

# G200S VoIP Gateway User Manual





# Safety Notices

1. Please use the power adapter that this device specified. If you have to use other manufacturers 'power adapter because of the special circumstances, please make sure the voltage and current provided in accordance with the provisions of this device. At the same time, we device you to use the power adapter which passed the safety certification devices, otherwise may cause fire or get an electric shock. When using this device, please do not damage the power cord, forcibly twisted, stretch pull or strapping, it cannot be pressured under heavy weights or clipped in the goods , or that may cause the power cord is damaged, and then resulting in fire or electric shock.

2. Before you use this device, please confirm the temperature and humidity of environment that the device is working in conform to what it needs.(If you move this device from the air conditioning room to natural temperature environment, the device may cause surface or internal components produce condense water vapor, please wait until this device natural drying and then open the power to make the device to work.)

3. Non technical service personnel must not remove or repair the device, otherwise improper repair or failure may cause electric shock, fire, etc, and lead to injury accident, your device warranty also will be invalid.

4. Please do not put your fingers, pins, wire or other metal objects, foreign body in the vents and gaps. It may be the cause of the current through the metal or foreign body, then make an electric shock, and lead to injury accident. If foreign bodies or a similar object fall into the device, please stop using it.

5. Please do not discard or put the device package that be packed in plastic bags on where the young children can get it, if the young children with them on the head, it may block their nose and mouth, thus lead to suffocation.

6. Please operate this device with correct operational method and position, if you use this device in bad posture for a long time, there may be some effect on your health.

7. Please use the device according to the indicating method of this user manual, otherwise may damage the device.



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## I. About Device

G200S is a new VoIP gateway, its core part is a proven solution for VOIP, and so the performance is stable and reliable. Compact appearance, intelligent software and simple interface, making IP gateway no longer limited to enterprise applications, but also for ordinary home users.

## **1. Device Appearance**



### 2. Indicator Lights Description

![](_page_4_Figure_6.jpeg)

After you insert the 12V DC power adapter to this device, power light starts to work, analog phone light comes on, then off! Registered light twinkles for a moment, WAN light and LAN light will be twinkling and then enter standby mode, when you pick up the analog phone, the analog phone light will keep on, when you hang up, the light off!

![](_page_5_Picture_0.jpeg)

Indicator lights	Description	Function		
	Dowerlight	off	Power is invalid.	
	Power Light	on	Power supply is normal.	
		off	SIP is not registered.	
$\bigcirc$	SIP Registered Light	twinkle	SIP registration is failed.	
$\smile$		on	SIP registration is successful.	
		off	LAN port is not connected.	
	LAN Light	twinkle	LAN port is transmitting data.	
		on	LAN port connection is normal.	
		off	WAN port is not connected.	
	WAN Light	Twinkle	WAN port is transmitting data.	
		on	WAN port connection is normal.	
	Analog Phone Light	off	Phone is in standby or not connected.	
$\bigcirc$		on	Phone is being off hook.	

## 3. Interface and Buttons Description

![](_page_5_Figure_3.jpeg)

![](_page_5_Figure_4.jpeg)

Description	Function
RST	Restore Default button. When the device is working properly, if you press this button with a sharp object (such as a pencil) until the CPU fast twinkling (about 5 seconds). Restore function will take effect after you release it.
POWER PORT	Connecting to a power source.
LAN PORT	Connecting to a computer or a PBX and so on.
WAN PORT	Connecting to the network.
FXS1	Connecting to the analog phone.
FXS2	Connecting to the analog phone.

![](_page_6_Picture_0.jpeg)

## **II. Getting Started**

Before you start using the G200S\_VoIP gateway, please make the following installation:

### 1. Connecting the Power and the Network

#### 1) Connecting the Network

Before this step, please make sure your environment can satisfy the requirement of broadband network access.

#### a) Broadband Router

Please connect one end of the network cable to the device's WAN port and the other end connect to your broadband router's LAN port. Now you have completed the network hardware connection. In most of the cases, you need to set your device's network as the DHCP mode. (The default mode of the device is DHCP)

![](_page_6_Figure_8.jpeg)

#### b) No Broadband Router

Please connect one end of the network cable to the device's WAN port and the other end connect to your broadband modem LAN port. Now you have completed network's hardware connection. In most of the cases, if you are using TV cable broadband, you need to set your device's network as the DHCP mode; If you are using ADSL, you'll need to set your device's network as the PPPoE mode. Detailed setting methods, please refer to the **IV. Web Configuration**.

#### 2) Connecting the Power

Before proceeding with this step, please make sure your power connector and electrical outlet for the agreement, at the same time, the voltage and current are also conform to what the device need.

- a) Put the DC port connect to the power port that on the back of the device.
- b) Put the AC adapter plug connect to an electrical outlet, the device starts to boot.
- c) At this point, all of your indicator lights (except the power light) will twinkle together. After the boot is completed, the indicator lights will be on according to your current configuration. (If your lights is unnormal, we need to further configure your network online mode)
- d) If the device has landed on the server, you can start a call right now.

![](_page_7_Picture_0.jpeg)

## **III. Basic Phone Operation**

### 1. Call Transfer

a) Blind Transfer:

During a conversation, you press FLASH (Flash) button, enter \* and the number you want to transfer, then press [#] key to confirm, you can transfer the current call to the third party. (In order to use the feature, you must enable the gateway Call Waiting function and Call Transfer function)

b) Attended Transfer:

During a conversation, you press FLASH (Flash) button, enter the number you want to transfer ,wait until telephone connected, hang up directly, you can transfer success. (In order to use the feature, you must enable the gateway Call Waiting function and Call Transfer function)

**%:** 1. Call Transfer function is implemented under certain condition: there is one way of the two calls is in idle state.

2. The call between Gateway (transfer side) and phone A is established, then the gateway and the phone C start another call, now you hang up the phone A, the gateway still can initiate a transfer.

