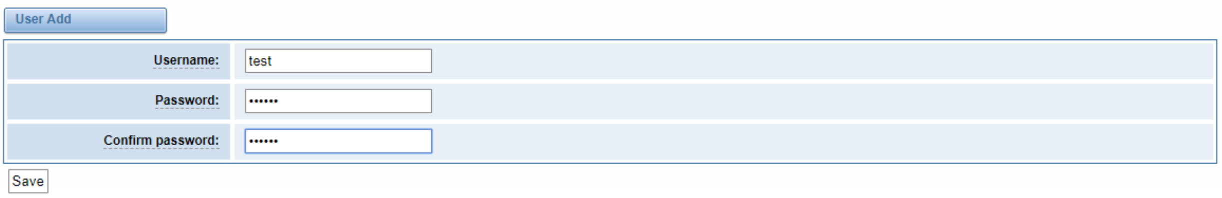
Options	**Definition**
Enable Cloud Service	Turn on/off the cloud management.
Choose Service	Currently supports three servers, China, the United States and Europe.
Account	Registered account or email on the cloud management platform.
Password	The password of the account registered on the cloud management platform.
Connection Status	Whether currently connected to the cloud management platform or not.

8 User

DGW Series T1/E1 Gateway allows you to create users and modify the permissions of users accessing the web interface.

