

DGW-L1User Manual



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Full text

The overall layout adjustment



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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





Product Appearance

The picture below is appearance of DGW-L1.





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 re	estore the d	evice.						
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						

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	Power Status indicator	Groop	On: Power is on						
	FOWEI Status multator	Uleen	Off: Power is off						
5 VGA	VGA monitor connector	A monitor connector							
6 Eth1	Network interface	etwork interface							
7 Eth0	Network interface								
8 USB	USB interface								
9 DC-12v	Power supply								

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
- 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



/indows 安全				
位于 T1/E1 Ga	teway 的服务器:	172.16.100.	1 要求用户名利	口密码。
	admin ●●●●● 同 记住我的凭望	倨		
			确定	取消



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

Interface Status																						
Port1																						
OK Down OREload																						
Channels Status																						
Port 1 2 3 4 5	6 7 8	9 10	11 12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
1 🕖 🕑 🖯 🗑				0	0	0	\bigcirc	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
😑 Idle 🛑 Busy 🛑 Local Blocke	d 😝 Remode Block	ked 😑 Local (Jnavailable	e 😑 F	Remode	e Unav	ailable	e 💮 (Disable	e 🔵 s	chanr	nel										
SIP Information																						
Endpoint Name	User Name		Host	Host				Re	gistrat	ion		s	Status									
9000	9000		172.16	.80.103				clie	ent			c	OK (1 ms)									
8001	8001		(Unspe	(Unspecified) server					U	UNKNOWN												
8002	8002		(Unspe	cified)				ser	server UNKNOWN													
IAX2 Information																						
Endpoint Name	User Name		Host			Registration Status																
Routing Information																						
Rule Name	From		То					Ru	es													
out sip-9000 Port-1						Callee_Dial_pattern +i[](- +)] Caller_Dial_pattern +i[](- +)]																
8001 sip-8001 Port-1							Ca + [] Ca + []	llee_D (- +) ller_Di (- +)	ial_pat al_patt	tern ærn												
Network Information																						

Name	MAC Address	IP Address	Mask	Gateway	RX Packets	TX Packets
eth0	A0:98:05:01:E0:69	172.16.100.88	255.255.0.0	172.16.0.1	47157848	39873915
eth1	A0:98:05:01:E0:6A	192.168.100.1	255.255.255.0	192.168.0.1	0	0

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"
Signaling Status	means it is unavailable; "UP" means the port is available.

Time

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.

Table 2-2-1Description of Time Settings

For example, you can configure like this:

Figure 2-2-1 Time Settings

Time Settings	
System Time:	2016-1-4 09:25:09
Time Zone:	Shanghai 🔹
POSIX TZ String:	CST-8
NTP Server 1:	0.cn.pool.ntp.org
NTP Server 2:	time.nist.gov
NTP Server 3:	time.windows.com
Auto-Sync from NTP:	ON
Sync from NTP Sync from Client	

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.

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.

Table 2-3-1Description of Login Settings

Figure 2-3-1 Login Settings

Web Login Settings	
User Name:	123456
Password:	
Confirm Password:	
Login Mode:	http and https
Port:	80
SSH Login Settings	
Enable:	ON
User Name:	super
Password:	admin
Port:	12345
Save	

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

Language Settings	
Language:	English •
Advanced:	
Language Debug:	TURN ON TURN OFF
Download:	Download selected language package.
Delete:	Delete selected language.
Add New Language:	New language Package: 选择文件 未选择任何文件 Add

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

Scheduled Reboot	
Enabled:	ON
Reboot Type:	By Day
Running Time:	By Week By Month
Save	By Running Time

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.

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Reboot Tools

You can choose system reboot and Asterisk reboot separately.