3. Only your network phone traffic service providers support the (RFC3515), can this function work properly

### 2. Call Hold

#### • Call Hold and Set Aside

During a conversation, you can press FLASH button, then enter the number to dial and the [#] key to confirm. You can keep your current call and build the third party at the same time. If you press the FLASH (flash) button again, you can switch back. You can only talk with one side while other parties cannot hear your conversation or talk with you. During a conversation, if you press the [\*] button, the device will enter the three-party call mode. (To use this feature, you must enable the Call Waiting function of the gateway. To achieve the three-way calling mode, you must enable the Gateway Three Way Call function)

#### • Call Hold and Accept Call Waiting

During a normal conversation, if there is third-party dial-in, the handset will be heard beep ~ beep ~ tips, you can use FLASH (flash) button to accept the call waiting. If you press this button again, you can switch back. You can only talk with one side while other parties cannot hear your conversation or talk with you. (To use this feature, you must enable the Call Waiting function of the gateway)

![](_page_8_Picture_0.jpeg)

## **IV. Web Configuration**

### 1. Ways to Configure

- G100S\_VoIP gateway offers two different configure ways to different users:
- Use web browser: the computer users who are familiar with the operation of computers.

(Recommended use)

• Use the telnet tool: command line users.

## 2. Password Configuration

The setting of the device's browser and command-line can be divided into two login modes: user mode and supervisor mode, under the manager mode, you can view and edit all of the options; while the <**Auto Provision**> option cannot be viewed under the user mode.

When a tip: 'Please enter your password' appeared on the device, you enter different information will into different modes:

- User mode:
  - Username: admin
  - Password: admin
- Manager mode:
  - Username: root
  - Password: admin

### 3. Browser Configuration

When the device and computer are connected to the network successfully, you enter the device WAN port IP address in the browser (gateway IP address can be get by dialing \* 111 #) http://xxx.xxx.xxx / to see the web management interface login page (as shown below). Enter username and password , click **[Login]** button ,you will enter the setting pages .

Username:	
Password:	

ſ.	~	~			
L	0	g	l	ľ	1

If you have not save your settings, the settings will be restored to the previous state unchanged when you boot phone next time .In order to save your settings, please click the **Save** button that belongs to configuration settings in the **System**, after this process, your device configuration will take effect immediately without reboot again.

![](_page_9_Picture_0.jpeg)

## 4. WEB Pages Function Explanation

## (1) Status

## a) Overview

Status	~
» Overview	
» Routes	
» System Log	
System	>
Network	>
VolP	>
Phone	>
Logout	>

System	
Model	G200S
Hardware Version	v1.0
Firmware Version	1.2.1220
Mac Address	00:a8:59:dc:f0:7e
Serial Number	20160721DCF07E
Local Time	Thu Aug 10 17:55:52 2000
Uptime	0h 6m 58s
Load Average	0.61, 0.50, 0.28
MEM Info	22508 kB / 61752 kB (36%)
Network	
WAN	Status
Connection Type	dhcp
IP Address	172.18.2.164
Subnet mask	255.255.0.0
Default Gateway	172.18.1.1
Primary DNS	172.18.1.1

Line Status				
	Line 1			
Account		N/A		
Server		N/A		
	Line 2			
Account		N/A		
Server		N/A		
	Port 1			
DND		OFF		
	Port 2			
DND		OFF		

Overview				
Name	Explanation			
System				
Model	Displays device model.			
Hardware Version	Displays device hardware version.			
Firmware Version	Displays device software firmware version number.			
MAC Address	Displays the current MAC address.			
Serial Name	Displays device serial number.			
Local Time	Displays the current system time			
Uptime	Displays device runtime			

![](_page_10_Picture_0.jpeg)

Load Average	Displays the current average load value				
MEM Info	Displays the current memory status				
Network (WAN Status)					
Connection Type	Displays the current networking way.				
IP Address	Displays the current IP address.				
Subnet Mask	Displays the current subnet mask.				
Default Gateway	Displays the default gateway.				
Primary DNS	Displays the primary DNS.				
Line status	Line status				
Displays the current SIP line 1-2 registries number 🕔 server and status.					
	Open this option, any dial-in call will be blocked, the caller will be prompted that the				
טאט	device cannot be used, but you can establish a call with the device. (Port1 or Port2)				

#### b) Routes

With this function, you can see the ARP table in the routes. The hosts IP MAC information that had connected with the device recently will be stored in the ARP list.

🗊 Status	~	Routes				
» Overview		The following rules are cu	urrently active on this syst	em.		
» Routes		ARP				
» System Log		IPv4-Addr	ess	MAC-Address	Interface	
		172.16.1	1	00:25:9c:52:02:36	br-	wan
System	>	172.16.7.9	90	08:62:66:2d:c3:59	br-	wan
Network	>					
VoIP	>	Active IPv4-Routes				
Phone	>					
	-	Network	Target	IPv4-Gateway	Metric	Table
Logout	>	wan	0.0.0/0	172.16.1.1	0	main
		wan	172.16.0.0/16		0	main

## c) System Log

It displays activity information of the system.

![](_page_11_Picture_0.jpeg)

## (2) System

## a) System

General Setting	35	
Status >	System	
System 🗸	Here you can configure the basic aspects of you	r device like its hostname or the timezone.
» System	System Properties	
» Administration	Local Time	Fri Jul 29 17:34:28 2016 Sync with browser
» Time Synchronization	Timezone	VoIP Asia/Shanghai ✓
» Backup / Flash Firmware		
» Auto Provision		Reset Save & Apply
» Debug		
» Reboot		
Network >		

Name	Explanation
General Settings	
Local Time	Displays the current system time
Hostname	Name of the device, similar to the computer' name. The default is VoIP, you can
	modify it by yourself.
Timezone	Set the time zone of the area where you are.

![](_page_12_Picture_0.jpeg)

## Logging

🔲 Status >	System	
System 🗸 🗸	Here you can configure the basic aspects of your dev	vice like its hostname or the timezone.
» System	General Settings Logging Language	
» Administration	System log buffer size	16 😰 kiB
» Time Synchronization	External system log server	0.0.0
» Backup / Flash Firmware	Log output level	Debug V
» Auto Provision	Cron Log Level	Normal
» Debug		
» Reboot		Reset Save & Apply
Network >		

Name	Explanation
Logging	
System log buffer size	Set the log buffer size.
External log system server	Set the address of the external log server.
External Log server port	Set the port of the external log server.
Log output level	Set the level of log output.
Cron log level	Set the level of Cron log.

#### Language

In this interface , you can configure the language that the device currently uses.