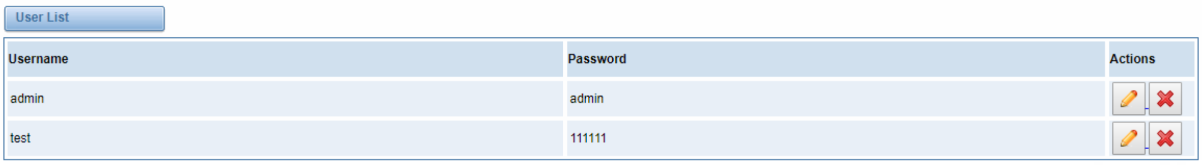
8.1 User Add

Figure 8-1-1 User Add interface



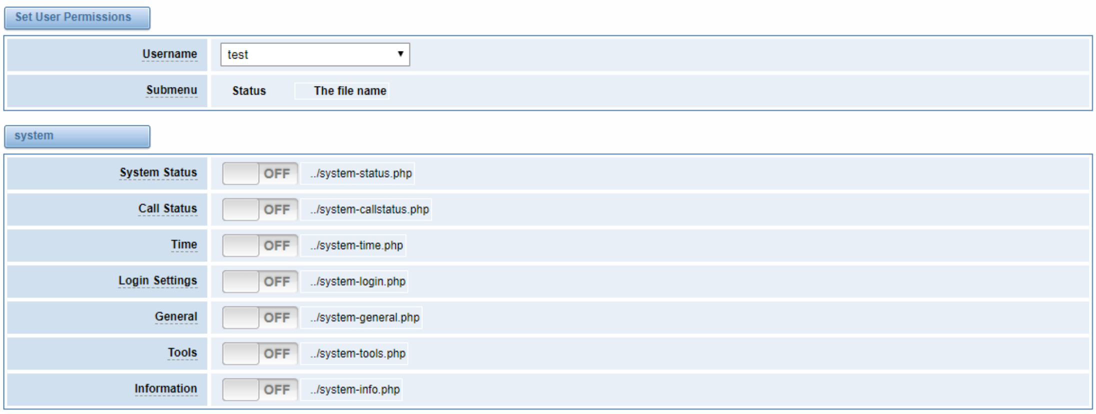
8.2 User List

Figure 8-2-1 User List interface



8.3 Permissions

Figure 8-3-1 Permissions

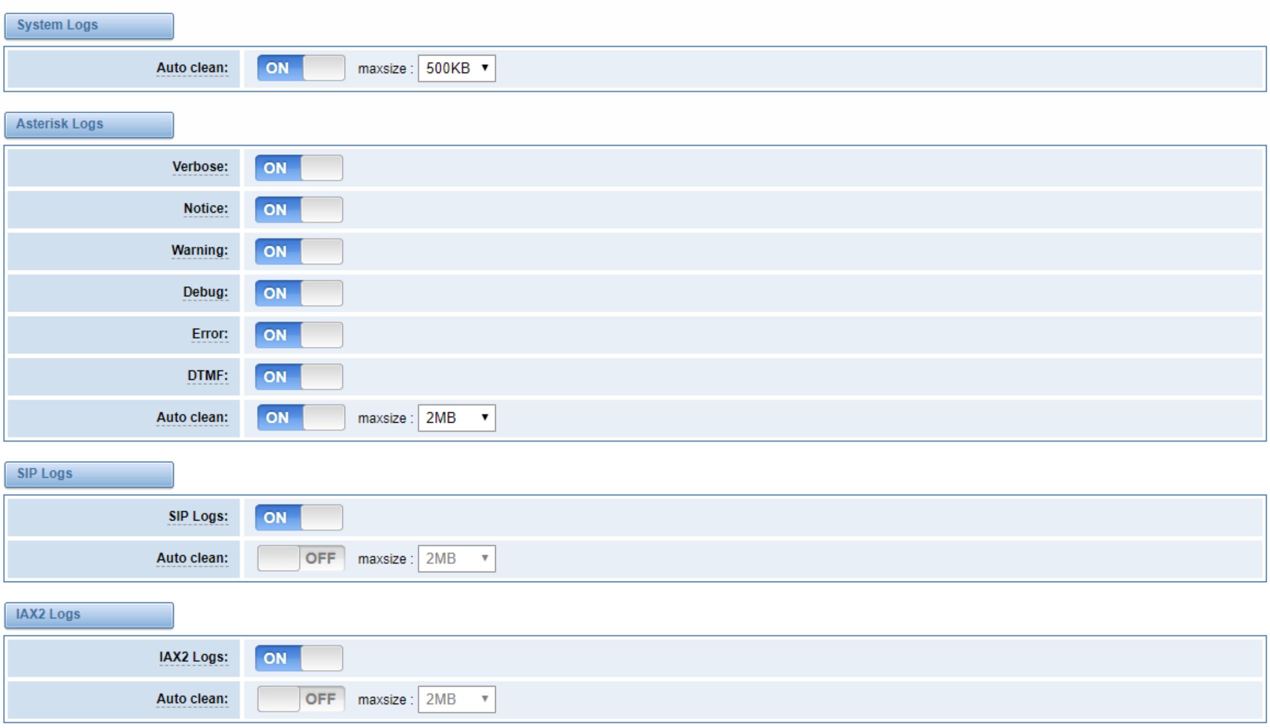


9 Logs

9.1 Log Settings

On the "**Log Settings**" page, you should set the related logs on to scan the responding logs page. For example, set "**SIP Logs**" on like the following, then you can turn to "**SIP**" page for sip logs, otherwise, sip logs are unavailable. And the same with other log pages.