	Figure 2-6-1 Reboot Proi	mpt	
TI/EI DATEWAY SYSTEM I TI/E'	VOIP ROUTING NETWO Login Settings General Auto Provision	RK ADVANCED LOG	S
SYSTEM	Are you sure to reboot your gateway now? You will lose all data in memory!	ation	19
Reboot the gateway and all the current calls will be dropped.	确定取消		System Reboot
Reboot the asterisk and all the current calls will be dropped.			Asterisk Reboot
Update Firmware			

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

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.

Table 2-6-1 Instruction of reboots

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.

Update Firmware	
New system file: 选择文件 未选择任何文件	System Update
New system file is depended from official upboits and undate system	Durtur Orline Histor
New system file is downloaded from official website and update system.	System Online Update

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

Upload Configuration	
New configuration file: 选择文件 未选择任何文件	File Upload
Backup Configuration	
Current configuration file version: 0.02.03	Download 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

Restore Configuration		
This will cause all the confi	iguration files to back to default factory values! And reboot your gateway once it finishes.	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

Model Name:	DGW-L1
Firmware Version:	1.1.0
Firmware Build:	1157
Hardware Version:	1.2
Port Amount:	1
Storage Usage:	9.3W/197.5M (5%)
Memory Usage:	9.84351 % Memory Clean
Kernel Build Time:	2015-Dec-25-15:42:53
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:	2016-1-4 12:16:31
System Uptime:	0 days 00:02:10



3. T1/E1

General

Figure 3-1-1 General Settings

General	
Locale:	United States
Interface Type:	© T1 ⊛ E1

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

General								
Loc	ale: Unite	ed States	•					
Interface T	vpe: 🔍 T1	• E1						
Port Details								
Port#	Timing Source Int	erface Framing	Coding	Line Build-out	CRC4	Signalling	Switch Type	Description
Port 1	0 • E1	CAS •	HDB3 V	0-133 feet (DSX-1) and 0 db (CSU) <	Off •	MFC/R2 •	Eurolsdn 🔻	

Save Apply Cancel

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,
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



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

Advanced: Interface Type	
RX Gain:	0
TX Gain:	0

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:0.0



ISDN: Signaling

Figure	3-2-2	ISDN:	Signaling
			<u>S-S</u>

VISDN: Signaling	
Q.SIG Channel Mapping:	Logical 🔻
Enable Caller ID:	
PRI Options	
PRI Dial Plan for Dialed Number	Unknown
PRI Dial Plan for Dialing Number:	Unknown
International Prefix:	
National Prefix:	
Local Prefix:	
Private Prefix:	
Unknown Prefix:	
Network Specific Facility Messages	None
Idle Bearer Reset:	OFF
Idle Bearer Reset Period:	never
Display Send:	Name
Display Receive:	Name
Overlap Dialing:	Disabled •
Allow Progress When Call Released:	ON
Out-of-Band Indications:	ON
Facility-based ISDN Supplementary Services:	
Exclusive Channel Selection:	
Ignore Remote Hold Indications:	
Block Outbound Caller ID Name:	OFF
Wait for Caller ID Name:	

Save Apply Cancel

Options	Definition
	Sets logical or physical channel mapping. In logical channel mapping,
Q.SIG Channel	channels are mapped to 1-30. In physical channel mapping, channels are
Mapping	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.

Table 3-2-2 Definition of Signaling



PRI Dial Plan for	PRI Dialplan: The ISDN_level Type Of Number or numbering plan, used					
	some ven unusual circumstances, you may need to set this to (dynamic)					
Dialed Number	some very unusual circumstances, you may need to set this to dynamic					
	or redundant.					
	PRI Local Dialplan: Only RARELY used for PRI(sets the calling numbre's					
PRI Dial Plan for	numbering plan). In North America, the typical use is sending the 10					
Dialing Number	digit; callerID number and setting the prilocaldialplan to 'national' (the					
	default); Only VERY rarely will you need to change this.					
Network Specific	Some switches (AT&T especially) require network specific facility IE					
Facility (NSF)	supported values are currently 'none', 'sdn', 'megacom',					
Messages	'tollfreemegacom', 'accunet'					
Idle Bearer Reset	Whether or not to reset unused B channels.					
Idle Bearer Reset	Time in seconds between reset of unused B channels.					
Period						
	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:					
Display Send	Use display text in other messages					
	NOTIFY/FACLITY for CLOP name update.					
	Name:					
	Name: Combined name_initial and name_update options.					
	Name: Combined name_initial and name_update options. Text:					
	Name: Combined name_initial and name_update options. Text: Pass any unused display text data as an arbitrary display message during					
	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.					



