WSS Analog Voice Gateway Series

User Configuration Guide

Amendent Records

Document Rev.2.1 (Jan 12, 2011)

Chapter 1

The configuration of WSS8-2S/2O and WSS8-6S/2O is added.

Document Rev.2.0 (Sep 30, 2010)

Chapter 1

1.3.3 Description of WSS60

Chapter 2

Modifications were made to the terms accuracy of description, etc. in the document. GUI version: 1.9.81.300.9

The following interfaces were added: fax; batch configuration of subscriber/trunk lines; call history on FXS/FXO; SIP message count; TDM Capture; Ethereal Capture.

Document Rev. 1.2 (Aug 31, 2010)

2.4.2 IP Table: modification of Example 3

2.6.9 Functional keys, datasheet of VoIP Gateway

Document Rev. 1.1 (May 10, 2009)

Modifications were made to the terms, accuracy of description, etc. in the document.

Chapter 1

The equipment structure of WSS120 was added.

Chapter 2

The content was modified according to the release of new Web interface.

Document Rev. 01 (June 16, 2005)

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1 Overview

1.1 Product Introduction

WSS Series intelligent VoIP Gateways (hereinafter called “WSS Gateways” or simply “Gateways”) are designed for bridging the traditional telecom terminal devics into IP networks through SIP or MGCP protocols. The main applications include:

- For carriers and value-added service providers to provide telephone, fax and voice-band data services to subscribers using IP access methods such as FTTB, HFC, and ADSL;
- Used to bridge the traditional telecom terminal equipments, such as PBXs, to the IP core networks of carriers;
- Connected with PBX of enterprises to provide IP-based voice private network solutions for institutions, enterprises and schools;
- Used as remote acces equipments for IP-PBXs in call center deployment.

WSS Gateways are suitable for placement on office desktops or installation on walls in the corridor and racks in the equipment room.

WSS Series includes WSS8, WSS60 and WSS120 subseries. Their features are similar with the main differences as follows:

Table 1-1 Differences Between WSS Gateway Series

	Capacity	Chassis	Subscriber Line Board Card	Installation	Power
WSS8	2-8 FXS/FXO Ports	Plastic Casing	Built-in	Desktop	5-9 VDC
WSS60	16-48 FXS/FXO Ports	19’’ wide and 1U High	Built-in	Rack	100-240 VAC
WSS120	24-48 FXS/FXO Ports	19” Wide and 1U High	Pluggable	Rack	100-240 VAC, -48 VDC (Optional)
WSS120	48-96 FXS/FXO Ports	19” Wide and 2U High	Pluggable	Rack	100-240 VAC, -48 VDC (Optional)

WSS Gateways use Freescale® PowerQUICC communications processors as main control processors (including 50MHz MPC852T, 200MHz MPC8250 and 300MHz MPC8247) and TI’s TMS320VC5509A high-performance digital signal processing chips as processors for voice and fax processing (equipped with 1-12 DSP chips based on the need of concurrent call capacity), and are integrated with 32MB-64MB SDRAM as system memory, 4MB-16MB FLASH as permanent file system. The powerful processing capability and sufficient hardware configuration ensure that all products of WSS Series can provide concurrent calls of full capacity and maintain good call quality.

All WSS Gateways run on stable and reliable embedded Linux operating system. On top of Linux OS, the driver layer handles hardware specific control in different product platforms. This makes single source application software running cross the full range of WSS product series, and ensures the consisten functions and stable performance in different WSS product lines.

WSS Gateways support SIP and MCGP protocols. They can provide

- PBX functions such as hunting group, second stage dialing, internal communications, caller ID (FSK/DTMF), call transfer, call waiting, call hold, call barring, caller ID restriction, hotline, corporate CRBT, three-way calling, ring group, fax and etc;
- FXO related functions such as PSTN failover, gain control, busy tone detection, voice prompt in inbound calls, polarity reversal detection;
- Media stream processing functions such as RTP redundancy, packet loss compensation, G.711/G.729A/G.723.1/iLBC/GSM voice codec, echo cancellation, and etc.

WSS Gateways support local and remote, distributed and centralized management modes, including Web access management，command line configuration based on Linux OS, auto-provision for firmware upgrade and configuration management based on TFTP/FTP/HTTP, SNMPv2, TR069 based ACS.

1.2 Functions and Features

- Connect analog telephone, PBX, facsimile machine and POS machine to the IP core network, or PSTN;
- Work with service platform to provide various telephone supplementary services;
- Support protocols: SIP, MGCP;
- Flexible configuration of FXS/FXO interfaces;
- Support static IP address configuration or dynamically obtain an IP address through DHCP and PPPoE;
- Support G.711, G.729A, G.723.1, GSM, iLBC;
- Support echo cancellation;
- Up to 500 routing rules can be stored in gateways;
- Support digitmap;
- Support T.30/T.38 fax mode;
- Support multiple local and remote maintenance & management modes such as Web, Telnet, auto-provision, and TR069/TR104/TR106 clinet;.
- security strategy: IP filter, encryption
- Support call progress tones for various countries and regions;
- Support FXO second stage dialing or voice prompt;
- Support PSTN failover through FXO ports;
- Support High Capacity SD Card (optional, only for WSS60)
- Support polarity inverse detection and busy tone detection
- Support three-way calling
- Compatible with unified communication solutions, such as CallManager, OCS and Asterisk

1.3 Equipment Structure

1.3.1 WSS8

WSS8 is the product with smallest capacity in WSS Gateway Series. Designed with small plastic structure for desktop placement, WSS8 can provide up to 8 analog line interfaces. WSS8 supports the following types of configuration:

Table 1-2 Common Configuration Combination of WSS8

Models	Number of FXS Ports	Number of FXO Ports
WSS8-2S/2	2	2
WSS8-6S/2	6	2
WSS8-4S	4	0
WSS8-8S	8	0
WSS8-4FXO	0	4
WSS8-8FXO	0	8
WSS8-4S/4	4	4

Figure 1-1 WSS8 Front Panel

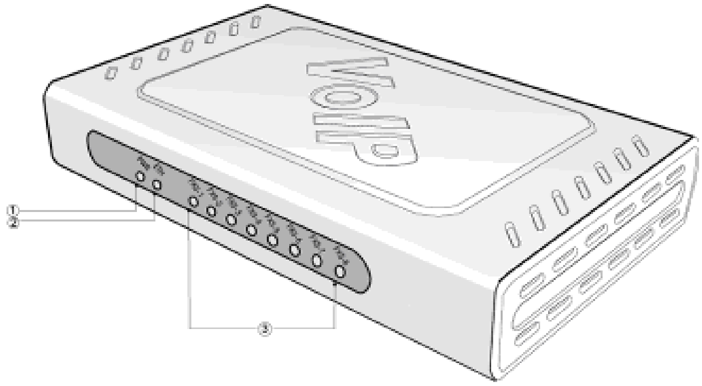


Table 1-3 Description of WSS8 Front Panel

#	Description
①	Power indicator (PWR), Light-on indicates that it has been powered.
②	Steady on indicates valid Ethernet link, flashing indicates Ethernet activities (receiving and/or transmitting)
③	Analog subscriber line (FXS) or analog trunk (FXO) interface indicator, Light-on indicates that it is in use.