Status >	System
System 🗸	Here you can configure the basic aspects of your device like its hostname or the timezone.
» System	System Properties
» Administration	Language English
» Time Synchronization	
» Backup / Flash Firmware	Reset Save & Apply
» Auto Provision	

![](_page_13_Picture_0.jpeg)

## b) Administration

In this interface, you can modify the current user's password.

Status >	Router Password		
🔗 System 🗸	Changes the administrator password for ac	ccessing the device	
» System	Password	<b>2</b>	
» Administration	Confirmation	<i>»</i>	
» Time Synchronization			
» Backup / Flash Firmware		Reset Save & App	ly

## c) Time Synchronization

Status >	Time Synchronisation		
System 🗸	Synchronizes the system time		
» System	General		
» Administration	Current system time Update interval (in seconds)	Fri Jul 29 17:37:04 2016 600	
» Time Synchronization	Count of time measurements	<pre>@ empty = infinite</pre>	
» Backup / Flash Firmware			
» Auto Provision	Clock Adjustment		
» Debug	Offset frequency	0	
» Reboot			
Network >	Time Servers		
VoIP >		Dest	
Phone >		FUIL	
Logout >	cn.pool.ntp.org	123	Delete
	Add		
			Posot Save & Apply

![](_page_14_Picture_0.jpeg)

Time Synchronization			
Name	Explanation		
General	General		
Current system time	Displays the current system time.		
	How long time the device request to the server to update, the default		
opuale interval(in seconds)	time less than 600 seconds.		
Count of time	Cat the number of time, space is unlimited length		
measurements	Set the number of time, space is uninnited length.		
Clock Adjustment			
Offset frequency	Set the time calibration offset value		
Time Servers			
Set the server address and ports updated by time.			

## d) Backup/Flash Firmware

🗊 Status >	Flash operations
System 🗸	Actions Configuration
» System	Backup / Restore
» Administration	nitial state, click "Perform reset" (only possible with squashfs images).
» Time Synchronization	Download backup:     Generate archive       Reset to defaults:     Perform reset
» Backup / Flash Firmware	To restore configuration files, you can upload a previously generated backup archive here.
» Auto Provision	Restore backup:浏览 Upload archive
» Debug	Flash new firmware image
» Reboot	Upload a sysupgrade-compatible image here to replace the running firmware. Check "Keep settings" to retain th e current configuration (requires an OpenWrt compatible firmware image).
Network >	Keep settings:
VoIP >	Image: 浏览 Flash image

Name	Explanation
Backup/Restore	
Backup / Restore the current system configuration file or reset PandoraBox. (Squashfs only valid	
firmware)	

![](_page_15_Picture_0.jpeg)

Flash new firmware image	
Keep settings	Preserving configuration that currently set. If the option is not selected, the device
	will automatically restore to factory configuration after upgrade.
Image	Selects the firmware you need to update, then click < <b>Flash image</b> >.It is set up.

🗊 Status >	Backup Configuration
System 🗸	Actions Configuration
» System	Backup / Restore
» Administration	Export config.txt
» Time Synchronization	import config
» Backup / Flash Firmware	کتار لائے است در دور تر میں اور
» Auto Provision	Import comig.oxt

Name	Explanation
Configuration	
Export current device configuration file.	

## e) Auto Provision

Status > Cor	nmon Settings	
System 🗸	Configuration File Version	2.0002
» System	Server Address Username	0.0.0.0 user
» Administration	Password Configuration File Name	
» Time Synchronization	Encryption Key Protocol Type	FTP 🗸
» Backup / Flash Firmware	Update Interval	1 Ø Hour
» Auto Provision	Update Type Check Digest	Disable 🗸
» Debug	Enable DHCP Option 66	
» Reboot		

![](_page_16_Picture_0.jpeg)

Г

Common Setting	
Name	Explanation
	Displays the version number of current system configuration file, if the Terminal
	founds the CFG configuration file that has downloaded same with configuration file
Configuration	that are running, the device will not run it. Or if the terminal matches configuration
File Version	file via Digest verification way, as long as the configuration on the server has been
	modified, or the terminal configure do not match with the configuration on the
	server, the terminal will to download and update.
	Configures the FTP server address. The server address can be a IP form, such as
	192.168.1.1, it may in the form of domain names also, such as <u>ftp.domain.com</u> . And
	the system also supports server setting subdirectory function, such as the system can
Server Address	configure server address as 192.168.1.1/ftp/Config/form or
	ftp.domain.com/ftp/config form .It means, the server address to access is 192.168.1.1
	or <a href="ftp.domain.com">ftp.domain.com</a> , the file storage path is / ftp / Config /. Subdirectory can end
	without "/".
	Configures username of FTP server; TFTP protocol need not configuration; if you are
Username	using ftp protocol download mode, here is no need to fill, the default is the default
	user anonymous FTP.
Password	Configures FTP server user's password.
Configuration	Configures the name of these configuration file need to upgrade; if you use the
File Name	automatic upgrade feature , this project configuration is empty generally, so our
	equipment will use its own MAC address as the file name to get the file on the server.
Encryption Key	If the configuration file need to update has been encrypted, you need to enter the
	encryption password in this configuration.
Protocol Type	Selects the server type. There are three types : FTP, TFTP and HTTP
Update Interval	Configures the interval upgrade time in hours.
	Automatic Update Types:
Update Type	1. Update after rebooting.
	2. Deactivated.
	3. Updated regularly.(How often interval updated)
Check Digest	Configure whether to use Digest mode.
Enable DHCP	Enable/Disable DHCP option 66
OPTION 66	

![](_page_17_Picture_0.jpeg)

» Backup / Flash Fi	rmware	Update Interval	1 <sup>(2)</sup> Hour
		Update Type	Disable 🗸
» Auto Provision		Check Digest	$\checkmark$
» Debug		Enable DHCP Option 66	
» Reboot			
Network	>	TR069 Settings	
VoIP	>	Enable TR069	
Phone	>	ACS Server Type ACS Server URL ACS User	0.0.0
Logout	>	ACS Password TR069 Auto Login	<i></i> <i>⊘</i>
		INFORM Sending Period	3600 ② Seconds
			Reset Save & Apply

TR069 Setting		
Name	Explanation	
Enable TR069	Enable/Disable TR069	
ACS Server type	Used for choosing ACS server type, the terminal supports telecommunications and	
	the general two kinds ACS server currently.	
ACS Server URL	Enter the ACS server address.	
ACS User	Enter the ACS server verification username.	
ACS Password	Enter the ACS server verification user password.	
TR069 Auto Iogin	If you selected the automatic login, after rebooting the phone, you will not be prompted to enter username and password, but the correct username and password you entered before link to the ACS server.	
INFORM Sending Period	Check the system every 6 minutes by default.	