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 modesending 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

Link Set Settings										
Link Set Name	Enabled	Enabled ST	Use Connect	Hunting Policy	subservice	t35	Variant	OPC	DPC	Action
siuc (default)	yes	no	yes	even_mru	auto	15000,timeout	ITU	0x1	0x32	0
Add New SS7 Link Se	t									

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

Edit Link Set "linkset-siuc"

SS7 Link Set Settings	
Name:	siuc
Enabled:	
Enabled_st:	OFF
Use Connect:	
Hunting Policy:	even_mru •
Subservice:	auto
<u>t35:</u>	15000,timeout
variant:	ITU •
OPC:	0x1
DPC:	0x32
Set to Default:	

Save Cancel

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 I(0), auto or decimal/hex value; The auto means that the subservice is



t35 The value and action for t35. Value is in msec, action is eith		
	or timeout; if you use overlapped dialing dial plan, you might	
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

Link Settings													
Link Name	iftype	Enabled	Link Set	Channels	Schannel	First CIC	Echo Cancel	Echo Cancel Train	Echo Cancel Taps	SLS	SLTM	Port	Action
11	E1	yes	siuc	1-15,17-31	16	1	no	350	128			1	0
12	E1	yes	siuc	1-31		32	no	350	128			2	0
13	E1	yes	siuc	1-31		63	no	350	128			3	0
14	E1	yes	siuc	1-31		94	no	350	128			4	0

You can click

button as shown below.

Figure 3-3-4 SS7 Link Settings

Edit Link "link-l1"

V S	S7 Link Settings	
•	Name:	и
	Enabled:	
	Interface Type:	E1
	Link Set:	siuc •
	Channels:	1-15,17-31 Example: 1-15,17-31
	Schannel:	16
	First CIC:	1
	Echocancel:	no (default)
	Echocan Train:	350 Range: 10-1000 , 300 is default value
	Echocan Taps:	128 (default) •
	sis:	
	sitm:	OFF
	Port:	1 •

Save Cancel

SS7 Config. File Backup and Restore



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

🐨 SS7 Config. File Backup	
Download SS7 Configuration File	Download Backup
▼ SS7 Config. File Restore	
New configuration file: 选择文件 未选择任何文件	File Upload

MFC/R2

Advanced: Interface Type

Figure 3-4-1 Advanced: Interface Type

V Advanced: Interface Type	
RX Gain	0
TX Gain:	0

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

▼ MFC/R2: Signaling	
Enable Caller ID:	ON
Init CAS Bit:	1101
Variant:	ITU T

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	С	3	2	*
Bolivia	01	yes	1	5	5	F	F	3	2	×
Brazil	01	no	1	5	5	x	F	3	2	×
China	11	yes	1	1	6	x	F	3	2	×
Colombia	01	yes	1	5	5	F	F	3	2	×
Costa_rica	01	yes	1	5	5	x	F	3	2	×
Czech_republic	01	yes	1	5	5	F	F	3	2	×
Ecuador	01	yes	1	5	5	F	F	3	2	×
India	01	yes	1	4	5	x	F	3	2	×
Indonesia	01	yes	1	6	6	F	F	3	2	×
Israel	01	yes	1	9	9	x	F	3	2	×
ITU	01	yes	1	5	5	F	F	3	0	
Korea	01	yes	1	5	5	x	F	3	2	×
Malaysia	01	yes	1	6	6	F	F	3	2	*
Malta	01	yes	1	0	5	x	F	3	2	×

you can click

button, then you could fine the below.

