





GSMGW1 – VoIP to GSM Gateway, 1 SIM

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GSMGW1

User Guide ver. 1.1



1 SIM, SIP, 2xRJ45

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1 Introduction

1.1 Overview

A VoIP GSM Gateway is an IP-based system that enables inbound and outbound VoIP and GSM Cellular calls. It is an alternative to a VoIP FXO Gateway especially in area where GSM service is readily available and cheaper for VoIP call termination. Many applications can be evolved from this technology using GSM termination and the immediate benefit is for great cost saving and fast and low cost system deployment. Typical applications are call centers, VoIP termination and cellphone roaming. The diagram below shows a typical topology for a VoIP GSM Gateway.



1.2 Protocol

TCP/IP V4 (IP V6 auto adapt) ITU-T H.323 V4 Standard H.2250 V4 Standard H.245 V7 Standard H.235 Standard (MD5 , HMAC-SHA1) ITU-T G.711 Alaw/ULaw, G.729A, G.729AB, and G.723.1 Voice Codec RFC1889 Real Time Data Transmission Proprietary Firewall-Pass-Through Technology SIP V2.0 Standard Simple Traversal of UDP over NAT (STUN) Web-base Management PPP over Ethernet (PPPoE) PPP Authentication Protocol (PAP)





Internet Control Message Protocol (ICMP) TFTP Client Hyper Text Transfer Protocol (HTTP) Dynamic Host Configuration Protocol (DHCP) Domain Name System (DNS) User account authentication using MD5 Out-band DTMF Relay: RFC 2833 and SIP Info

1.3 Hardware Specification

ARM9E Processor DSP for voice codec and voice processing Two 10/100 BaseT Ethernet ports with full compliant with IEEE 802.3 LEDs for Ethernet port status One GSM Connection

1.4 Software Specification

LINUX OS Built-in HTTP Web Server PPPoE Dial-up NAT Broadband Router Functions DHCP Client DHCP Server Firmware On-line upgrade Caller ID Multiple Language Support Support Multi_devices Cooperate Mode

1.5 List of the Package

- a) One GSM Gateway main unit
- b) One DC5V/300mA power adaptor
- c) One Ethernet cable (3M)

1.6 Appearance







VoIP GSM Gateway (GSMGW1)

1) LAN(WAN-side)

Connect this port to an Ethernet Switch/Router, the Ethernet of a DSL modem, or other network access equipment.

2) PC(LAN-side)

Connect a computer or other network device to this port.

3) POWER (DC5V/300mA)

Connect the 5V/300mA Adapter provided to this power jack.

4) Reset

Press this button to reset the GSMGW1 Gateway to factory defaults.

2 Installation

2.1 Connection Diagram

Please follow the connection diagram above to install the GSMGW1 Gateway. a) Insert a GSM SIM card in the SIM card compartment located at the bottom of the GSMGW1 Gateway's.

b) Connect an Ethernet cable the LAN port of the GSMGW1 Gateway and the other end to your existing network equipment.

c) Connect an Ethernet cable to the PC Port of the GSMGW1 Gateway and the other end to a PC or other network device (Optional).

d) Connect the power adapter provided to the power jack of the GSMGW1 Gateway.

The diagram below shows a typical installation of the device.







2.3 LED Indicators

The following table defines the status of the LEDS located on the top case and on the RJ-45 connectors.

LED	DESCRIPTION
RUN	1. When the GSMGW1 is booting, this LED will flash 100ms ON and 100ms OFF.
	2. When the GSMGW1 is login your softswitch, this LED will flash 1s ON and 1s OFF.
GSM	When the GSMGW1's GSM login the ISP's system, this LED will flash 1s ON and 1s OFF.





3 Configuration Guide

To configure the GSMGW1 Gateway, you must login to its Web server via the LAN or PC port. The LAN port is factory preset to obtain an IP from the local DHCP host and the PC port is set to the fixed IP 192.168.8.1.

If a local DHCP host is available, the LAN will obtain an IP address automatically. To listen to the IP address assigned, just dial a call via the GSMGW1 Gateway's SIM card phone number. When the call is connected, you will hear a dial tone. Just dial "*01#" for English voice prompt on the LAN IP.

The LAN IP Address can also be obtained by sending a SMS message to the GSM phone number. The GSMGW1 will then reply with a SMS message containing the LAN IP address. If you want obtained LAN port IP by sending a SMS message, please send" INFO "or" info".