Figure 1-2 WSS8 Back Panel

Table 1-4 Description of WSS8 Back Panel

#	Description
①	Power interface, 5-9 VDC input
②	10/100 M Ethernet Interface, RJ45
③	Analog subscriber line (FXS) or analog trunk (FXO) interface

Table 1-5 Configuration Description of Analog Line Interfaces for All WSS8 Models

WSS8 Models	RJ11 Interface Configuration							
	1	2	3	4	5	6	7	8
WSS8-2S/2	Trunk Line 1	Trunk Line 2	Subscriber Line 1	Subscriber Line 2	NA	NA	NA	NA
WSS8-6S/2	Trunk Line 1	Trunk Line 2	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Subscriber Line 5	Subscriber Line 6
WSS8-4S	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	NA	NA	NA	NA
WSS8-8S	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Subscriber Line 5	Subscriber Line 6	Subscriber Line 7	Subscriber Line 8
WSS8-4FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	NA	NA	NA	NA
WSS8-8FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	Trunk Line 5	Trunk Line 6	Trunk Line 7	Trunk Line 8
WSS8-4S/4	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4

1.3.2 WSS60

Designed with a 1U high and 19’’ wide compact chassis, WSS60 is suitable for installation in a standard cabinet. The interface card of WSS60 uses a RJ-45 socket and is connected to the distribution panel in equipment room using CAT-5 cables supplied with the unit. It has a built-in 110-220V power module. WSS60 offers up to 48 interfaces of FXS/FXO. WSS60 supports the following types of configuration.

Table 1-6 Configuration combination of WSS60

Models	Numbers of FXS Ports	Numbers of FXO Ports
WSS60-16S	16	0
WSS60-24S	24	0
WSS60-32S	32	0
WSS60-48S	48	0
WSS60-8S/8	8	8
WSS60-24S/8	24	8
WSS60-40S/8	40	8
WSS60-16S/16	16	16
WSS60-32S/16	32	16
WSS60-24S/24	24	24

Figure 1-3 WSS60 Front Panel

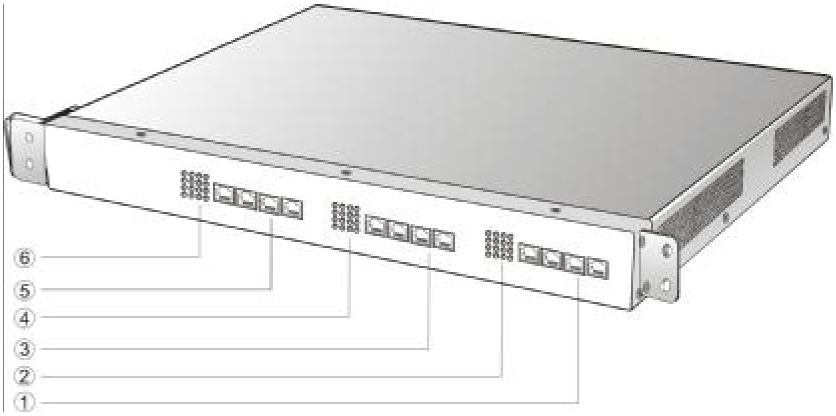


Table 1-7 Description of WSS60 Front Panel

#	Description
①③⑤	Three interface slots; each can correspond with four RJ45 sockets; each RJ45socket can correspond with four pairs of analog lines. Note: numbers of interface slots vary from different configuration.
②④⑥	Matrix of 4 x 4 LED status indicator on interface cardl

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation. Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-8 Pin Specifications for WSS60 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-4 Schematic Diagram of WSS60 Subscriber Line Connection