Figure 9-1-1 Logs Settings



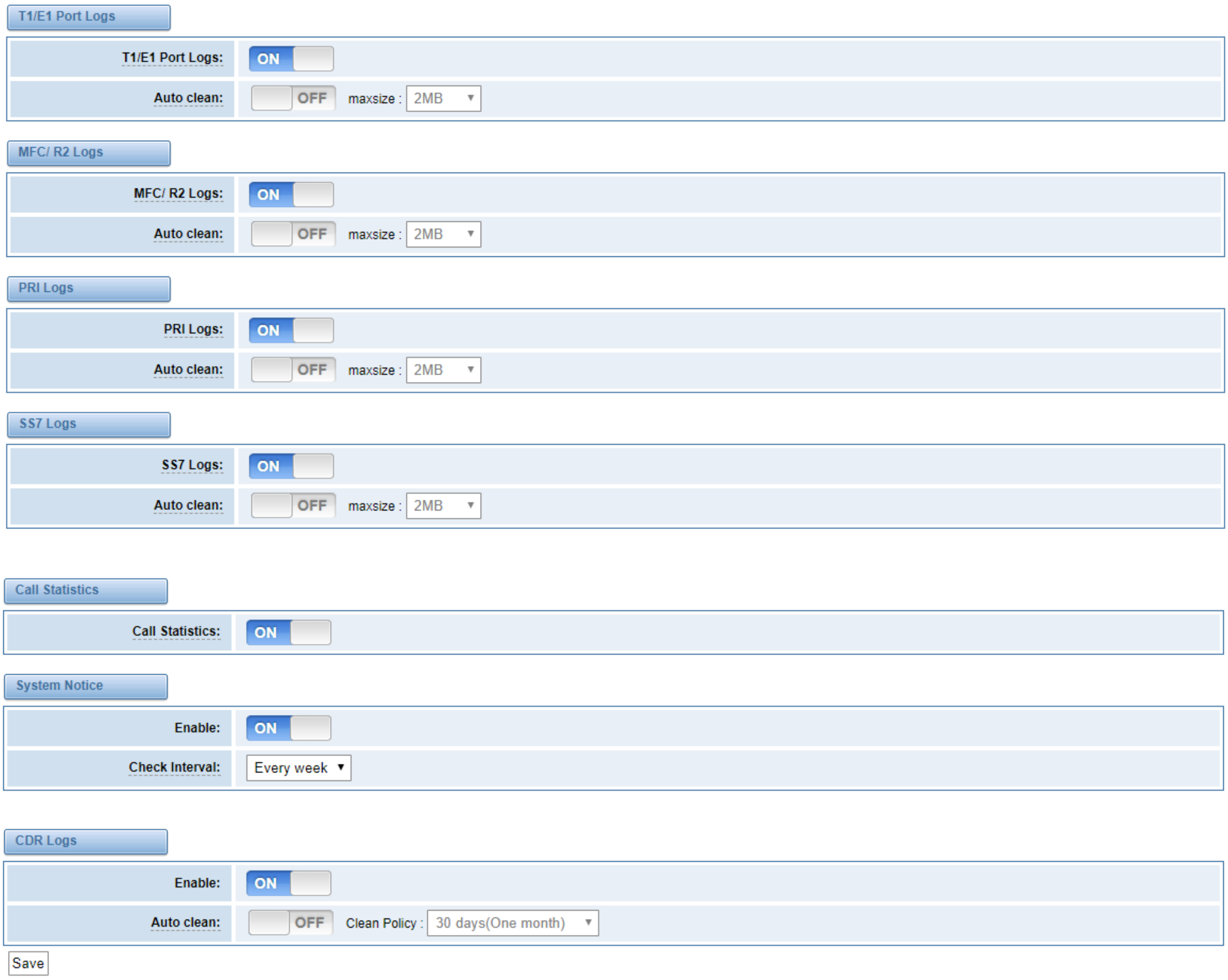


Figure 9-1-2 System Logs Output

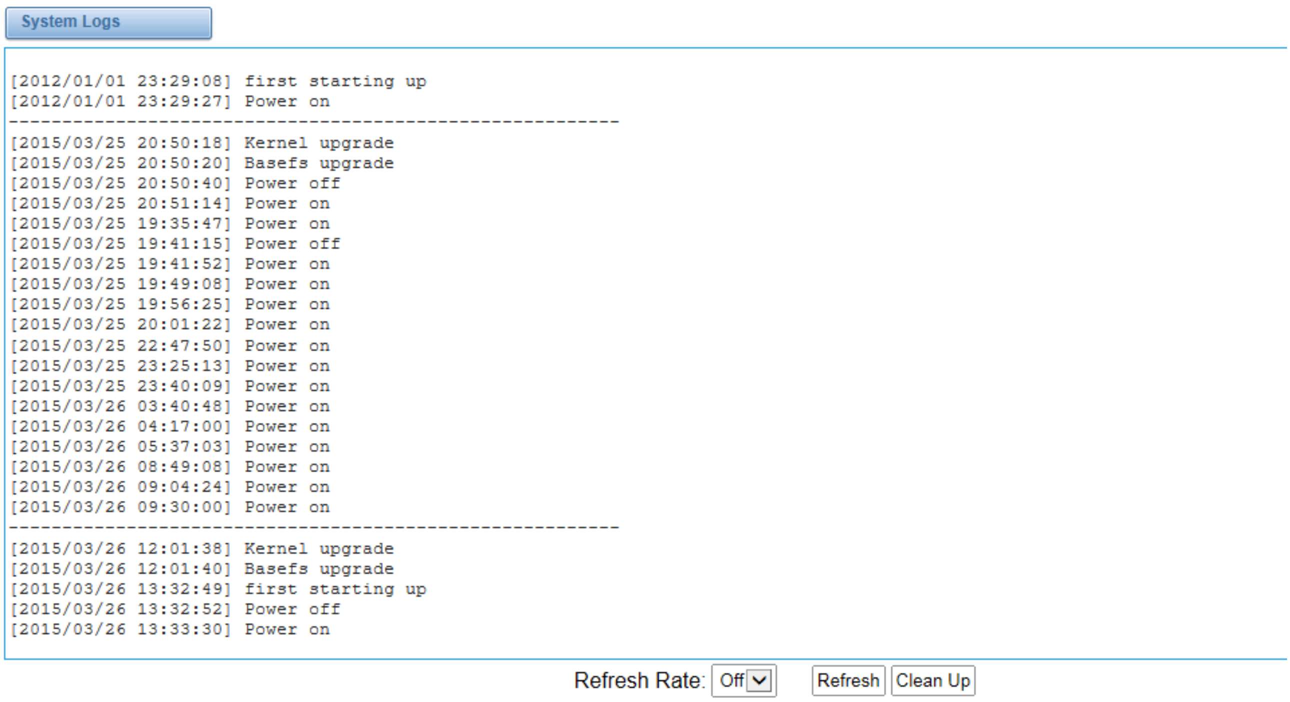


Table 9-1-1 Definition of Logs

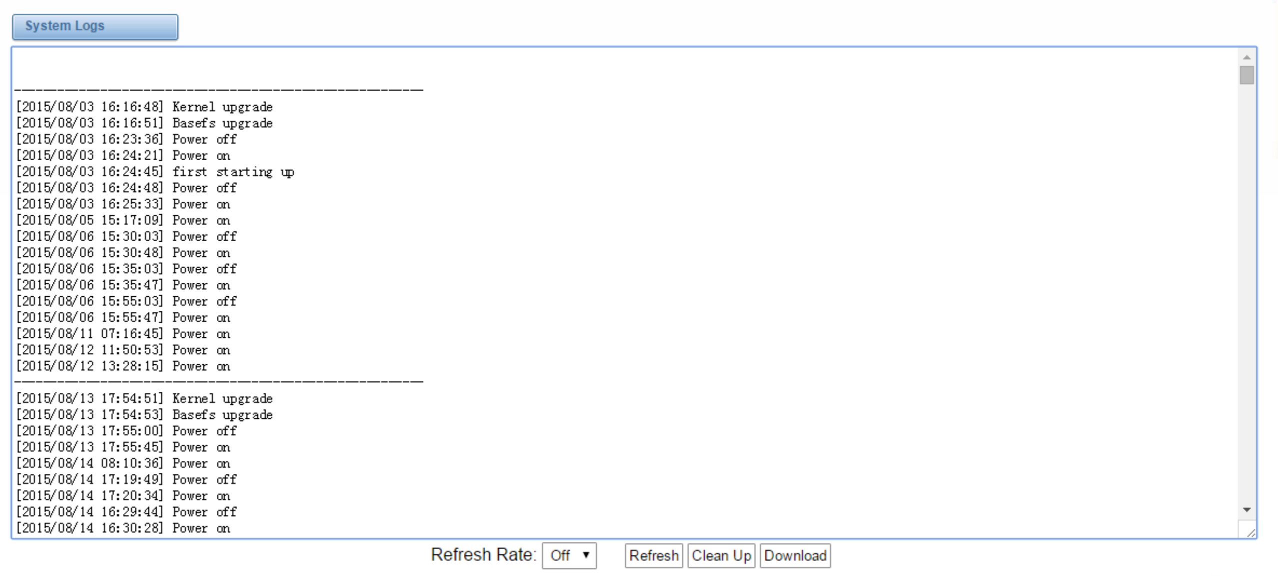
Options	**Definition**
Auto clean (System Logs)	**Switch on:** when the size of log file reaches the max size, The system will cut a half of the file. New logs will be retained. **Switch off:** logs will remain, and the file size will increase gradually. default on, default size=1MB.
Verbose	Asterisk 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=2 MB.
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=2 MB.
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=2 MB.
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=2 MB.
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=2 MB.
.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=2 MB.
Call Statistics	Whether enable or disable Call Statistics.
System Notice	The notification from system firmware upgrade and Auto provisioning.