If a local DHCP host is not available, you can then access the GSMGW1 Gateway via the PC port. You will need to change the PC LAN configuration via the Network Connections under the Control Panel.

Windows:

Control Panel Network Connections Local Connections Property **TCP/IP Protocol Property**

Internet Protocol (TCP/IP) Properties					
General					
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.					
00	otain an IP address automatically	,			
- 💽 U:	● Use the following IP address:				
IP ac	ldress:	192.168.8.10			
Subr	net mask:	255 . 255 . 255 . 0			
Defa	ult gateway:				

Set an unused IP address that is in the same segment as the PC port address.

Once either the IP address of the LAN or PC port is known, you are now ready to access the Web server of GSMGW1 Gateway.





3.1 Web Configuration Menu

If your PC is connected to the GSMGW1 Gateway via the LAN port network segment, you need to type the LAN IP address of the GSMGW1 Gateway in your Web Browser to access the Web server of the GSMGW1 Gateway. If not, you should type the PC IP address (192.168.8.1) in the Web Browser.

File Edi	t View Favori	tes Tools	Help				
G Back	• •	2 2 🤇	Search	Ravorites	Θ	Ø• 🎍	0
Address	192.168.8.1						

If the connection is correct, the Web Browser will prompt you to enter the "User name" and "Password: as shown below.

2 🛛
The server 192.168 at "Please Login" requires a username and password.
Warning: This server is requesting that your username and password be sent in an insecure manner (basic authentication without a secure connection).
User name: 🙍
Password:
Remember my password
OK Cancel

Enter the User name and Password and the press OK to access the GSMGW1 Gateway Web Server. The default for both user name and password is "**admin**".

GSMGW1





3.2 Status

The Status page shown below is the default / home page of the GSMGW1 Web server.

Status						
Phone Information		Network Info	rmation	GSM Module Information		
Serial Number		LAN Port	192.168.2.226	GSM Model	GTM900A	
Firmware Version	GHS-3.01	LAN MAC		GSM Signal	31	
Hardware	GolP	PC Port	192.168.8.1	GSM Status	LOGIN	
Model Phone Status	LOGIN	PPPoE	Disabled			
i nono otatao	20011	Default Rout	e 192.168.2.254			
		DNS Server	202.96.134.133			

3.2.1 Phone Information

A. Serial Number

Each Gateway has a unique serial number assigned by the factory such as HT304O12VTEST01. This number is important for centralized configuration, technical support, and warranty. This number is printed on the bottom of the Gateway and is associated with your software license.

B. Firmware Version

Firmware version identifies the firmware version of the Gateway such as GHS-3.01.

C. Hardware Mode

This field shows terminal's hardware type.

D. Phone Status

This field shows the status of Line's connection status. If the connection is successful, this field displays LOGIN; otherwise, it displays LOGOUT.

3.2.2 Network Information

A. LAN Port Configuration

This field displays the status of the LAN port.

B. PC Port Configuration

This field displays the status of the LAN port.

C. PPPoE

If PPPoE is enabled, it displays its status.

D. Default Route

This field displays the IP address of the default routing Gateway.

E. DNS Server

This field displays the IP address of the Domain Name Server.





3.2.3 GSM Module Information

A. GSM Module

This field displays the GSM module type.

B. GSM Signal

This field displays the GSM signal type.

C. GSM Status

This field shows the status of GSM connection status. If the connection is successful, this field displays LOGIN; otherwise, it displays LOGOUT.

3.3 Configurations

Click on the "Configurations" tab on the left hand column to access the device configuration menu:

Preference, Network, Call Settings, Call Divert, Save Changes, and Discard Changes.

GSMGW1			_			简体中文
Statuc	Preference					
Status	Language(语言)	简体中文	~	Network Tones	China	~
Configurations	Time Zone	GMT+8				
Droforoncos	Time Server	pool.ntp.org				
Ficicicios	Auto-provision	O Enable O Disable	е			
Network	Network Confi	guration				
Call Settings	LAN Dort	DHCP		DC Dort	Static IP	
Call Divert	CARPON AND	OF CHILD		ID Address	102 169 0 1	L.M.
	802.10 VLAN	C Enable C Disabi	e	IP Address	192.100.0.1	
Save Changes		Advanced>>		Subhet Mask	0.5.11.01	
Discard Changes				DHCP Server	O Enable O I	Disable
Tools					Advanced>>	
10013	Call Settings					
	Endpoint Type	H.323 Phone	~		Advanced Settin	igs>>
	Endpoint Mode	Gatekeeper Mode	~		Media Settings	

Click on "**Preference**" in the left menu of the configuration web, and the screen will be displayed as below:

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Preference			
Language(语言)	简体中文 🛛 🖌 🖌	Network Tones	Australia 🛛 🗸
Time Zone	GMT+8	GSM Group Mode	As Client 🔽
Time Server	pool.ntp.org	Server Address	
DTMF Min Detect Time Gap	5	GSM Number	
Auto-provision	🔘 Enable 💿 Disable		

3.3.1 Language

Currently GSMGW1 supports English, Simplified Chinese and Traditional Chinese. Select the language desired and the Web page will be shown in the language selected accordingly.