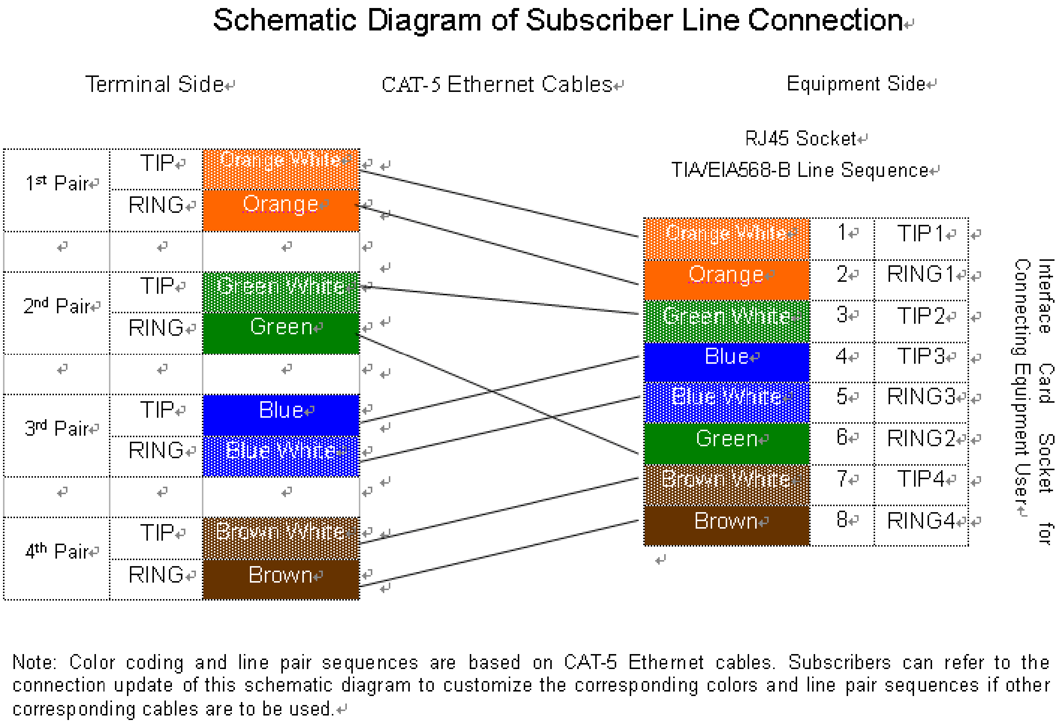


Figure 1-5 WSS60 Back Panel

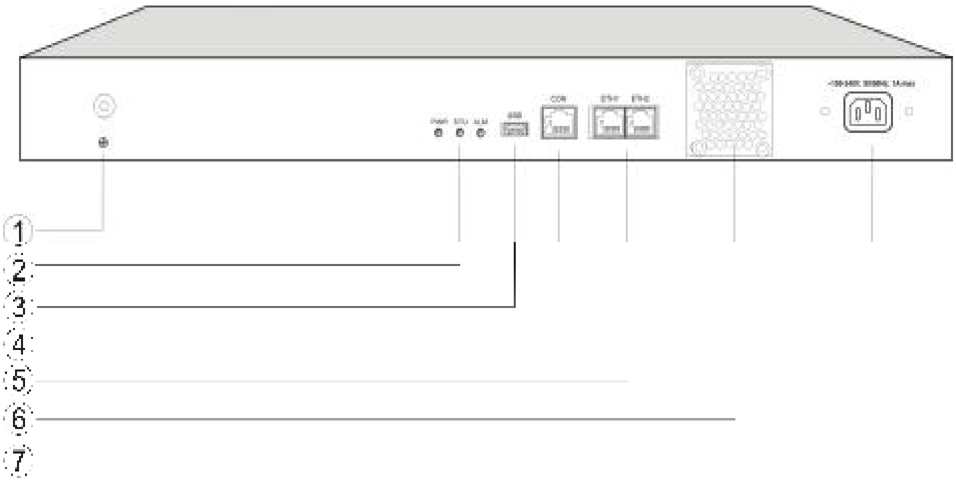


Table 1-9 Description of WSS60 Back Panel

#	Description
①	Ground Pole
②	Indicator, see Table 1-17 for description
③	USB Interface, reserved for future use
④	Configuration interface (CON), Ethnet lines used for local management and debugging

#	Description
⑤	Two Ethernet interfaces: one IP address
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input

Table 1-10 Meanings of WSS60 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Off	No alarms
		Red Flash	New alarms occurred but not confirmed.
		Red Constant	System in the process of powerup and not in the normal operation mode
		Red	Alarms existed and all alarm information confirmed.

1.3.3 WSS120 1U

Designed with 1U high and 19’’ wide compact chassis and a swappable modular structure, WSS120 can offer up to 48 analog lines. The interface card of WSS120 uses a RJ45 socket and is connected to the distribution panel in equipment room using CAT-5 cables supplied with the unit.

The device of WSS120 1U can hold two interface cards which enable to flexibly configure FXS and FXO ports. And each card equips up to 24 ports. It supports the following configurations:

Table 1-11 Configuration Combination of WSS120:

Models	Number of FXS Ports	Number of FXO Ports
WSS120-24FXO	0	24
WSS120-32FXO	0	32
WSS120-48FXO	0	48
WSS120-40S/8	40	8
WSS120-36S/12	36	12
WSS120-32S/16	32	16
WSS120-28S/20	28	20
WSS120-24S/24	24	24

Figure 1-6 WSS120 Front Panel

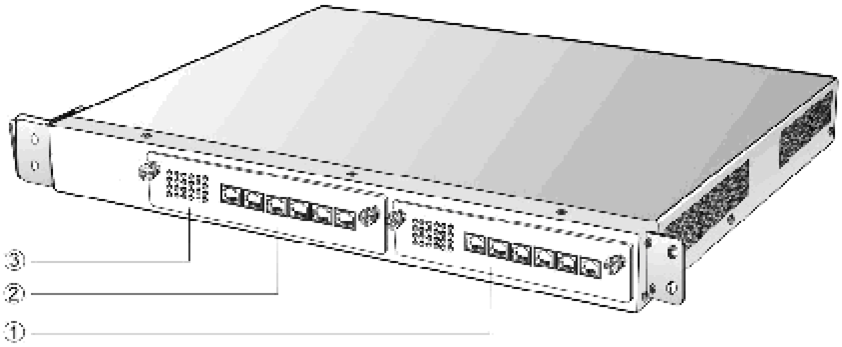


Table 1-12 Description of WSS120 Front Panel

#	Description
① and ②	Two interface slots; each can contain one 24-port interface card.
③	Matrix of 6×4 LED status indicator on interface card

WARNING

Do not plug and remove the interface cards of WSS120 when equipment is powered on.

Numbering definition of system interface slots: On the left side of main chassis is #1 slot (marked with No.1 to 24), on the right side of main chassis is #2 slot (marked with No.25 to 48).

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation. Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-13 Pin Specifications for WSS120 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-7 Schematic Diagram of WSS120 Subscriber Line Connection

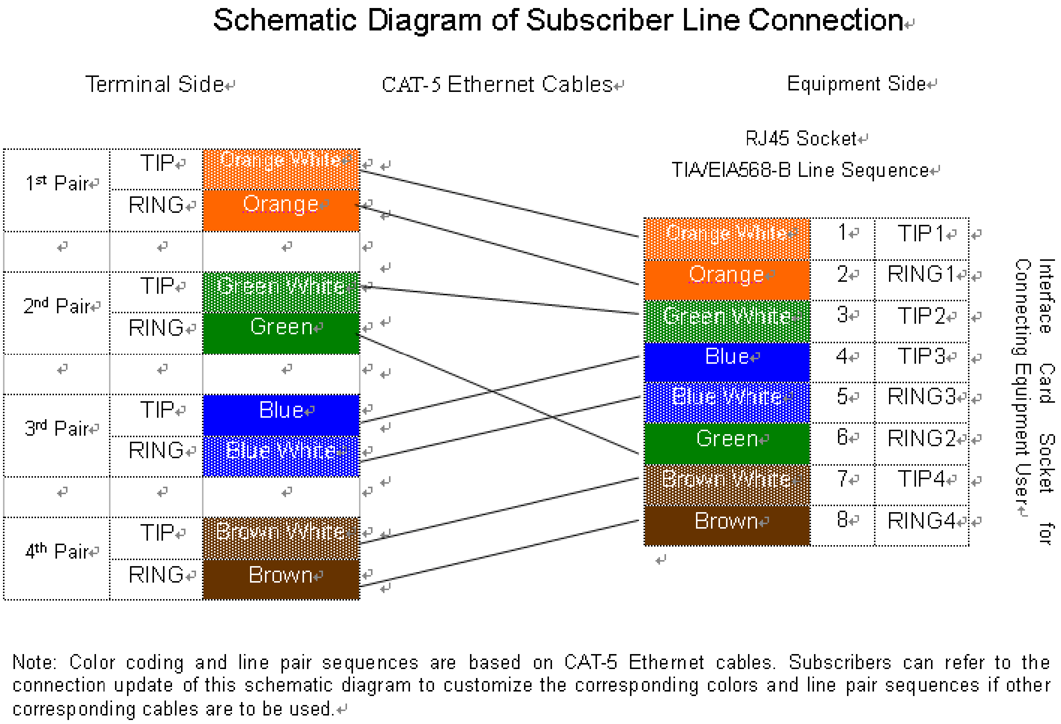


Table 1-14 Corresponding Relation Between WSS120 RJ45 Socket and Line Number

RJ45 Socket No. (From Left to Right)	1	2	3	4	5	6
Line No. of This Card	1 ~ 4	5 ~ 8	9 ~ 12	13 ~ 16	17 ~ 20	21 ~ 24