Modify R2 Variant

Figure 3-4-4 General

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

|--|

Options Definition	
--------------------	--

Τ

T

Variant Name	Variant Name
R2 Category	Usually national_subscriber works just fine
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
CAS NonR2 Bits	Which bits are never used
CAS_R2_Bits	Which bits will be used

Figure 3-4-5 Timer

Timer	
MF Back Cycle:	5000
MF Back Resume Cycle:	150
MF Fwd Safety:	30000
R2 Seize:	8000
R2 Answer:	60000
Metering Pulse:	400
R2 Double Answer:	400
R2 Answer Delay:	150
CAS Persistence Check:	0
DTMF Start Dial:	500
DTMF Detection End:	5000

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.				

Table 3-4-4 Definition of Timer

Figure 3-4-6 Group A

Group A	
Request Next DNIS Digit:	1 •
Request DNIS Minus 1:	2 •
Request DNIS Minus 2:	7 •
Request DNIS Minus 3:	8 🔻
Request All DNIS Again:	INVALID •
Request Next ANI Digit:	5 •
Request Category:	5 •
Request Category And Change To Gc:	INVALID T
Request Change To G2:	3 •
Address Complete Charge Setup:	6 •
Network Congestion:	4 •



Figure 3-4-7 Group B

Group B	
Accept Call With Charge:	6 •
Accept Call No Charge:	7 •
Busy Number:	3 •
Network Congestion:	4 •
Unallocated Number:	5 •
Line Out Of Order:	8 •
Special Info Tone:	2 •
Reject Collect Call:	INVALID T
Number Changed:	INVALID •

Figure 3-5-8 Group C

Group C	
Request Next ANI Digit:	INVALID V
Request Change To G2:	INVALID V
Request Next DNIS Digit And Change To Ga:	INVALID V
Network Congestion:	INVALID V

Figure 3-4-9 Group 1

Group 1	
No More Dnis Available:	INVALID T
No More ANI Available:	C •
Caller ANI Is Restricted:	F T



٦

Figure 3-4-810Group 2

Group 2	
National Subscriber:	1 •
National Priority Subscriber:	2 •
International Subscriber:	7 •
International Priority Subscriber:	9 🔻
Collect Call:	INVALID V
Test Equipment:	3 •

Save Variant Cancel

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

SIP Endpoint			
Endpoint Name	Registration	Credentials	Actions
9000	client	9000@172.16.80.103	2 🗙
8001	server	8001	2
8002	server	8002	2 🗙

Add New SIP Endpoint

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 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

Add New SIP Endpoint

Main Endpoint Settings	
Name:	
User Name:	Anonymous
Password:	
Registration:	None
Hostname or IP Address:	
Transport:	UDP •
NAT Traversal:	Yes 🔻
Advanced:Registration Options	
Call Settings	
Fax Options	
Save Apply Cancel	

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

Add New SIP Endpoint	
V Main Endpoint Settings	
Name:	
User Name:	Anonymous
Password:	
Registration:	Endpoint registers with this gateway
Hostname or IP Address:	dynamic
Transport:	UDP •
NAT Traversal:	Yes
Advanced:Registration Options	
Call Settings	
Fax Options	
Save Apply Cancel	

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

Add New SIP Endpoint

Cancel

Save

Apply

Main Endpoir	nt Settings	
	Name:	
	User Name:	Anonymous
	Password:	
	Registration:	This gateway registers with the endpoint •
Ho	ostname or IP Address:	
	Transport:	UDP V
	NAT Traversal:	Yes 🔻
Advanced	d:Registration Options	
Call Settings		
Fax 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.		