Language(语言)	简体中文	~
	English	
	简体中文	
	繁體中文	

The language can also be selected at the top of the web page. Once selected, the webpage language is refreshed immediately. However, the language selection is not saved until the **Save Changes** icon is clicked.

3.3.2 Time Zone and Time Server

The GSMGW1 Gateway supports Network Time Protocol (NTP) to obtain the date and time information from an NTP server (Time Server). The time zone is specified as in GMT \pm offset. For example, the Pacific Standard Time is GMT-8, and the Pacific Daylight Time is GMT-7.

Time Zone	GMT+8
Time Server	pool.ntp.org

Note: The GSMGW1 Gateway supports CDR and Billing Information, it is important to set up these two parameters properly.

3.3.3 Auto-Provision

The GSMGW1 Gateway supports Auto Provisioning which enables configuration parameters to be set automatically without human intervention. The Auto Provisioning supports both HTTP and TFTP protocols. For higher security, encrypted configuration file is also supported. This feature requires external Auto Provisioning Server. Please contact your service provider for further information on this.

Auto-provision

Enable
Disable

Provision Server

Provision Interval

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3.3.4 Network Tone

Network Tones are a set of tones used for VoIP calls. Select one of the countries defined or customized to define your own Network Tones.

Network Tones	China 🗸
	Australia
	China
	Hong Kong
	New Zealand
	United Kingdom
	United States
	Customized

You can configure the Network Tones as Customized option:

Network Tones	Customized 🗸
Dial Tone	
Ring Back Tone	
Busy Tone	
Indication Tone	

Each tone listed above is defined in the following format: nc, rpt, c1on, c1off, c2on, c2off, c3on, c3off, f1, f2, f3, f4, p1, p2, p3, p4 Where: nc is the number of cadences **rpt** is the repeat counter(0 - infinite, 1~n - repeat 1~n times) **c1on** is the cadence one on (in milliseconds) cloff is the cadence one off (in milliseconds) c2on is the cadence two on (in milliseconds) **c2off** is the cadence two off (in milliseconds) c3on is the cadence three on (in milliseconds) c3off is the cadence three off (in milliseconds) f1 is the tone #1 frequency (300Hz-3000Hz) f2 is the tone #2, frequency (300Hz-3000Hz) f3 is the tone #3 frequency (300Hz-3000Hz) f4 is the tone #4 (300Hz-3000Hz) p1 is the attenuation index for f1, 0~31(0=3dB, -1dB increments) p2 is the attenuation index for f2, 0~31(0=3dB, -1dB increments) p3 is the attenuation index for f3, 0~31(0=3dB, -1dB increments) p4 is the attenuation index for f4, 0~31(0=3dB, -1dB increments)

For example, the tone definition for a tone of 450Hz with a cadence of 700 ms on and 1000 ms off is **1,0,700,1000,0,0,0,450,0,0,20,0,0,0**





3.3.5 GSM Group Mode

The GSM Group mode enables multiple GSMGW1 devices to simulate a multi-channel GSM gateway. In this mode, only one GSMGW1 acts as a **Server** and the others act as clients of the server and reports its GSM number and status to the server. The number of clients is not restricted. When the server receives a GSM call, it finds a idle client (not engaged in a GSM call) and then forward the call to this client. This enables a scalable multi-channel VoIP GSM Gateway. A typical application is to implement a Call Center that is accessed via a single phone number (GSM).

GSM Group Mode	Disable	~
	Disable	
	As Server	
	As Client	

When **Client Mode** is selected, the Server IP Address and the Client GSM Number are required to be filled in as shown below.

GSM Group Mode	As Client	~
Server Address		
GSM Number		

Note: Each GSMGW1 still needs to register to VoIP server or proxy separately.

3.4 Call Settings

Click on the "**Call Settings**" to configure the VoIP call settings. The first thing to set is the Endpoint Type: H.323 or SIP.