There is a 6 × 4 LED indicator matrixes on the left side of interface board. Each row of LED indicator matrixes matches four telephone lines on a RJ45. The first row on the left matches Line 1-4 respectively from top to bottom, the first row on the right matches Line 21-24 respectively from top to bottom, and the middle rows in the same manner.

LED indicators are used for multiple purposes as follows

- Line status indication: This is the most common mode during normal use of equipment. In this mode, if a line is idle, the indicator corresponding to it goes off; if a line is in call or in use status (such as ringing, offhook and caller ID transmission of FXS interface, ringing, offhook and caller ID detection of FXO interface) the indicator corresponding to it goes on.
- Line type indication: This is the mode for cable wiring check when installing the equipment. This mode can be entered by disconnecting Ethernet cables (Both WAN and LAN ports must be disconnected) at installation stage. After entering this mode, steady on LED indicates that the corresponding line is equipped as analog subscriber line type, flashing LED indicates that the corresponding line is equipped as analog foreign exchange line type, off LED indicates that the corresponding line is not equipped or not ready for use.
- System operation status indication: This is the mode for displaying information on system operation of equipment in specific conditions. Usually, this mode is entered when some prompts are required to give operator during equipment startup, diagnosis or operation. In this mode, LED flashes to display numbers, letters or other patterns in matrix. Please refer to the Appendix: Check List for Operation Status Indication of WSS120 System.

Figure 1-8 WSS120 Back Panel

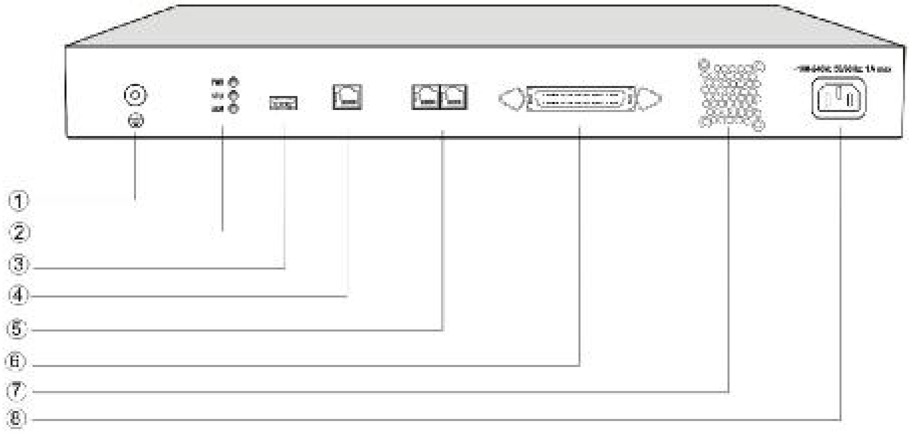


Table 1-15 WSS120 Back Panel

#	Description
①	Ground Pole
②	Indicator, see Table 1-16 for description.
③	USB interface, reserved for future use.
④	Configuration interface (CON), used for local management and debugging.
⑤	Two Ethernet interfaces: ETH1 and ETH2, only ETH1 has been set when the equipment is delivered from factory, default IP address: 192.168.2.240
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input.

Table 1-16 Meanings of WSS120 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Green	No alarms
		Red Flash	New alarms occurred but not confirmed
		Red	Alarms existed and all alarm information confirmed

1.3.4 WSS120 2U

The device of WSS120 2U can hold four interface cards which enable to flexibly configure FXS and FXO ports. And each card equips up to 24 prots. WSS120 2U can provide up to 96 ports. It supports the following configurations:

Table 1-17 Configuration Combination of WSS120 2U:

Models	Number of FXS Ports	Number of FXO Ports
WSS120-72S	72	0
WSS120-96S	96	0
WSS120-72FXO	0	72
WSS120-96FXO	0	96
WSS120-64S/8	64	8
WSS120-88S/8	88	8
WSS120-60S/12	60	12
WSS120-84S/12	84	12
WSS120-56S/16	56	16
WSS120-80S/16	80	16
WSS120-52S/20	52	20
WSS120-76S/20	76	20
WSS120-48S/24	48	24
WSS120-72S/24	72	24
WSS120-44S/28	44	28
WSS120-68S/28	68	28
WSS120-40S/32	40	32
WSS120-64S/32	64	32
WSS120-36S/36	36	36
WSS120-60S/36	60	36

Figure 1-9 WSS120 2U Front Panel

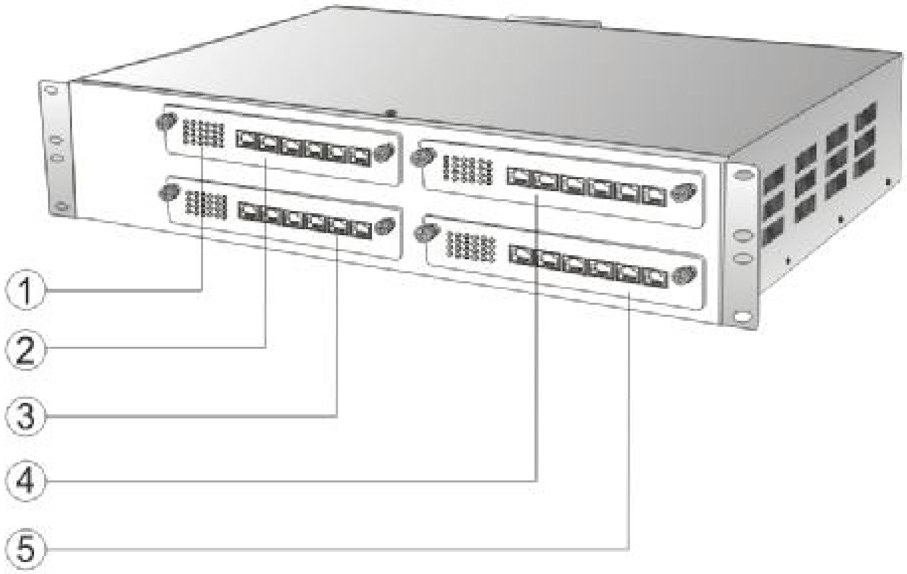


Table 1-18 Description of WSS120 Front Panel

#	Description
①	Matrix of 6×4 LED status indicator on interface card
②③④⑤	Four interface slots; each can contain one 24-port interface card.