Table 4-1-1 Definition of SIP Options



NAT Traversal

Addresses NAT-related issues in incoming SIP or media sessions.

Advanced: 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
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
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	How often, in seconds, to check the endpoint's connection
Outbound Proxy	A proxy to which the gateway will send all outbound signaling instead of sending signaling directly to endpoints.

Table 4-1-2 Definition of Registration Options

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

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	Whether or not to add;' user=phone' to URIs that contain a valid phone
user=phone to	number.
URI	
Add Q.850	Whether or not to add Reason header and to use it if it is available.
Reason Headers	
Honor SDP	By default, the gateway will honor the session version number in SDP packets
Version	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	Whether or not to allow 302 or REDIR to non-local SIP address .Note that
Promiscuous	promiscredir when redirects are made to the local system will cause loops
Redirects	since this gateway is incapable of performing a 'hairpin' call.
Max Forwards	Setting for the SIP Max-Forwards header (loop prevention).
Send TRYING on	Send 100 Trying when the endpoint registers.
REGISTER	

Table	4-1-4De	finition	of Sig	naling	Options
Lable		/ interor	OI DIG	incoming.	Options

Advanced Timer Settings

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
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

Table 4-1-5 Definition of Timer Options



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.

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

Table 4-1-6 Definition of Fax Options

Advanced SIP Settings

Networking



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.
	The maximum number of seconds a client has to authenticate. If
TCP Authentication Timeout	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 DNS SRV lookups on outbound calls Note: the gateway
	only uses the first host in SRV records Disabling DNS SRV lookups
Enable Hostname Lookun	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

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	Local IP address list that you added.

Table 4-2-2 Definition of NAT Settings Options



	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,
Subscribe	chan_sip will renew all outbound registrations when the monitor detects
Network Change	any sort of network change has occurred. By default this option is
Event	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
	The external hostname (and optional TCP port) of the NAT.
External	External Hostname=hostname[:port] is similar to
Hostname	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

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.

Table 4-2-3 Definition of RTP Settings Options

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.

Table 4-2-4	Instruction	of Parsing	and Com	natibility
1abic +-2-+	monucuon	or i ar sing	and Com	paubility



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	Minimum length of registrations/subscriptions (default 60).
Default Registration	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

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.

Table 4-2-5 Instruction of Security



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

▼ Codec Settings	
Codec Priority 1:	G.711 u-law 🔻
Codec Priority 2:	G.711 a-law ▼
Codec Priority 3:	GSM •
Codec Priority 4:	G.722 •
Codec Priority 5:	G.723 •
Codec Priority 6:	G.729 •



Advanced IAX2 Settings

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	Enable No checksums.
Enable Delay	You may specify a default AMA flag for iaxtel calls. It must be one of
Reject	'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 4-3-1 Instruction of General

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
	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)

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 4-3-3 Instruction of Codec Settings

Options	Definition
Jitter Buffer	Global default as to whether you want the jitter buffer at all
Force Jitter	In the ideal world, when we bridge VoIP channels we don't want to jitter
Buffer	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	A maximum size for the jitter buffer
Buffers	
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	The maximum number of interpolation frames the jitter buffer should
Interps	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	Number of milliseconds by which the new jitter buffer will pad its size. The
Extra	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 4-3-4 Instruction of Jitter Buffer

Table 4-3-5 Instruction of Misc Settings

Options	Definition
IAX Thread Count	Establishes the number of iax helper thread to handle I/O



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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_Nonvalidate	used to set the combined number of call numbers that can be
d	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

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	Maximum Transmission Rate
Transmission Rate	
Minimum	Minimum Transmission Rate
Transmission Rate	
Send Progress/Status	Manager events with 'call' class permissions will receive events
events to manager	indicating the steps to initiate a fax session. Fax completion events are
session	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