Call Settings		
Endpoint Type	H.323 Phone 🛛 🗸	
	H.323 Phone	
	SIP Phone	

3.4.1 H.323 Phone

For H.323 protocol phone, 2 configuration modes are supported: **Single Configuration** and **Configuration by Group**.

Config Mode

Single Config	*
Single Config	
Config by Group	





3.4.1.1 Single Configuration

The **Single Configuration** supports only one VoIP number to a single H.323 Gatekeeper.

Call Settings			
Endpoint Type	H.323 Phone 🛛 🗸		Advanced Settings>>
Endpoint Mode	Gatekeeper Mode 🛛 👻]	Media Settings≻≻
Config Mode	Single Config 🛛 🗸 🗸		
Phone Number]	
GateWay Prefix]	
Display Name]	
H.323 ID]	
Gatekeeper Address]	
	Enable Authentication		

A. H.323 Phone Number

H.323 phone number: fill the login number (E164) here.

B. Gateway Prefix

If login with a Prefix method fill the prefix number (do not fill the Phone number)..

C. Display Name

Display name is the name to be displayed on the called VoIP party.

D. H.323 ID

If the system requires an H.323 ID as a method of Authentication, enter the H.323 ID provided.

E. Gatekeeper Address

This field assigns the IP address or the domain name of the gatekeeper. The port number can be added with the colon ":" symbol. For example: 192.168.1.70:8080.

F. Enable Auth

If H.235 Authentication is required, enable this field and fill in the values as provided.

H.235 Auth H.235 Id Password



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3.4.1.2 Configuration by Group

The "**Config by Group**" mode allows a user to setup the GSMGW1 Gateway to have 4 identities by registering to the same gatekeeper with different phone numbers or to different gatekeepers with different phone numbers or the same phone number, or a combination of both. The GSMGW1 Gateway can be assigned to each group individually. This allows the VoIP Chunnel to be shared by each group.

Call Settings	
Endpoint Type	H.323 Phone 🛛 👻
Endpoint Mode	Gatekeeper Mode 🛛 👻
Config Mode	Config by Group 🛛 🗸
📀 Group 1 🔘 Group	2 🔿 Group 3 🔿 Group 4
Phone Number	
H.323 ID	
GateWay Prefix	
Gatekeeper Address	
	H.235 Auth
H.235 ID	
Password	
Activated Lines in Gro	up 1
🗌 L1	

3.4.1.4 Advance Settings

Click "Advance Settings" to access additional H.323 parameters as shown below.

	Advanced Settings<<
RAS Port	
Q.931 Port	
H.245 Port	
Fast Start	💿 Enable 🔘 Disable
Register Mode	Register Multiple Nur 🗸
DTMF Signaling	Outband 🛛 🗸
Signaling QoS	None 💌
Signaling NAT Traversal	None 👻

A) RAS Port

RAS Port is an unreliable channel which is used to convey the registration, admissions, bandwidth change, and status messages between two H.323 entities.



B) Q.931 Port (Call Signaling Port)

Call Signaling Port is a reliable channel which is used to convey the call setup and release messages between two H.323 endpoints.

C) H.245 Port (Media Control Ports)

Media control port is the port or port range used by the H.245 media control protocol.

D) Fast Start

Enable or disable the Fast Start in H.225.0. Most H.323 terminals or Gateways support the **Fast Start** feature.

E) Register Mode

Register Multiple Numbers: The GSMGW1 Gateway sends registration request in one signaling packet to the gatekeeper. In the mode, one signaling packet includes two VoIP line's registration information.

Register Mode

Register Multiple Nur 🔽
Register Multiple Numbe
Register Multiple Times

Register Multiple Times: In this mode, the GSMGW1 Gateway will register like two terminals.

F) DTMF Signaling

1) DTMF TYPE

DTMF signals can be sent over to the called party once a call is established. GSMGW1 Gateway supports both **Inband** and **Outband** DTMF signal types.

DTMF Signaling

Οu	itband	~
Inb	and	
Ou	tband	

For **Inband** DTMF type, DTMF signals are generated locally at the calling phone and then send to the called party as part of the voice signals. This method is not reliable since the quality of the DTMF signals is subject to the Codec used and the quality of the network traffics.

For **Outband** DTMF type, DTMF signal commands are sent to the called party and the actual DTMF signals are actually generated by the called party. This method allows more reliable DTMF signaling. However, it requires the called party to support this feature in order for this to work properly. GSMGW1 Gateway supports **RFC2833** Outband DTMF protocols.