WARNING

Do not plug and remove the interface cards of WSS120 when equipment is powered on.

Numbering definition of system interface slots: On the low-left side of chassis is #1 slot (marked with No.1 to 24), on the low-right side of chassis is #2 slot (marked with No.25 to 48), on the up-left side of chassis is #3 slot (marked with No.49 to 72), and on the up-right side of chassis is #4 slot (marked with No.73 to 96).

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation. Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-19 Pin Specifications for WSS120 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-10 Schematic Diagram of WSS120 Subscriber Line Connection

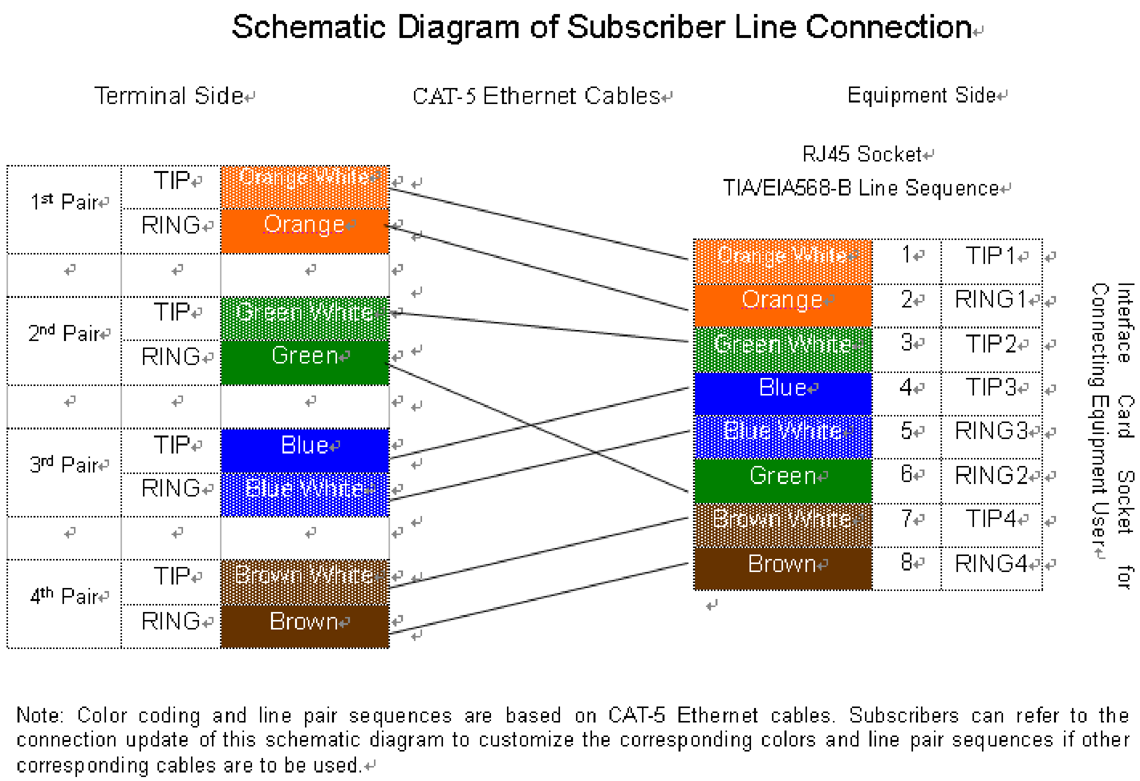


Table 1-20 Corresponding Relation Between WSS120 RJ45 Socket and Line Number

RJ45 Socket No. (From Left to Right)	1	2	3	4	5	6
Line No. of This Card	1 ~ 4	5 ~ 8	9 ~ 12	13 ~ 16	17 ~ 20	21 ~ 24

There is a 6 × 4 LED indicator matrixes on the left side of interface board. Each row of LED indicator matrixes matches four telephone lines on a RJ45. The first row on the left matches Line 1-4 respectively from top to bottom, the first row on the right matches Line 21-24 respectively from top to bottom, and the middle rows in the same manner.

LED indicators are used for multiple purposes as follows

- Line status indication: This is the most common mode during normal use of equipment. In this mode, if a line is idle, the indicator corresponding to it goes off; if a line is in call or in use status (such as ringing, offhook and caller ID transmission of FXS interface, ringing, offhook and caller ID detection of FXO interface) the indicator corresponding to it goes on.
- Line type indication: This is the mode for cable wiring check when installing the equipment. This mode can be entered by disconnecting Ethernet cables (Both WAN and LAN ports must be disconnected) at installation stage. After entering this mode, steady on LED indicates that the corresponding line is equipped as analog subscriber line type, flashing LED indicates that the corresponding line is equipped as analog foreign exchange line type, off LED indicates that the corresponding line is not equpped or not ready for use.
- System operation status indication: This is the mode for displaying information on system operation of equipment in specific conditions. Usually, this mode is entered when some prompts are required to give operator during equipment startup, diagnosis or operation. In this mode, LED flashes to display numbers, letters or other patterns in matrix. Please refer to the Appendix: Check List for Operation Status Indication of WSS120 System.