9.2 System log

System log record every time power on, power off and firmware upgrade information.

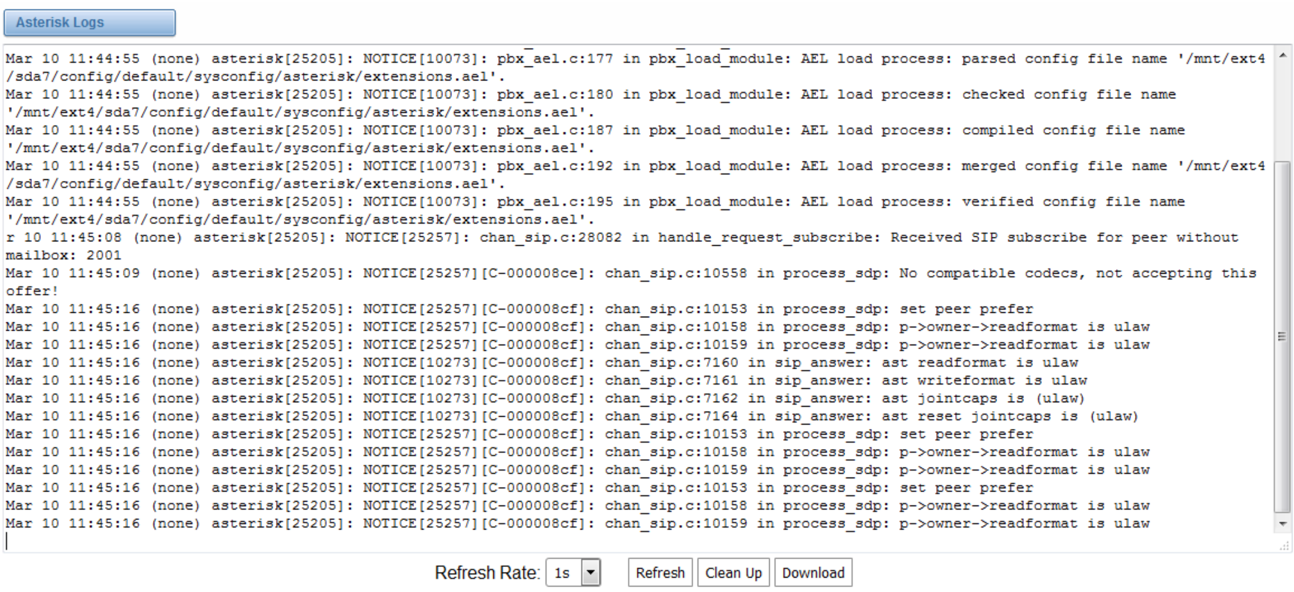
Figure 9-2-1 System Log



9.3 Asterisk logs

On the pages of “**Asterisk**”, “**SIP**”, “**IAX2**”, “**SS7**”, “**PRI**” and “**MFC/R2**”, there owns the some functions—Displays the log by port, refresh regularly and log download.