![](_page_18_Picture_0.jpeg)

### f) Debug

🗊 Status >	Syslog Set		
System 🗸	Server IP	0.0.0.0	
» System	Server Port Enable Syslog		
» Administration			
» Time Synchronization	CM Log Level	None	
» Backup / Flash Firmware	SIP Log Level	None	
» Auto Provision		Reset	Save & Apply
» Debug			
» Reboot			

Syslog Set		
Name	Explanation	
Server IP	Configures syslog server IP or domain name.	
Server port	Configures syslog server ports.	
Enable Syslog	Configures to enable/disable syslog.	
CM Log Level	Configures the MGR Log Level.	
SIP Log Level	Configures the SIP Log Level.	

#### g) Reboot

Clicking the **<Execute Restart>** button, the device will reboot.

Rebooting will not lost the saved configuration ,in the process of rebooting, the network connection will be interrupted temporarily .

Note:During the rebooting, please ensure stable power supply, avoid forced interruption.

![](_page_19_Picture_0.jpeg)

🗊 Status >	Reboot
System 🗸 🗸	Reboots the operating system of your device
» System	Perform reboot
» Administration	
» Time Synchronization	
» Backup / Flash Firmware	
» Auto Provision	
» Debug	
» Reboot	
Network >	

## (3) Network

## a) WAN

Status >	Global Network Options		
System >			
Network	Bridge Mode Save & Apply		
» WAN			
» Static Routes	Network Configuration		
» Diagnostics	Basic Settings VPN Settings		
» QoS	Connection Type DHCP V		
VoIP >	Save & Apply		
WAN			
Name	Explanation		
Global Network	Configuration		
Bridge Mode	Use the bridge mode (transparent mode): Bridge mode will make the device no longer set the IP address for achieving the LAN port, LAN port and WAN port will be connected to the same network.		

![](_page_20_Picture_0.jpeg)

#### Network Configuration (Basic Settings)

The mode of device connects to the network .According to the network environment, you need to select the appropriate network mode. The device offers three modes:

Static IP	If your ISP server provides a fixed IP address, you can select this mode. After	
	selecting, you must fill in a static table: Static IP address / Subnet Mask / Gateway /	
	DNS and other related information. If you do not know this information, contact your	
	ISP provider or network administrator for assistance.	
DHCP	When you select this mode, the network-related information will be automatically	
	obtained from the DHCP server, you do not need to manually enter these fields.	
PPPoE	When you select this mode, you must input your ADSL account and password	
Use Custom	When you select this mode, you must enter the DNS server address. If you have n	
DNS	IS the device will get the DNS server address automatically.	

Status	>	<b>Global Network Options</b>	
System	>		
Network	~	Bridge Mode	
» WAN			
» Static Routes		Network Configuration	
» Diagnostics		Basic Settings VPN Settings	
» QoS		Protocol	PPTP 🔽
VoIP	>	Server Address  PAP/CHAP username PAP/CHAP password	
Phone	>	Bring up on boot Save & Apply	

Name	Explanation	
Network configuration (VPN Settings)		
Drotocol	Choose the PPTP type, the L2T represents the VPN L2TP, and the PPTP represents	
Protocol	VPN PPTP, you must only choose one of them as the current state.	
Server Address	Configures the VPN server address.	
РАР/СНАР	Configures the V/DN conversion	
Username Configures the VPN server username.		
PAP/CHAP Passwor	Configures the VPN server password.	
Bring Up on Boot	Configures VPN settings auto start after rebooting.	

![](_page_21_Picture_0.jpeg)

### b) Static Routes

Status	>	Routes					
System	>	Routes specify over whi	ch interface and gateway	a certain host or network car	be reached.		
Network	~	Static IPv4 Routes	5				
» WAN		Interface 📻	Target	IPv4-Netmask	IPv4-Gateway	Metric	мти
» Static Routes			Host-IP or Network	if target is a network			
» Diagnostics		Add	This s	section contains no values yet			
» QoS							
VoIP	>				Re	eset Sav	/e & Apply

Name	Explanation	
Interface	Configures the interfaces sent out by packets.	
Target	onfigures the destination IP address that packets needs to reach	
IPv4-Netmask	Configures the subnet mask of target IP address	
	When you Specify an IP address, the device will next transmit the data packets meet	
IPv4-Galeway	the requirements to this address.	
Metric	The maximum steps number send by packets.	
MTU	The maximum bytes number send by packets.	

### c) Diagnostics

#### Diagnostics

#### ♦ Ping Communication Test

Enter the destination address, which can be a legitimate IP address or a legitimate domain. Click the <**Ping**> button, the device will send a ping packet to detect whether the destination can be reached, and the results will appear in the box below.

![](_page_22_Picture_0.jpeg)

![](_page_22_Figure_1.jpeg)

#### ♦ Traceroute Detection

Enter the destination address, which can be a legitimate IP address or a legitimate domain. Click <**Traceroute**> button, the device will send tracert packets to detect through which routes to arrive at the destination address, and the test results will be displayed in the box below.

Status	>	Diagnostics
System	>	Network Utilities
Network	~	bing.com 172.16.2.44.com bing.com ×
» WAN		Ping Traceroute Nslookup
» Static Routes		traceroute to 172.16.2.44.com (184.168.221.58), 30 hops max, 38 byte packets
» Diagnostics		2 124.126.160.1 4.140 ms 3 * 4 219.141.162.181 2.500 ms
» QoS		5 202.97.53.98 7.680 ms 6 202.97.58.122 8.060 ms 7 202.97.52.166 157.720 ms 8 202.97 50 30 163 480 ms
VoIP	>	9 4.53.210.109 263.180 ms 10 4.69.144.207 259.380 ms 11 4.69.144.15 163.840 ms
Phone	>	12 4.53.230.102 264.320 ms 13 * 14 *

#### ♦ Network Detection

Enter the destination address, which can be a legitimate IP address or a legitimate domain . Click <**Traceroute**> button, the test results will be displayed in the box below.