Figure 1-11 WSS120 2U Back Panel

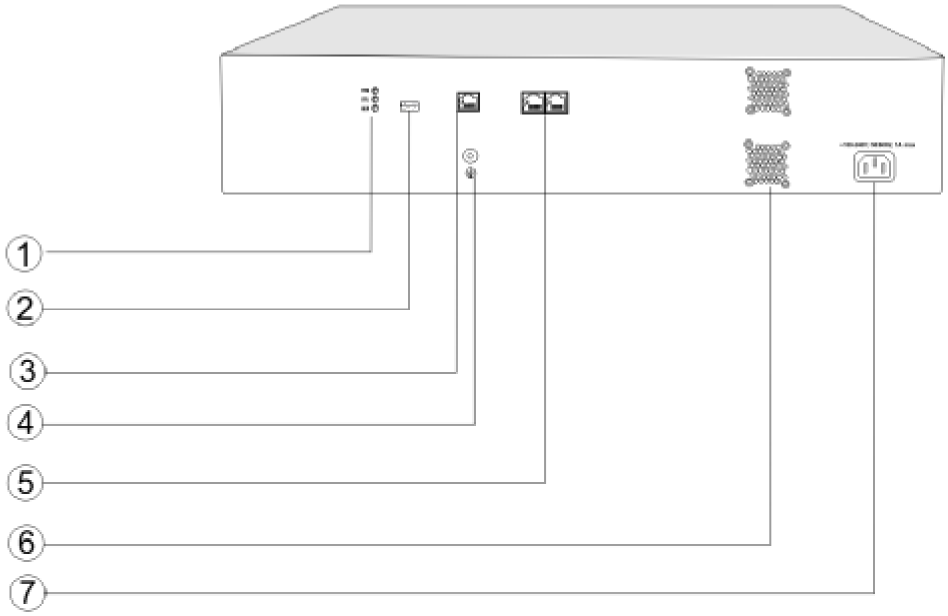


Table 1-21 WSS120 Back Panel

#	Description
①	Indicator, see Table 1-28 for description.
②	USB interface, reserved for future use.
③	Configuration interface (CON), used for local management and debugging.
④	Ground Pole
⑤	Two Ethernet interfaces: ETH1 and ETH2, only ETH1 has been set when the equipment is delivered from factory, default IP address: 192.168.2.240
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input.

Table 1-22 Meanings of WSS120 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Green	No alarms
		Red Flash	New alarms occurred but not confirmed
		Red	Alarms existed and all alarm information confirmed

2 Parameter Setting

2.1 Login

2.1.1 Obtain Gateway IP Address

WSS8 Gateways start DHCP service by default, and automatically obtain an IP address on the LAN; users can use the factory default gateway IP address if it is unable to be obtained (e.g. when connected directly with a computer).

WSS60 and WSS120 Gateways use a static IP address by default.

Table 2-1 Default IP Address of Gateway

Type	Default DHCP Service	Default IP Address	Default Subnet Mask
WSS8	Enabled	192.168.2.218	255.255.0.0
WSS60	Disabled	192.168.2.240	255.255.0.0
WSS120	Diasabled	192.168.2.240	255.255.0.0

- DHCP Used in Network

  Users can dial "# #"to obtain the current gateway IP address and version information of firmware using the telephone connected to the subscriber line (FXS interface) after the equipment is powered on.

  If the gateways are only configured with FXO ports for analog trunks without FXS ports for subscriber lines (e.g. WSS8-4FXO), users can dial into the gateway by connecting a PBX extension line or PSTN POTS line to a FXO port, and press "# #" to obtain the current gateway IP address and version information of firmware after receiving the second dial tone.

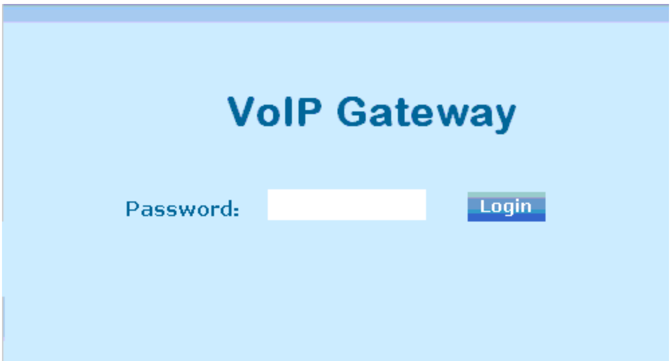
- Fixed IP Address Used
  - If the DHCP service on the network is not available or the gateway is directly connected with a computer, the gateways will use the factory default IP address.
  - A user could fail to log in with the default IP address if the IP address of user's computer and the default gateway IP address are not at the same network segment. It is recommended that the IP address of user's computer is changed to be identical with the same network segment of gateway. For example, if the gateway IP address is 192.168.2.240, it is recommended to set the computer's IP address to any address at the network segment of 192.168.2.XXX).
- PPPoE Used

  In "Basic Configuration> Network Configuration", the gateways will automatically obtain the WAN address returned by access network after PPPoE service is started and user name and password are set. Users can dial "# #" on the gateways to receive the IP address and version information of firmware the gateways has obtained.

2.1.2 Log on Gateway

Double-click the icon to open IE browser, and enter the gateway IP address in the browser address bar (eg. 192.168.2.218); you can enter the login interface for gateway configuration by entering a password on the login interface.

Figure 2-1 Login Interface for WSS Gateway Configuration



Both Chinse and English version of WEB are offered.

2.1.3 Permission of Gateway Administrator

Logon users are classified into "administrator" and "operator". The default password is seen Table 2-2. The password is shown in a cipher for safety.

Table 2-2 Default Passwords of Gateway

Type	Default Administrator Passwords (lowercase letters required)	Default Operator Password
WSS8	voip	operator
WSS60	voip	operator
WSS120	voip	operator

- The administrator can browse and modify all configuration parameters, and modify login passwords.
- The operator can browse and modify part of configuration parameters.

The gateways allow multiple users to log in:

- The administrator has permission for modification and the operator has permission for browsing;
- When multiple users with same level of permission log in, the first has permission for modification, while the others only have permission for browsing.

CAUTION

The system will confirm timeout if users do not conduct any operation within 10 minutes after login. They are required to log in again for continuing operations.

Upon completion of configuration, click "Logout" button to return to the login page, so as not to affect the login permission of other users.

2.2 Buttons Used on Gateway Management Interface

"Submit" and "Restore Default Configuration" buttons are at the bottom of configuration interface.

- Submit" Button: Submit configuration information. Users click "Submit" button after completion of parameter configuration on a page. A success prompt will appear if configuration information is accepted by the system; if a "The configuration takes effect after the system is restarted"dialog box appears, it means that the parameters are valid only after system restart; it is recommended that users press the "Restart" button on the "Tool" page to validate the configuration after changing all parameters to be modified.

- "Default" Button: Click this button to use default configuration of gateway. A success prompt will appear on the interface after the system restores parameters on the configuration page to default configuration. For part of parameters, it is required to restart the software to validate the default configuration, and in this case "The configuration takes effect after the system is restarted" will appear on the interface. Subscribers can click "Restart" on the "Tool" page to restart.

2.3 **Basic Configuration**

2.3.1 Network Configuration

After login, click "Basic > Network" tab to open the configuration interface.