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

5. Routing

Move	Order	Rule Name	From	То	Rules	Actions
\$	1	out	sip-9000	Port-1	Callee_Dial_pattern +[[]((-+)] Caller_Dial_pattern +[[](-+)]]	2 🗶
¢	2	8001	sip-8001	Port-1	Callee_Dial_pattern + [] (- +)] Caller_Dial_pattern + [] (- +)	2 🗙
New Call	Routing F	Rule Save Orders				
_				New Call Ro	uting Rule	· · ·
'ou ar	e allo	owed to set up	new routing ru	le by	, and a	after setting routing
ules.	move	e rules' order b	ov pulling up ar	nd down, click 🙋	button to edit t	he routing and
ures,						
lelete	it. Fi	nally click the	Save Orders b	utton to save what	you set. Rules	shows current routir
ules.	Othe	rwise you can	set up unlimite	ed routing rules.		

Figure 5-1-1 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

Figure 5-1-2							
processing	prepend	prefix	Match	SdfR	StA	RdfR	Caller
rules			pattern				Name
Calling Transf	086	159	$\times \times \times \times$	4	0755		OpenVox
ormation			\times \times \times \times				
Called	086	136	$\times \times \times \times$	2	88		N/A
transformation			$\times \times \times \times$				

You can click

New Call Routing Rule

button to set up your routings.



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

<u> </u>			
Create	a Cal	l Routing	Rule

V Call Routing Rule]						
Routing Name:	support						
Call Comes in From:	9000 🔻						
Send Call Through:	Port-1 •						
Advance Routing Rule							
CalleeID/callerID Manipulation	<i>y</i>						
Callee_Dial_pattern Prepend	+ Prefix	[Match Pattern]]	(- SDfR	+ StA) RdfR		
Caller_Dial_pattern Prepend	+ Prefix	I Match Pattern]	(- SDfR	+ StA) RdfR	Caller Name	×
+ Add More Dial Pattern Fields						,	
Time Patterns that will use this Ro	oute						
Time to start: - ▼ : - Time to finish: - ▼ : -	· •	Week Day start: - Week Day finish: -	v	Month Day s Month Day fin	tart: - ▼ ish: - ▼	Month start: - Month finish: -	• 💥
+ Add More Time Pattern Fields	•						
Forward Number							
Forward Number	ŗ						
Failover Call Through Number Add a Failover Call Through Pro	wider						
Save Apply Cancel							

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
Call Comes in From	The launching point of incoming calls.
Send call Through	The destination to receive the incoming calls.



Options	Definition
Options Dial Patterns that will use this Route	Definition 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 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.
	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.

Table 5-1-2 Description of Advanced Routing Rule



Failover Call ThroughThe gateway will attempt to send the call out each of these inNumberthe 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

Advance Routing Rule							
CalleeID/callerID Manipulation							
Callee_Dial_pattern Prepend	+ Prefix	I Match Pattern] (-SDfR	+ StA) RdfR		
Caller_Dial_pattern Prepend	+ Prefix	I Match Pattern] (-SDfR	+ StA) RdfR	Caller Name	*
+ Add More Dial Pattern Fields							

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

Time Patterns that will use this Route				
Time to start: 00 ▼ : 00 ▼	Week Day start: Monday	Month Day start: 01 Month Day finish: 31	Month start: January	×
+ Add More Time Pattern Fields	Thuisday	Monur Day Innan. 31	March V	

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

Forward Number	
Forward Number	

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

Figure 5-1-7 Failover Call Through Number

Failover Call Through Number	
Failover Call Through Number 1:	None •
Add a Failover Call Through Provider	

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

Routing Groups	
Group Name:	
Туре:	T1/E1 •
Policy	Ascending
Members	NO. All 1 Port-1

Save Apply Cancel



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.