![](_page_23_Picture_0.jpeg)

Status	>	Diagnostics
System	>	Network Utilities
Network	~	bing.com 172.16.2.44.com
» WAN		Ping Traceroute Nslookup
» Static Routes		Server: 8.8.8.8 Address 1: 8.8.8.8 google-public-dns-a.google.com
» Diagnostics		Name: 172.16.2.44.com Address 1: 184.168.221.58 ip-184-168-221-58.ip.secureserver.net
» QoS		Network Packets Capture

#### > Network Packets Capture

When you need to catch the packets that through the device wan port, click **<Start>** button, and then choose to save or open according to the dialog box .

![](_page_23_Figure_4.jpeg)

![](_page_24_Picture_0.jpeg)

d) QoS

🗊 Status	>	Quality	of Service						
System	>	With QoS you	can prioritize ne	etwork traf	fic selected	by addresses, po	orts or services.		
Network	~	Interfaces							
» WAN		WAN							Delete
» Static Routes			Enable Classification	aroup		default	$\checkmark$		
- Static Hoates			Calculate over	erhead					
» Diagnostics			Half-duplex	lineau					
			Download sp	eed (kbit/	s)	1024			
» QoS			Upload spee	d (kbit/s)		128			
VoIP	>								
Phone	>		]			Add			
Logout	>	Classificat	ion Rules						
		Target Sour	ce ho Destinatio t n host	Service	Protocol	Ports	Number of byte s	Comment	Sort
		<b>~</b> a		all 🗸	a 🗸	22,53 🗸		ssh, dns	<ul><li>♦</li><li>Delete</li></ul>
		✓ a	✓ al ✓	all 🗸	TV	20,21,2 🗸		ftp, smtp	Delete
		✓ a	✓ al ✓	all 🗸	a 🗸	5190 🗸		AOL, iCł	Delete
		Add							
								Rese	t Save & Apply

QOS	
Name	Explanation
WAN	
Enable	Set if the current interface to enable QoS service
Classification	Selects the interface classification group
Group	Selects the interface classification group
Calculate	Set whether to enable the calculation of overhead functions for calculating packets,
Overhead	reduce upload and download rates to avoid link saturation
Half-duplex	Set whether to enable half duplex mode.
Download Speed	Set download speed of the interface, units (kbit / s)
Upload Speed	Set upload speed of the interface units ,(kbit / s)

![](_page_25_Picture_0.jpeg)

<b>Classification Rule</b>	S	
Target	Selects the target priority	
Source Host	Selects a specific IP as the source host	
<b>Destination Host</b>	Selects a specific IP as the destination host	
Server	Selects the application service of what the category groups need .	
Protocol	The protocol can be selected as :all optional , TCP, UDP, IGMP, etc.	
Ports	The port number can be selected as all or a custom specified port.	
Number of Bytes	Fills the data need to be limited.	
Comment	Fills comment information.	
	Set the taxonomic groups sorting , when the targets have same level, the device will	
5011	prefer to use the preceding rules	

## (4) VoIP

## a) Line1 & Line2

Here, you can configure the SIP server of line 1 and line 2.

Status	>	Line Settings	
System	>	Status	
Network	>	Unapplied	
		Settings	
S VolP	~	Basic Settings Advanced Settings	
» Line1		Server Address	
		Server Port	5060
» Line2		Authentication User	
		Authentication Password	<b>2</b>
» Common		Phone Number	
		Display Name	
» Dial Peer		Realm	
		Enable Registration	
Phone @	>	Port Select	Port 1
Logout	>		
3	-		Recet Save & Apply

Line Settings				
Name	Explanation			
Status				
SIP registration status	SIP registration status display: if the registration is successful, it will display 'Registered', while not			
successful will display 'F	successful will display 'FAILED'.			
Settings (Basic Setting	Settings (Basic Settings)			
	Configures Your SIP registration server address, supports these address in the			
Server Address	form of domain name.			

![](_page_26_Picture_0.jpeg)

Server Port	Configures SIP registration server signaling ports.	
Authentication User	Configures SIP registration account.	
Authentication	Configures SID registration recoverd	
Password	Configures SIP registration password.	
Dhone Number	Configures the number registered to the SIP server, if it is empty, will not	
Phone Number	initiate registration.	
	Configures the display name, when you make a call, the called party (did not	
Display Name	give you a name) can display the configuration parameters that be allowed to	
	input the English alphabet.	
	Configures SIP local domain name. If the server does not require local domain	
	SIP terminal is the specified domain, local domain can be configured with the	
Realm	address or domain name that same with server. To simplify user input, users	
	can do not enter the domain name, the system will automatically get the	
	server address to fill it as 'domain realm'.	
Enable registration	Configures enable/disable registration.	
Port Select Configure port selection (Port1 or Port2)		

Status	>	Line Settings		
System	>	Status		
Network	>	Unapplied		
VoIP	~	Basic Settings Advanced Settings		
» Line1		Outbound Proxy User Agent	Voip Phone 1.0	
» Line2		DTMF Type	RFC2833	
» Common		Server Type		
» Dial Peer		Forward Number		
Phone	>	MWI Number Outgoing Call Without Registration		
Logout	>	Blocking Anonymous Call Transfer Timeout	D Seconds	
		Response Single Codec		
		Use STUN		
		Registration Expiration	3600 ② Seconds	
		Keep Alive Interval	30 2 Seconds	

![](_page_27_Picture_0.jpeg)

	Seconds
Keep NAT Alive	$\checkmark$
Keep Authentication	
Enable Strict Proxy	
Enable DNS SRV	
Enable Rport	
Enable PRACK	
Long Contact	
Convert URI	
Enable Session Timer	
Subscribe Expiration	300 ② Seconds
Enable Subscribe	
Transportation Protocol	UDP 🗸
Auto TCP	
SIP Version	RFC3261
Local Port	5060
RFC Privacy Edition	NONE
Use Quote in Display Name	
Enable GRUU	
RTP Encryption Key	
RTP Encryption	