2) DTMF Payload Type

DTMF Payload Type is by RFC2833 protocol to carry the tone definitions for various applications. The default DTMF Payload Type is 96. Please consult your VoIP service provider for the proper setting if required.

3.4.1.5 H.323 Direct Mode

The **Direct Mode** allows peer-to-peer calls without registering to a gatekeeper.





3.4.2 SIP Phone

Set the "**Endpoint Type**" to SIP Phone for connections to SIP Servers. GSMGW1 Gateway's SIP configure page as follow:

Call Settings	
Endpoint Type	SIP Phone 🖌 🖌
Single Server Mode	
Phone Number	
Display Name	
SIP Proxy	
SIP Registrar	
Register Expiry(s)	
Outbound Proxy	
Home Domain	
Authentication ID	
Password	
Dial Plan	
Call Forward Type	Not Forward 🛛 👻
Call Forward Number	
Backup Server	🔿 Enable 💿 Disable

A) Phone Number

Enter a SIP phone number.

B) SIP Proxy

Enter the SIP proxy IP address or domain name. If the registration port isn't 5060, then add """ and the port number. An example is **sip.hybertone.com:8080**.

C) SIP Registrar Server

If the Registrar Server is different from the SIP Proxy, enter its IP address or domain name in this field. If the registration port isn't 5060, then add "" and the port number. An example is **sip.hybertone.com:8080**.

D) Home Domain

SIP Networks sometimes use the Home Domain name as an identifier. Enter this field as required.

E) Authentication ID

Enter the Authentication ID as provided.

F) Password

Enter the authentication password as provided.

G) Display Name

Enter this field for the name to be displayed on the called VoIP party.





H) Backup Server

The GSMGW1 Gateway supports one Backup Server as an alternative to the main server. Once registration to the main server fails, the GSMGW1 Gateway will try to register to the Backup Server.

Backup Server	⊙Enable ○Disable
Backup SIP Proxy	
Backup SIP Registrar Backup Home Domain Fail-retry Interval(1- 60s)	

I) Outbound Proxy

OutBound proxies are devices that will forward SIP signaling (and frequently RTP media traffic too). OutBound proxies are used for a number of reasons, including, firewall traversal – both in parallel with a firewall and situated in the Internet as a Session Border Controller, and also for hiding customer IP addresses – calls are all routed through one point so that a public ITSP address can be used for accessing the customers, rather than the customer's own IP address. If required, enter this field with the outbound proxy IP address or domain name as provided.

3.4.2.1 Advanced Settings

Click on "**Advance Settings**" tab on the top right corner of the Call Setting page to display all the parameters available, as shown below, for programming. These parameters allow more advanced control over the SIP signaling and media preference.

	Advanced Settings<<
Signaling Port	5060
NAT Keep-alive	💿 Enable 🔘 Disable
	Advanced Timing>>
DTMF Signaling	Inband 🛛 👻
Signaling QoS	None 🗸 🗸
	Enable RC4 Encryption
	Enable Fast Encryption
Signaling NAT Traversal	None 🗸

A) Signaling Port (SIP Local port)

The default SIP port is 5060. Change this as required.





B) NAT Keep-alive

The NAT Keep-alive feature sends a null packet to the SIP Proxy periodically in order to keep the NAT open for incoming data traffics.

NAT Keep-alive 💿 Enable 🔘 Disable

C) Advanced Timing Settings

Some SIP proxies may have special timing requirements. Change these parameters as required.

	Advanced Timing<<
No Answer Expiry (32-180s)	
NICT Expiry(2-180s)	
ICT Expiry(5-360s)	
Retransmit T1(200- 2000ms) Retransmit T2(2000- 8000ms)	

D) Signaling Qos

Signaling QoS improves the performance of SIP signaling. If local network device supports Qos, select this field accordingly. Please consult your network administrator for further information.

Signaling QoS

None	¥
None	
IP TOS	
DiffServ	

E) DTMF Signaling

1) DTMF TYPE

DTMF signals can be sent over to the called party once a call is established. GSMGW1 Gateway supports both **Inband** and **Outband** DTMF signal types.

DTMF Signaling

Outband	*
Inband	
Outband	

For **Inband** DTMF type, DTMF signals are generated locally at the calling phone and then send to the called party as part of the voice signals. This method is not reliable since the quality of the DTMF signals is subject to the Codec used and the quality of the network traffics.