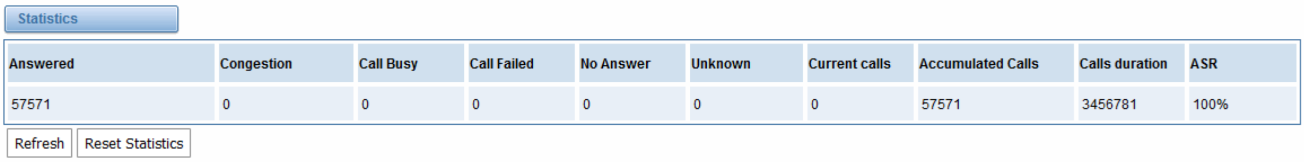
Figure 9-3-1 Asterisk Log



9.4 Call Statistics

The figure of call statistics, you'll find "**Answered**", "**congestion**", "**Call busy**", "**Call failed**", "**No answer**", "**Current calls**", "**accumulated calls**", "**Calls duration**" and "**ASR**". "**ASR**" stands for Answer Seizure Ratio. "**Calls duration**" will count the whole calls in the gateway. The call statistics will be saved before power off. It will be loaded after power on. It can be refreshed by itself. You can reset the statistics manually.

Figure 9-4-1 Call Statistics



Statistics

Answered	Congestion	Call Busy	Call Failed	No Answer	Unknown	Current calls	Accumulated Calls	Calls duration	ASR
57571	0	0	0	0	0	0	57571	3456781	100%

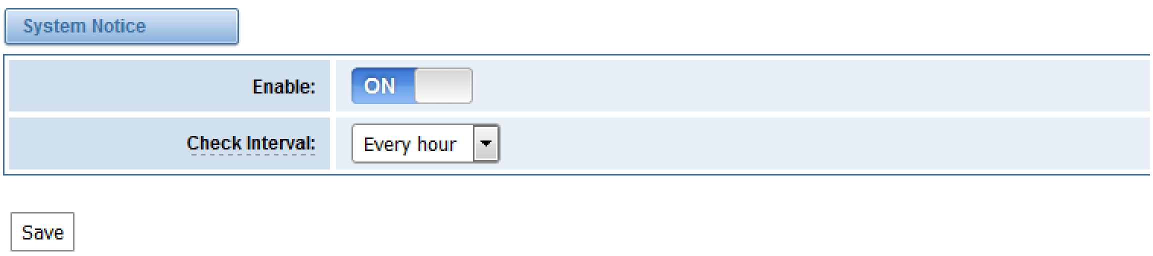
Refresh Reset Statistics

Note: Do not forget to enable call statistics in "**Log Setting**" if you want to statistics the calls.

9.5 System Notice

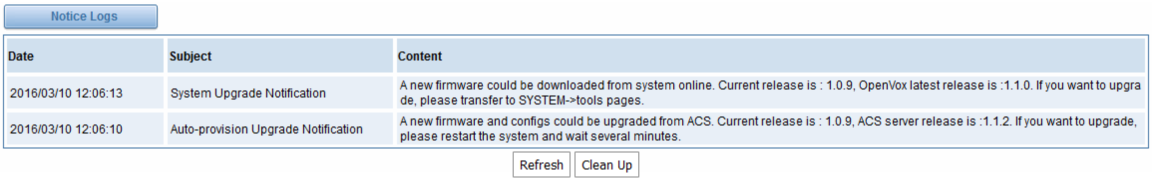
The system notice could be generated by system to inform the network manager of what is going on if it has been enabled. Firmware upgrade messages from official website and auto provisioning messages from ACS are main notice right now. And at first, enable the system notice function in "**Log Setting**" page like figure 9-5-1.

Figure 9-5-1 enable system notice function



After about an hour, a system message is received in the web like 9-5-2.

Figure 9-5-2 enable system notice function



Notice Logs

Date	Subject	Content
2016/03/10 12:06:13	System Upgrade Notification	A new firmware could be downloaded from system online. Current release is : 1.0.9, OpenVox latest release is :1.1.0. If you want to upgrade, please transfer to SYSTEM->tools pages.
2016/03/10 12:06:10	Auto-provision Upgrade Notification	A new firmware and configs could be upgraded from ACS. Current release is : 1.0.9, ACS server release is :1.1.2. If you want to upgrade, please restart the system and wait several minutes.

Refresh Clean Up

Note: Do not forget to enable system notice and check interval in "**Log Setting**" if you want to receive system messages.