Reset Save & Apply

Line Settings			
Name	Explanation		
Settings (Advanced	Settings)		
User Agent	User Agent Terminal		
	Set DTMF transmit mode, there three types totally : The default is In-band		
	● In-band		
DTMF Type	• RFC2833		
	<ul> <li>SIP_INFO</li> </ul>		
	Different service providers may offer different modes.		
Server Type	Selects signaling encryption type or special server type		
Server Name	Names the server.		
	Selects call forward mode. Call Forward (off by default)		
	Off: close call forward		
	<ul> <li>Unconditional: inbound calls will be forwarded to the specified number</li> </ul>		
	• Busy: inbound calls will be forwarded to the specified number when the		
Forward Type	device is buys.		
	• No answer: Inbound calls have not been answered after specified time, will		
	be forwarded to the assigned number, during the this proceed, the device will		
	prompt a call.		
Forward Number	Configures the forward number.		

![](_page_28_Picture_0.jpeg)

MWI Number	Configures the MWI number, achieve Listen to achieve sip voicemail notification and listening.			
Outgoing Call Without Registration	If you configure this item, you can also call through a proxy server without registration.			
Blocking Anonymous Call	Configures blocking anonymous call.			
Transfer Timeout	In order to adapt a platform, when you make the attended transfer and hang up, the session will end after the expire time, the device will send 'bye' initiatively; the default is 0(when you hang up, the device will send a BYE to end the session immediately).			
Response Single Codec	As the called, only in response to the supported Codec.			
Use STUN	Configures enable/disable the SIP STUN.			
Registration Expiration	Configures the SIP server registration expiration time, defaults 3600 seconds. If the server requires registration expiration time is more than or less than the device configuration time, the device can be automatically changed as the server recommendation expiration time and re-register.			
Keep Alive Interval	Configures the server detection time interval, if the gateway opens SIP detection server function, the gateway will detect whether the server responds every configured time.			
Keep NAT Alive	Configures automatic detection server. Some servers prohibit the registration time is too short, but there have not packet (maintain the device terminal NAT actively) to send, you can open this function and set the interval to send this package is less than NAT duration of time.			
Keep Authentication	Enable/Disable Keep Authentication System will take the last authentication field which is passed the authentication by server to the request packet. It will decrease the server's repeat authorization work, if it is enable.			
Enable Strict Proxy	Matches with a special server (when the return message ,the device will use the other party source address ,no longer use the address in via field)			
Enable DNS SRV	When you open it ,the device will support RFC2782			
Enable Rport	Configures if the device support RFC3581,rport mechanism is used in the Intranet, need be supported by SIP server for maintaining the NAT connection of Intranet devices and Extranet devices			
Enable PRACK	Configures whether let the device support the SIP PRACK function (mostly used for ring tones), we recommend you to use the default configuration			
Long Contact	Configures Contact field carries more parameters; the item is used with SEM			

![](_page_29_Picture_0.jpeg)

	server.				
Convert URI	Convert # to %23 when URI is sending massage.				
Enable Session	Configures whether the device support rfc4028 function and refresh the SIP				
Timer	sessions function.				
Subscribe	Configures the offective time of subserintion				
Expiration	configures the effective time of subscription				
Fnahla Subseriba	After successful registration, subscription information can subscribe to others				
Enable Subscribe	state or voice mail, etc.				
Transportation	Configures the using transport protocol, TCD or UDD the default is UDD				
Protocol	configures the using transport protocol, TCP or ODP, the default is ODP				
	When the message body exceeds 1300 bytes ,the device will automatically use				
AutoTCP	the TCP transport protocol to guarantee the availability of transport				
	Configures the protocol version. When the device needs to communicate with				
SIP Version	the gateway like CISCO5300 which uses SIP1.0, you need to configure this item				
	into RFC2543, so it can communicate normally. The default is RFC3261.				
Local Port	Configures individual port of each line.				
	Configures whether you use anonymous security call out, it supports RFC3323				
RFC Privacy Edition	and RFC3325 .				
Use Quote in	When the device sending signaling, whether add the quotes before the display				
Display Name	name.				
Enable GRUU	Configures supporting GRUU				
RTP Encryption	Configures voice encryption key				
RTP Encryption	Configures whether to support voice encryption				

![](_page_30_Picture_0.jpeg)

## b) Common

🕞 Status	>	STUN Status
System	>	False
Network	>	
VoIP	~	STUN Settings
» Line1		Sapur Address
» Line2		Server Port 3478
» Common		Binding Period 50 Seconds
» Dial Peer		
Phone	>	SIP Settings
Logout	>	Registration Failure Retry Interval 30
		Sip Invite Restrict
		Receive Call Only from UA

Reset Save & Apply

Common			
Name	Explanation		
STUN Status			
Displays STUN pen	etrate judgment, true means STUN is penetrable, false means impenetrable.		
STUN Settings			
Server Address	Configures SIP STUN Server Address.		
Server Port	Configures SIP STUN server Port.		
	The interval that STUN detects NAT type ; When NAT finds a connection have no		
Binding Period	activity over a period of time, it will close the map, so you must send a packet out at		
	intervals to ensure keep alive.		
SIP Settings			
Registration			
Failure Retry	Configures how often the device initiate registration again after registration failure.		
Interval			
SIP Invite Restrict	Whether match Invite field strictly. If you selected this item, the SIP message via		
	field that the device received must begin with z9hG4k, or the device will not		
	respond to the SIP message received.		
	<b>NOTE</b> : This configuration will take effect in all SIP accounts.		

![](_page_31_Picture_0.jpeg)

 Receive Call Only
 Configures whether match UA strictly.

 from UA
 Configures whether match UA strictly.