For **Outband** DTMF type, DTMF signal commands are sent to the called party and the actual DTMF signals are actually generated by the called party. This method allows more reliable DTMF signaling. However, it requires the called party to support this feature in order for this to work properly. GSMGW1 Gateway supports both RFC2833 and SIP INFO **Outband** DTMF protocols.

2) DTMF Payload Type

DTMF Payload Type is by RFC2833 protocol to carry the tone definitions for various applications. The default DTMF Payload Type is 96. Please consult your VoIP service provider for the proper setting if required.





3.4.3 Media Setting

Click on "**Media Settings**" in the "Call Setting" menu to access the parameters available for media settings.

	Media Settings≺≺	
RTP Port (range)	16384 - 32768	
Packet Length (ms)	20	
Jitter Buffer Mode	Fixed 🗸	
Minimum Jitter	60	
Maximum Jitter(soft limit)		
Media QoS	None 🖌 🗸	
	Enable RC4 Encryption	
	Symmetric RTP	
Media NAT Traversal	None 🗸 🗸	

Audio Codec Preference>>

A) RTP Port Range

This parameter specifies the range of the RTP (Real Time Protocol) Ports used by the GSMGW1 Gateway. If your network limits the usable port range, this parameter may need to be modified. Please consult your network administrator for more information.

B) Packet Length

This parameter defines the voice packet length. The default setting is 20ms. The range is from 5ms to 40ms at an increment of 5 ms. Please note that some codes have a minimum packet length of more than 5 ms.

C) Jitter Buffer Mode

Jitter Buffer Mode	Fixed 🔽
Minimun Jitter	
Maxinum Jitter(soft limit)	

Since data packets may arrives at different orders, the Jitter Buffer is used to hold the data packets received for re-arrangement according to the packet sequence number. Three Jitter Buffer Modes are supported: Adaptive, Sequential, and Fixed. The default is set to Adaptive mode with a minimum jitter of 60 ms and a maximum jitter of 120ms. Please consult your network administrator for more information on the network environment in order to determine the optimal settings.

D) Media Qos

Media QoS

None	¥
None	
IP TOS	
DiffServ	

Similar to the Signaling QoS, the Media Qos in intended to improve the voice performance or quality If your local network supports QoS







3.4.4 Codec Preference

Codec Preference allows a user to select the codes to be used and its priority to be selected for a voice call.

Audio Codec Preference<<

UP DOWN	✓ alaw ✓ ulaw ✓ g729 ✓ g729a ✓ g729ab
	₽ g7231

Click on the check box to enable a codec. Select a codec and then press the UP or DOWN button to move the position of the codec on the codec list with a priority in descending order.

3.4.5 NAT Traversal

3.4.5.1 Signaling NAT Traversal

Signaling NAT traversal may be required if the GSMGW1 Gateway is put behind a NAT (or multiple NATs). Depending on your network environment and SIP Server capabilities, this feature may or may not be turn on.

Signaling NAT Traversal

None 🗸
None
STUN(RFC 3489)
Relay Proxy

A) None

Select None to turn off this feature.

B) STUN (RFC 3489)

STUN (Simple Traversal of UDP (User Datagram Protocol) through NATs

(Network Address Translators)) is a network protocol allowing a client behind a NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet-side port associated by the NAT with a particular local port.

Select **STUN (RFC 3489)** to use a STUN server for Signaling NAT Traversal. Enter the IP Address or the domain name of the STUN server to be used.

C) Relay Proxy

Relay proxy is a proprietary NAT traversal technology. Please consult your service provider for more information.

3.4.5.2 Media NAT Traversal

Similar to Signaling NAT Traversal, this feature allows media packets (RTP) to be routed properly in various network environments.

Media NAT Traversal None



A) None

Select **None** to disable this feature.

B) STUN (RFC 3489)

STUN (Simple Traversal of UDP (User Datagram Protocol) through NATs

(Network Address Translators)) is a network protocol allowing a client behind a NAT



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(or multiple NATs) to find out its public address, the type of NAT it is behind and the internet-side port associated by the NAT with a particular local port. Select **STUN(RFC 3489)** to use a STUN server for Signaling NAT Traversal. Enter the

IP Address or the domain name of the STUN server to be used.

C) Port forwarding Support

Port forwarding (sometimes referred to as tunneling) is the act of forwarding a network port from one network node to another. This technique can allow an external user to reach a port on a private IP address (inside a LAN) from the outside via a NAT-enabled router.