Figure 2-2 Network Configuration Interface

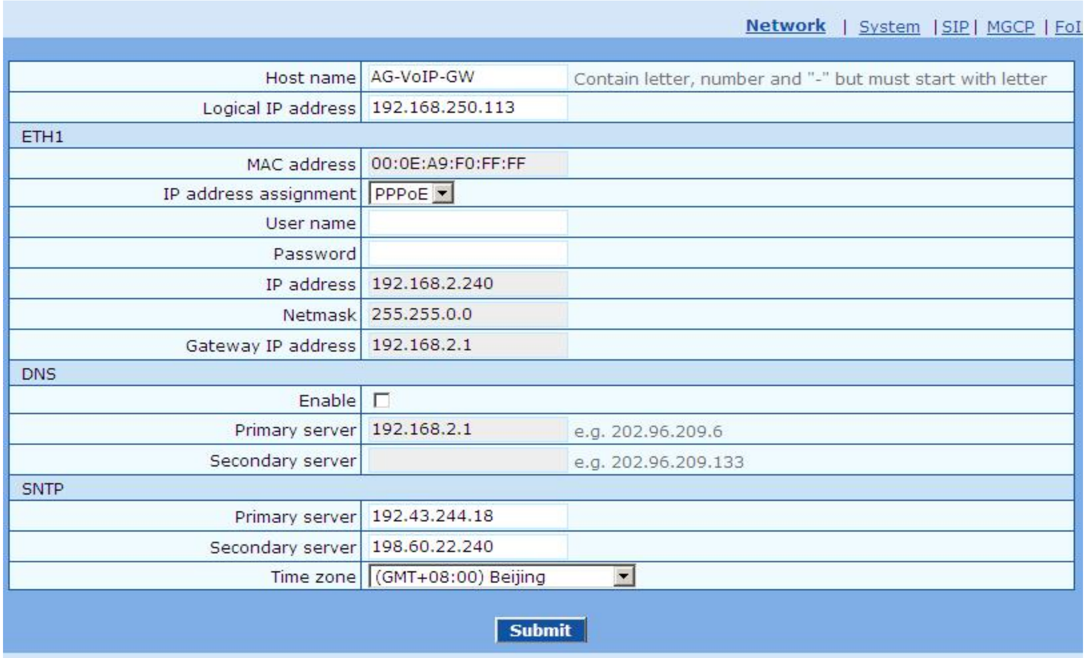


Table 2-3 Network Configuration Parameters

Name	Description
Host name	This is the equipment name of a configuration gateway. The default values of WSS8, WSS60 and WSS120 are WSS8-VoIP-AG, WSS60-VoIP-AG and WSS120-VoIP-AG respectively. Users can set a different name for each gateway to distinguish from each other according to the deployment plan. A host name can be a maximum of 48 characters, either letters (A-Z or a-z), numbers (0-9) and minus sign (-). It may not be null or space, and it must start with a letter.
Logical IP address	This parameter only exists in WSS100-TG, used to display the actual gateway IP address in use.
ETHn	
MAC address	Display the MAC address of gateway.

Name	Description
IP address assignment	Methods for obtaining an IP address l static: Static IP address is used; l DHCP: Activate DHCP service and use the dynamic host configuration protocol (DHCP) to allocate IP addresses and other network parameters; l PPPoE: PPPoE service is used.
User name	Enter an authentication user name if PPPoE service is selected, and there is no default value.
Password	Enter an authentication password if PPPoE service is selected, and there is no default value.
IP address	If "Static" or "DHCP" is selected for the network type but an address fails to be obtained, the gateways will use the IP address filled in here. If the gateways obtain an IP address through DHCP, the system will display the current IP address automatically obtained from DHCP by the gateways. This parameter must be set due to no default value.
Netmask	The subnet mask is used with an IP address. When the gateways use a static IP address, this parameter must be entered; when an IP address is automatically obtained through DHCP, the system will display the subnet mask automatically obtained by DHCP. This parameter must be set due to no default value.
Gateway IP address	LAN gateway IP address where the gateways are located. When the gateways obtain an IP address through DHCP, the system will display the LAN gateway address automatically obtained through DHCP. This parameter must be set due to no default value.
ETH1	Only apply to WSS100-TG
MAC address	Display the MAC address of gateway
IP address	Fill in IP address of ETH1
Netmask	The subnet mask is used with an ETH1 IP address.
DNS	
Enable	Activate DNS service.
Primary Server	If DNS service is activated, the network IP address of preferred DNS server must be entered, and there is no default value.
Secondary Server	If DNS service is activated, the network IP address of standby DNS server can be entered here. It is optional and there is no default value.
SNTP	
Primary Server	Enter the IP address of preferred time server here. This parameter must be set due to no default value.
Secondary Server	Enter the IP address of standby time server here. This parameter must be set due to no default value.

Name	Description
Time Zone	Select a time zone, and the parameter values include: l (GMT-11:00) Midway Island l (GMT-10:00) Honolulu. Hawaii l (GMT-09:00) Anchorage, Alaska l (GMT-08:00) Tijuana l (GMT-06:00) Denver l (GMT-06:00) Mexico City l (GMT-05:00) Indianapolis l (GMT-04:00) Glace_Bay l (GMT-04:00) South Georgia l (GMT-03:30) Newfoundland l (GMT-03:00) Buenos Aires l (GMT-02:00) Cape_Verde l (GMT) London l (GMT+01:00) Amsterdam l (GMT+02:00) Cairo l (GMT+03:00) Moscow l (GMT+03:30) Teheran l (GMT+04:00) Muscat l (GMT+04:30) Kabul l (GMT+05:30) Calcutta l (GMT+05:00) Karachi l (GMT+06:00) Almaty l (GMT+07:00) Bangkok l (GMT+08:00) Beijing l (GMT+09:00) Tokyo l (GMT+10:00) Canberra l (GMT+10:00) Adelaide l (GMT+11:00) Magadan l (GMT+12:00) Auckland

2.3.2 System Configuration

After login, click “Basic > System” tab to open the configuration interface.