WAN Setting	
Interface:	eth0
Туре:	Static 💌
MAC:	A0:98:05:01:DB:4B
IP Address:	172.16.100.1
Netmask:	255.255.0.0
Default Gateway:	172.16.0.1
LAN Setting	
Interface:	eth1
Enable:	ON
MAC:	A0:98:05:01:DB:4C
IP Address:	192.168.100.1
Netmask:	255.255.2
Default Gateway:	192.168.0.1

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.
Туре	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.

DNS Servers	
DNS Server 1:	8.8.8.8
DNS Server 2:	
DNS Server 3:	
DNS Server 4:	

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

DDNS Settings	
DDNS	ON
Туре:	phddns 🔻
User Name:	
Password:	
Your domain:	

Save

Table 6-2-1	Definition	of DDNS	Settings
-------------	------------	---------	----------

Options	Definition
DDNS	Enable/Disable DDNS(dynamic domain name server)
Туре	Set the type of DDNS server.
Username	Your DDNS account's login name.
Password	Your DDNS account's password.

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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

www.google.com	Ping	
www.google.com	Traceroute	
Report		
		ping -c 4 www.google.com
PING www.google.com (64.233.16 64 bytes from 64.233.162.83; seq= 64 bytes from 64.233.162.83; seq= 64 bytes from 64.233.162.83; seq= www.google.com ping statistics 4 packets transmitted, 3 packets re round-trip min/avg/max = 316.249/	2.83): 56 data bytes 1 ttl=37 time=316.708 ms 2 ttl=37 time=317.361 ms 3 ttl=37 time=316.249 ms ceived, 25% packet loss 316.772/317.361 ms	
		Result

Successfully ping [www.google.com] .



7. Advanced

Asterisk API

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

General	
Enable:	ON
Port:	5038
Manager	
Manager Name:	admin
Manager secret:	admin
Deny:	0.0.0/0.0.0.0
Permit:	172.16.100.110/255.255.0.0&192.168.1.0/2
Rights	
System:	read: 🗹 write: 🗹
Call:	read: 🗹 write: 🗹
Log:	read: 🗹 write: 🗹
Verbose:	read: 🗹 write: 🗹
Command:	read: 📃 write: 🗹
Agent:	read: 🗹 write: 🗹
User:	read: 🖉 write: 🖉

Figure 7-1-1 API Interface

Fable 7-1- 1	Definition	of Asterisk API
---------------------	------------	-----------------

write: 🗹

Options	Definition
Port	Network port number

Config:

,

read: 🗹

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 & asseparator.Example:0.0.0.0/0.0.0.0or192.168.1.0/255.255.255.0&10.0.0/255.0.0.0	
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&10.0.0/255.0.0.0	
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	
Verbose	Verbose information. Read-only. (Defined but not yet	
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

P 172.16.100.110 - PuTTY
[wh@IX130 tmp]#telnet 172.16.100.110 5038
Asterisk Call Manager/1.3
action: login
username: admin
secret: admin
Response: Success
Message: Authentication accepted
Event: FullyBooted
Privilege: system,all
Status: Fully Booted

Asterisk CLI

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

Figure 7-2-1 Asterisk Command Interface

Asterisk CLI	
Command:	? Execute

Output:

Execute a shell command
acl show Show a named ACL or list all named ACLs
ael reload Reload AEL configuration
ael set debug {read tokens mac Enable AEL debugging flags
agent logoff Sets an agent offline
agent show Show status of agents
agent show online Show all online agents
agi dump html Dumps a list of AGI commands in HTML format
agi exec Add AGI command to a channel in Async AGI
agi set debug [on off] Enable/Disable AGI debugging

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	
<u>ystem.conf</u>		92	
sip.conf		105	
sip endpoints.conf		1683	
logger.conf		4765	
extensions.conf		122	
sip_general.conf	o general.conf		
extensions macro.conf	extensions_macro.conf		
extensions routing.conf		1111	
dahdi-channels.conf		184	
chan dahdi.conf		871	
<u>17.conf</u>		391	
Config. Files List			
File Name	File Size		
aciconf	2817		
agents.conf	2531		
alarmreceiver.conf	2084		
amd.conf	767		
sterisk.conf 4237			
andar.conf 5171			
ccss.conf	8827		
cdr.conf	133		
cdr_custom.conf	1617		