### c) Dial Peer

Status	>	Dial Peer Tabl	e				
System	>	Number	Destination	Port	Alias	Suffix	Deleted Length
Network	>						
VoIP	~			5060		no suffix	0 Delete
» Line1		Add					
» Line2							Reset Save & Apply
» Common							
» Dial Peer							
Phone	>						

![](_page_32_Picture_0.jpeg)

## (5) Phone

## a) Audio

Status >	Audio Settings	
System >	Audio Codec 1 Type	G.711u 🗸
	Audio Codec 2 Type	G.711A 🗸
Network >	Audio Codec 3 Type	G.729
	Audio Codec 4 Type	G.726-32
VolP	Mic Gain	
🙆 Phone 🔹 🗸	Handset Volume	1 () (-2~+2)
» Audio	Fax Type	Т.38
» Port1	Caller ID Type	Bellcore FSK(US)
» Port2		
	I one Standard	China
» Call Feature	Hook Flash Min Time	200 (>=50ms)
» Dial rules	Hook Flash Max Time	800 @ (50-1000ms)
	SLIC Impedance	6000hm 🗸
Logout >	DTMF Payload Type	101 @ (96-127)
	Enable MAD	

Reset Save & Apply

Audio Settings(Port1 or Port2)			
Name	Explanation		
Audio Settings			
Audio Codec 1 Type	Selects DSP the first priority speech coding algorithm: G.711A/u,G.726-32, G.729		
Audio Codos 2 Tupo	Selects DSP the second priority speech coding algorithm: G.711A/u,G.726-32,		
Audio Codec 2 Type	G.729		
Audio Codec 3 Type	Selects DSP the third priority speech coding algorithm: G.711A/u,G.726-32, G.729		
Audio Codoc 4 Typo	Selects DSP the fourth priority speech coding algorithm: G.711A/u,G.726-32,		
Audio Codec 4 Type	G.729		
MIC Gain	Set the microphone volume level.		
Handset volume	Set the handset volume level.		
Fax Туре	Set the fax type.		
Caller ID Type	Set the PSTN phones that only support transfering Caller ID under DTMF mode.		
Tone Standard	Selects tone standard.		
Hook Flash Min	Set the minimum time of incerted enring detection		
Time	set the minimum time of inserted spring detection.		
Hook Flash Max	Sat the maximum time of incerted spring detection		
Time	Set the maximum time of inserted spring detection.		

![](_page_33_Picture_0.jpeg)

SLIC Impedance	Set subscriber line interface circuit impedance, the default is 600 ohms.		
DTMF Payload Type Effective load of dual-tone multifrequency.			
Enable VND	Silence detection; if it is enabled VAD, G.729 payload length cannot be set higher		
	than 20ms		

## b) Call Feature

On this page, you can set the hotline, call transfer, call waiting, three way call, black list, Blocking list and so on.

Status	>	Call Feature	
System	>	Hotline number	
Network	>	Warm Line Timeout	0 ② (0~9 seconds)
		No Answer Time	20 (0~60 seconds)
S VOIP	>	Do Not Disturb(DND)	
Phone	~	Blocking Outgoing Call	
		Enable Three Way Call	
» Audio		Enable Call Waiting	
		Enable Call Transfer	
» Call Feature			
» Port1		P2P IP Prefix	
		Accept Any Cell	
» Port2		Accept Any Can	
» Dial rules			
1	1		
Logout	>		Black List
			This section contains no values yet
		Add	
			Blocking List
			we are a construction of the construction of t
			Inis section contains no values yet
		Add	

Reset Save & Apply

Call Feature(Port1 or Port2)			
Name	xplanation		
Call Feature			
Hotline Number	Set the Hotline Number. If you set this number, as long as you goes off-hook, the device will automatically dial the hotline number and you cannot dial any other		

![](_page_34_Picture_0.jpeg)

number.			
Set automatically dial the hotline number time after off-hook .If you set this item			
as 0, device will call the hotline number immediately after you off-hook.			
Set the no answer time.			
If you selected this item, the device will reject any incoming calls, the caller will be			
prompted that the other side gateway is unavailable; your device can dial out			
without any effect.			
When it is enable, the device will send buys tone and prompt you to hang up when			
you off-hook and then dial.			
Enable three way call			
Enable call waiting			
Cat what has to enable call Transfer			
Set point-to-point IP call prefix, such as the other side IP is 192.168.1.119, then			
you defined here as 192.168.1., just dial # 119, can make a point-to-point IP call.			
When you selected this option, as long as the other side call you, the device will			
allow to make a conversation regardless of the number correctness.			

#### Black List

Add / Delete blacklist. If you do not want to answer a certain number ,can add the number to this list, when the number in black list call your device, the device will reject it;

It supports x format, that is, match to any one digit, such as 4xx represents these three digits number begin with 4 will be forbidden dial in;

Supported .formats, that is match to any length, including the null; Such as 6. represents the number more one digit and begin with 6 will be forbidden dial in;

If only allow a certain number /number segment to dial in ,you can configure white list rules to this list, the specific configuration should be "-" + "number", such as:

-4119

It represents other numbers are rejected to dial in except 4119;

Note: the white list must end with"."

**Blocking List** 

![](_page_35_Picture_0.jpeg)

Call limit, set the number prefix form : if 010 as configured number prefix, you will hear busy tone and be prompted to hang up when you dial 010,so you cannot continue dial: if 0 as configured number prefix, you will cannot dial all numbers begin with 0;

It supports x form, that is, match any one digit, such as 4xx represents all three digit number begin with 4 will be forbidden to dial out;

It supports . form ,that is match to any length, including the null; such as 6. Represents all number begin with 6 will be forbidden to dial out .

**Note:** Black List and Blocking List can match 10 records maximumly. If more than 10, it will prompt the list is full.

### c) Dial rules

🕞 Status	>	Dial rules	
System	>	Press # to invoke dialing	
Network	>	Use Fixed Length Fixed Length Invoke Calling after Timeout Timeout 5	
VoIP	>		
Phone	~	Seconds	
» Audio			
» Call Feature		Digital Rule table	
» Dial rules		Rules	
Logout	>		
		This section contains no values yet	
		Add	

Reset Save & Apply

Phone				
Name	Explanation			
Dial Rules				
Press # to invoke	Set the gateway to press the # key to finish receiving number.			
dialing				
Use Fixed Length	Set enable\disable use fixed length to dial.			
	Set the gateway to receive a fixed length number; for example, if you set it as 11,			
Fixed Length	when you finished dialing 11 digits number, the gateway automatically dial the 11			
	digits number.			