In order for this feature to work, the local network Gateway must support this feature and be set up properly. Please consult your network administrator for help to enable this **Port forwarding** feature.

D) Relay Proxy

Relay proxy is a proprietary NAT traversal technology. Please consult your service provider for more information.

Currently, the following 3 kinds of packaging mechanism are supported:

Mode 1: The media uses UDP packets and (or) encrypt with multiple UDP port;

Mode 2: The media uses UDP packets and (or) encrypt with single UDP port; Mode 3: The media uses TCP packets and (or) encrypt (UDP over TCP).

Media NAT Traversal	Relay Proxy 💌
Address	
Port	
User Name	
Password	
	Encryption
Relay Mode	○ 1 ○ 2 ○ 3

3.5 Call divert

The Call divert feature controls the routing of calls between VoIP and GSM.

Call Divert	
Forward to PSTN	📀 Enable 🔘 Disable
Forward Number (VoIP To PSTN) Forward Password (VoIP To PSTN) Dial Plan(VoIP to PSTN)	
Forward to VoIP	💿 Enable 🔘 Disable
Forward Number (PSTN To VoIP) Forward Password (PSTN To VoIP) Dial Plan(PSTN to VoIP)	





Call Forward (From VoIP to GSM) Forward Number

Enter this field to forward all incoming VoIP calls to this number (PSTN or Mobile). Using "," to add a 500ms delay to the dialing sequence. If this field is blank, calls will not be forwarded. The GSMGW1 Gateway answers an incoming VoIP call and generates a dial tone. The caller can then dial a number (PSTN or Mobile) desired. Please see below if the

Forward Password

This field sets the password protection for using the GSM connection. If a password is entered, the GSMGW1 Gateway will generate an indication tone and wait for the call to dial the **Call Forward (From GSM to VoIP)**

Forward Number

Forward all incoming calls from the GSM connection to the VoIP number specified in this field. Forward Password is not required once this field is set. If this field is blank, the GSMGW1 answers an incoming GSM calls and then generates the VoIP dial tone. Please see below if the Forward Password is set. The caller can then dial a VoIP number manually. At the end, a pound (#) can be dialed to activate the dialing of the VoIP number immediately. If not, the VoIP number is dialed after a preset timeout.

Forward Password

This field sets the password protection for incoming GSM calls. If a password is entered, the GSMGW1 Gateway will generate an indication tone after answering an incoming call. The caller is then ready to dial the password. Once the password is correctly entered, the GSMGW1 Gateway generates a VoIP dial tone and waits for the caller to dial a VoIP number.

3.6 Gain Settings...

A hidden webpage is provided to set the receiving and transmit gains of VoIP Chunnel. The URL link is: *http://xxx.xxx.xxx/default/en_US/gain.html*

THIS PAGE IS INTENDED FOR AN EXPERIENCED USER OR AN ADMINISTRATOR ONLY. PLEASE SET THE GAINS WITH CAUTIONS.

GSMGW1				简体中文
	Gain Settings			
	Line 1			
	Line 1 Output Gain	0	~	
	Line 1 Input Gain	+2	~	
	Save Reset			







3.7 Network Configuration

Click on "Network" tab in the left menu column to configure the LAN and PC ports.

Network Configuration				
LAN Port	PPPoE 💌	PC Port	Static IP 🛛 🗸	
802.1q VLAN	⊙ Enable		Advance>>	
	Advance>>			

3.7.1 LAN Port

Three LAN Port modes are supported: DHCP, Static IP, PPPoE.

Network Configuration		
LAN Port	PPP0E	
User name	DHCP Static IP	
Password	PPPoE	
802.1q VLAN	⊙Enable ○Disable	
VLAN Id		
VLAN QoS		
	Advance<<	
Ethernet(MAC) Address		
IP Broadcast Address		

1) DHCP

Choose **DHCP** if a local DHCP host is available. This allows the GSMGW1 Gateway to obtain network information (IP Address, Subnet Mask, Default Route, Primary DNS, Secondary DNS, and other DHCP options) from the DHCP host.

2) Static IP

Network Configuration		
LAN Port	Static IP 🛛 👻	
IP Address		
Subnet Mask(optional)		
Default Route		
Primary DNS		
Secondary DNS(optional)		

Choose Static IP if your network topology requires. Please fill in Fill in the IP Address, Subnet Mask, Default Route, Primary DNS, and Secondary DNS (optional) as provided by your network administrator.