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

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 V

Figure 8-1-1 Logs Settings



MFC/ KZ LOgs	
MFC/R2 Logs:	OFF
Auto clean:	ON maxsize : 2MB V
PRI Logs	
PRI Logs:	OFF
Auto clean:	ON maxsize : 2MB V
SS7 Logs	
SS7 Logs:	OFF
Auto clean:	ON maxsize : 2MB V
Call Statistics	
Call Statistics:	ON
System Notice	
Enable:	OFF
Check Interval:	Every week •
Save	

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 23:29:0	08] first starting up
[2012/01/01 23:29:2	27] Power on
[2015/03/25 20:50:1	18] Kernel upgrade
[2015/03/25 20:50:2	20] Basefs upgrade
[2015/03/25 20:50:4	40] Power off
[2015/03/25 20:51:1	14] Power on
[2015/03/25 19:35:4	47] Power on
[2015/03/25 19:41:1	15] Power off
[2015/03/25 19:41:5	52] Power on
[2015/03/25 19:49:0	08] Power on
[2015/03/25 19:56:2	25] Power on
[2015/03/25 20:01:2	22] Power on
[2015/03/25 22:47:5	50] Power on
[2015/03/25 23:25:1	13] Power on
[2015/03/25 23:40:0	09] Power on
[2015/03/26 03:40:4	48] Power on
[2015/03/26 04:17:0	00] Power on
[2015/03/26 05:37:0	03] Power on
[2015/03/26 08:49:0	08] Power on
[2015/03/26 09:04:2	24] Power on
[2015/03/26 09:30:0	00] Power on
[2015/03/26 12:01:3	38] Kernel upgrade
[2015/03/26 12:01:4	40] Basefs upgrade
[2015/03/26 13:32:4	49] first starting up
[2015/03/26 13:32:5	52] Power off
[2015/03/26 13:33:3	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 (*mporarily unavailable Dec 31 09:5740 (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]: cham_dahdi.c:4094 in dahdi_r2_on_os_error: OS error on cham 20: Resource temporarily unavailable Dec 31 13:21:55 (none) asterisk[1187]: ERROR[1785][C-000017e5]: cham_dahdi.c:4395 in dahdi_r2_write_log: Cham 29 - Failed to read from chammel 29: Resource temporarily unavailable umavailable
Dec 31 13:21:55 (none) asterisk[187]: ERROR[1785][C-000017e5]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 29: Resource temporarily unavailable
Dec 31 13:24:07 (none) asterisk[187]: 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[187]: ERROR[1451]: chan_dahdi.c:4395 in dahdi_r2_on_os_error: 05 error on chan 7: Resource temporarily unavailable
Dec 31 13:24:07 (none) asterisk[187]: ERROR[1451]: chan_dahdi.c:4395 in dahdi_r2_on_os_error: 05 error on chan 7: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[1445]: chan_dahdi.c:4395 in dahdi_r2_on_os_error: 05 error on chan 1: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[1445]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 1: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[1445]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 1: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[145]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 1: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[145]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 1: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[145]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 1: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[145]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 11: Resource temporarily unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[145]: chan_dahdi.c:4094 in dahdi_r2_on_os_error: 05 error on chan 11 = Failed to write MF tone to channel 11: Resource
temporariy unavailable
Dec 31 13:29:58 (none) asterisk[187]: ERROR[1737][C-000193c]: chan_dahdi.c:4395 in dahdi_r2_write_log: Chan 11 - Failed to write MF tone to channel 11: Resource
temporariy unavailable
Dec 31 13:29:5 temporarily unavailable (temporarily unavailable Dec 31 13:35:66 (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".