![](_page_36_Picture_0.jpeg)

Invoke Calling	Enable (disable calling after timeout		
After Timeout			
	Set the timeout calling length in seconds. Gateway default is 5 seconds, that is ,		
Timoout	when the gateway received a number, it will deem that you have finished dialing the		
Timeout	number after you have not continue dialing a number in 5 seconds , and it will dial		
	the received number.		

#### **Digital Rule Table**

The following is user-defined number-receiving rule table:

[] is defining the number range. It can be a range, can be separated by a comma, it can also be a digit of list;

x means can math to any one digit;

. means can match to any length, including the null;

Tn means the device will stop receiving number after n seconds. n is mandatory, the range is 0-9 seconds. Tn must be the last two digits setting. If Tn is not specified, it will be assumed as T0, the receiving number will end immediately.

As shown be	low:	
» Call Feature	Digital Rule table	
» Dial rules	Rules	
Logout	>	
	[1-8xxx]	Delete
	9xxxxxx	Delete
	911	Delete
	99T4	Delete
	9911x, T4 ×	Delete
	Add	
		Reset Save & Apply

[1-8]xxx, means all the four digits number(1000-8999) will be sent out immediately after receiving four digits number.

9xxxxxx, means all number begin with 9 will be send out immediately after receiving eight digits number.

911, means the 911 number will be sent immediately after dialing.

99T4, means when you finish dialing 99, the number will be send out after 4 seconds.

![](_page_37_Picture_0.jpeg)

9911x.T4, means when you finish dialing a number begin with 9911, 5 digits at least, it will be sent out after 4 seconds.

Other ways is unchanged.

**Note:** Press # to invoke dialing, Use fixed length, Invoke Calling after Timeout, Digital Rules table can be used simultaneously, as long as when you finish dialing a number, the number satisfies any of these judgments ,the device will end receiving number and send the number out.

#### (6) Logout

Click **[Logout]** button, you will exit web page. If you want to enter it next time, you need input username and password again.

### V. Appendix

#### 1. Specification

#### a) Hardware

Communication Protocol		SIP 2.0(RFC-3261)
Main Chip		MT7628
SDRAM		64MB
Flash		16MB
Ports	WAN	10/100BASE-T RJ-45 for WAN
	LAN	10/100BASE-T RJ-45 for LAN
	FXS	1 RJ11 for Phone(FXS)
	Power	DC12V In port
Adapter		Input: AC100~240V Output: DC12V 0.5A
Operation Temperature		0°C to 40°C
Relative Humidity		10% - 65%
Gateway Size		85mm x 67.6mm x 35mm
Packing Size		150mm x 125mm x 55mm

#### b) Voice Features

- Support SIP 2.0 (RFC3261) and SIP-related rfc
- Codec: G.711A / u, G.729, G.726-32k
- Echo cancellation: Support G.168
- Support voice volume adjustment, VAD, CNG
- NAT penetration, support STUN penetration type

![](_page_38_Picture_0.jpeg)

- SIP supports SIP domain, SIP authentication (none, basic, MD5), DNS, point to point(DIALPEER setting and IP call) call
- SIP can register two SIP accounts simultaneously. Using Pubic Server / Private server, you can make a call with any account
- Support call Line selection automatically, when the public server cannot connect with the device , the device can automatically switch to the private server call.
- DTMF mode support: in-band, RFC2833 and SIP INFO
- Support SIP application, including SIP Call forward / transfer / hold / waiting / 3 ways talking
- Call control features: flexible number receiving, support Hotline, reject blacklist, empty calling reject, limit call, DND, flexible dial peer configuration call rules
- support T.38 fax

#### c) Network Features

- Support PPPoE for xDSL, and support automatically redial after break.
- WAN / LAN port: support bridge mode or router mode
- gateway can do ping test via keyboard commands
- Support DHCP Client on WAN port
- Support DHCP server on the LAN port
- Support basic NAT and NAPT
- Support NTP
- Support VLAN (DATA VLAN and VOICE VLAN)
- WAN port supports main DNS and secondary DNS server function
- VPN (L2TP, PPTP) function
- QoS support Diffserv
- Support DNS relay, support NTP Client, supports simple firewall function
- Support network tools: including ping, trace route, telnet client

#### d) Maintenance and Management

- Support safe mode
- Can be updated via safe mode
- Support different user management
- Can through Web, keypad, Telnet to configure
- Can via HTTP, FTP, TFTP to update software and configuration files
- Support DHCP option 66 and customizing options
- Support Syslog (System Log)

![](_page_39_Picture_0.jpeg)

## 2. Using Place

- The telecom operators and (ITSP) network telephone service providers
- Large companies (for international and domestic long-distance call or internal communications, mainly in free call way)
- Small and medium enterprises who have import and export business, such as foreign travel agencies, studying abroad intermediary organization, immigration agencies intermediary organization and so on
- Foreign companies / joint ventures, China offices of foreign companies, representative offices and agents, etc.
- Foreign hotel (can be placed in the guest rooms and the business center or rent)
- Government in dealing with foreigners more departments at all levels, such as the foreign trade sector, the CPAFFC, sports units, literary unit, Bureau of Foreign Experts Affairs and foreign affairs departments, etc
- Schools and research institutes, such as the Sino-school, common school or the Foreign Affairs Department of research institutes, etc
- IP supermarkets, IP words (Setting more on the places where migrant workers, students and other low-income people get together usually)
- Personal and home users, such as immigrant families, host families, student hostels, personal long separated from the family for work, person maintain contact with family or friends who live abroad, etc.

#### Solution Symptom **POWER light** 1. Check if the power connection is correct. is not on Check if the power adapter is suited. 1. Check if the cable connection is valid, check if the PC network card indicator light is on. WAN/LAN 2. Check if the network card is working properly, the specific approach is to see if there is the device with "?" Or "!" under the Network Adapter" of the PC. If so, please delete the link light is not on device and reinstall. Otherwise, put the network card in another slot, if not yet, change the network card. Such as the common access modes(your PC have already installed dial-up software): 1. Make sure the front problem does not exist. Can not 2. Make sure the dial-up software is properly installed and set. access the Make sure you entered correct username and password. network 4. If you call successfully dial, but cannot access the network, please make sure if the IE browser's proxy server is set correctly.

### **3.** Common Problems

![](_page_40_Picture_0.jpeg)

5. Please try to log more websites to confirm if it is because of a Web server failure.