3) PPPoE



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Network Configuration		
LAN Port	PPPoE	*
User name		
Password		

PPPoE is a common dial up method for you network modem (Cable / xDSLs). Choose this if your network environment requires. Enter the **User Name** and **Password** as provided by your ISP.

4) 802.1q VLAN

This QoS feature requires your QoS support of your network to improve voice data traffics. Please consult your network administrator for proper settings.

5) Advanced...

	Advance<<
Ethernet(MAC) Address	
IP Broadcast	
Address	

The **Advanced** settings allow the user to set the broadcast address and to clone a MAC address instead of using the factory preset MAC address. Please consult your network administrator for further information.

3.7.2 PC port configurations

The PC Port allows addition network devices to be attached behind the GSMGW1 Gateway. It offers both Bridge and Static IP modes to meet your network topology. It is factory preset to the Static IP mode with the IP address 192.168.8.1.

PC Port	Static IP 🛛 🗸	
IP Address	192.168.8.1	
Subnet Mask	255.255.255.0	
DHCP Server	💿 Enable 🔘 Disable	
Starting Address	192.168.8.150	
Ending Address	192.168.8.200	
Static DNS(optional)		
	Advanced>>	

1) Bridge Mode

Select **Bridge** mode if your network topology requires the network devices (PC or others) to be in the same network segment as the GSMGW1 Gateway. In this case, the GSMGW1 Gateway functions as an Ethernet Switch.





2) Static IP Mode (Default Setting)

Select **Static IP** mode for a new network segment for the network devices behind the GSMGW1 Gateway. In this case, the GSMGW1 Gateway functions as an Ethernet Router. Fill in the **IP Address** field with a new segment address that is different from that for the LAN port. Please select the **Subnet Mask** accordingly. A commonly used value is 255.255.255.0.

PC Port	Static IP 🗸 🗸
IP Address	
Subnet Mask	
DHCP Server	O Enable 💿 Disable

3) Enable the **DHCP Server** if you want the GSMGW1 Gateway functions as a local DHCP host for the PC segment. This will enables the GSMGW1 Gateway to assign IP Addresses to network devices that are attached to the PC port segment.

DHCP Server	💿 Enable 🔘 Disable
Starting Address	
Ending Address	
Static DNS(optional)	

Specify the Starting Address. Ending Address, and Static DNS accordingly.

4) Advanced...

The **Advanced** settings allow the user to set the broadcast address and to clone a MAC address instead of using the factory preset MAC address. Please consult your network administrator for further information.

	Advance<<
Ethernet(MAC) Address IP Broadcast Address	

3.8 Save Configuration

To confirm and commit all changes made, click on the **Save Changes** tab. Otherwise, all changes will be discarded. Once all changes are saved, the following screen message is displayed.

Configuration saved!
确定

3.9 Discard Changes

To discard all changes made, click on the **Discard Changes** tab.





3.10 Tools Menu

Select the **Tools** to access the following functions: **Online Upgrade**, **Change Password**, **Reset Config**, and **Reboot**.

Status	Online Upgrade
Configurations	Last Upgrade Time:
Tools	Current Version: GHS-3.01
	Upgrade Site: Start
Online Upgrade	
Change Password	
Decet Config	
Reset Coning	
Reboot	

3.10.1 Online Upgrade

To perform a firmware upgrade, select the **Online Upgrade** tab to access the page below.

01	
	Start
	01

Enter the update link as provided by your service provider. A sample link is: <u>http://202.155.200.154/update/A34HS-3.07-18.pkg</u> Please contact re-seller !!! Click the **Start** button to start the firmware upgrade. **WARNING: POWER SHUTDOWN / FAILURE DURING FIRMWARE UPGRADE MAY PERMINENTLY DAMAGE THE GSMGW1 GATEWAY.**

3.10.2 Change Password

Click on the **Change Password** tab to access the page below.

User Level		
New Password:		
Confirm Password:		Change
Administration Lev	vel	
New Password:		
Confirm Password:		Change

A) User Password

This is the password for the user name/ID "user". The default password is "1234".



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This user name is limited to access the Network Configuration menu.

B) Administrator Password (default: admin)

This is the password for the user name/ID "**admin**". The default password is "**admin**". This user name allows full access to all configuration settings available.

3.10.3 Reset Configuration

Click on the **Reset Config** tab to reset the GSMGW1 Gateway to its factory default settings.

3.10.4 Reboot the Device

Click on the **Reboot** tab to reboot the GSMGW1 Gateway. The web page is then not accessible until the device completes the reboot process.