Figure 2-3 System Configuration Interface

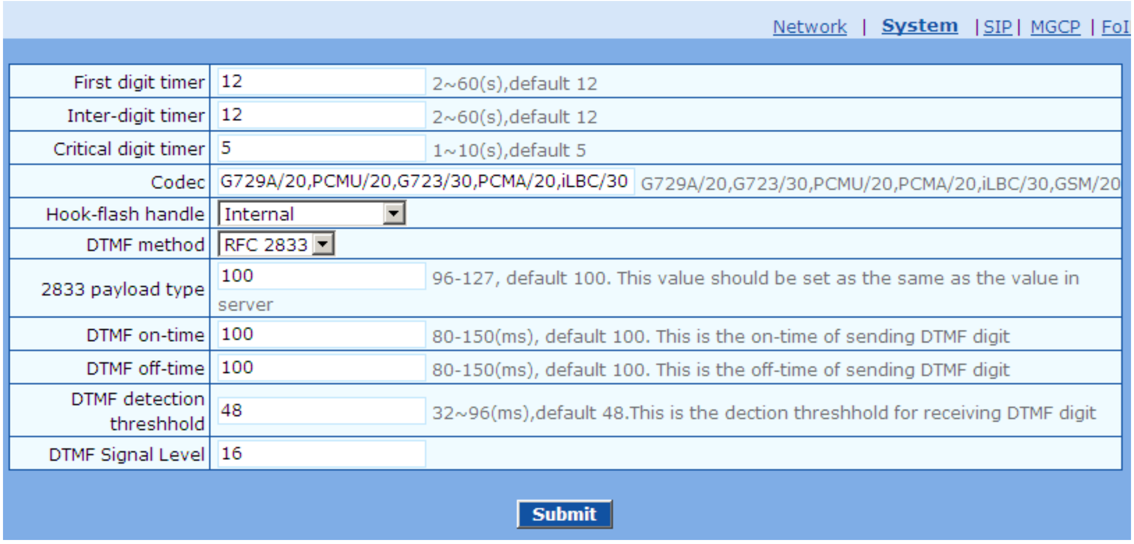


Table 2-4 System Configuration Parameters

Name	Description
First digit timer	If a subscriber hasn’t dialed any number within a specified time by this parameter after offhook, the gateways will consider that the subscriber has given up the call and prompt to hang up in busy tone. Unit: second; Default value: 12 seconds.
Inter-digit timer	If a subscriber hasn’t dialed the next number key from the time of dialing the last number key to the set time by this parameter, the gateways will consider that the subscriber has ended dial-up and call out the dialed number. Unit: second; Default value: 12 seconds.
Critical digit timer	This parameter is used with the "x.T" rule set in dialing rules. For example, there is "021.T" in the dialing rules table. When a subscriber has dialed 021 and hasn’t dialed the next number within a set time by this parameter (eg. 5 seconds), the gateways will consider that the subscriber has ended dial-up and call out the dialed number 021. Input intergers, not decimal fractions Unit: second; Default value: 5 seconds.
Codec	Codecs methods supported by the gateways include G729A/20, G723/30, PCMU/20, PCMA/20, iLBC/30 and GSM/20 (as shown in table 2-5). This parameter must be set due to no default value. Several encoding methods can be configured in this item at the same time, separated with “,” in the middle; the gateways will negotiate with the platform in the order from front to back when configuring the codec methods
Hook-flash handle	The gateways provide the following processing modes after detecting hook flash from subscriber terminals: processing the hook flash internally; transmitting the hook flash to platform with RFC 2833, and transmitting the flash-off to platform with SIP INFO.
DTMF method	Transmission modes of DTMF signal supported by the gateways include Audio, RFC 2833 and SIP INFO. The default value is Audio. Audio: DTMF signal is transmitted to the platform with sessions; SIP INFO: Separate DTMF signal from sessions and transmit it to the platform in the form of SIP INFO messages; RFC 2833: Separate DTMF signal from sessions and transmit it to the platform through RTP data package in the format of RFC2833.

Name	Description
2833 payload type	Used with "RFC 2833" in the DTMF transmission modes. The default value of 2833 payload type is 100. The effective range available: 96 ~ 127. This parameter should match the setting of far-end device (eg. platform).
DTMF on-time	This parameter sets the on time (in ms) of DTMF signal sent from FXO port. The default value is 100 ms. Generally, the duration time should be set in the range of 80 ~ 150 ms.
DTMF off-time	This parameter sets the off time (ms) of DTMF signal sent from FXO port. The default value is 100 ms. Generally, the interval time should be set in the range of 80 ~ 150 ms.
DTMF detection threshhold	Minimum duration time of effective DTMF signal. Its effective range is 32-96 ms. The greater the value is set, the more stringent the detection is.

Table 2-5 Codec Methods Supported by Gateways

Codec Supported by WSS	Bit Rate (Kbit/s)	Time Intervals of RTP Package Sending (ms)
iLBC	13.3/15.2	20/30
GSM	13	20
G729A	8	10/20/30/40
G723	5.3/6.3	30/60
PCMU/PCMA	64	10/20/30/40

2.3.3 SIP Configuration

After login, click "Basic> SIP" tab to open the SIP configuration interface.

Figure 2-4 SIP Configuration Interace

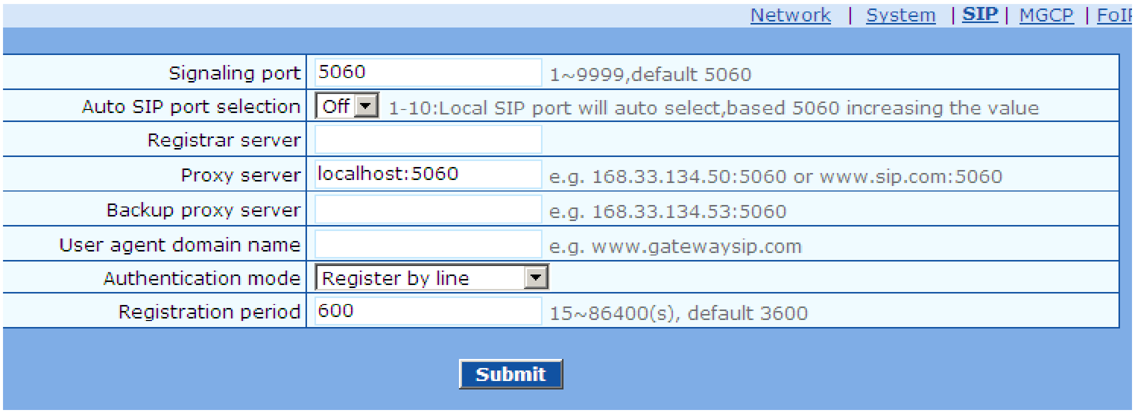


Table 2-6 SIP Configuration Parameters

Name	Description
Signaling port	Configure the UDP port for transmitting and receiving SIP messages, with its default value 5060. Note: The signaling port number can be set in the range of 1-9999, but cannot conflict with the other port numbers used by the equipment.

Name	Description
Change signaling port	If “n”(ranked from 1-10) is chosen, after the failure registration of signaling port’s original configuration, the range of signaling port’s change varies from “original signaling port, original signaling port +n”. Register with the new signaling port value (signaling port +1) until it succeds.
Register server	Configure the address and port number of SIP register server, and the address and port number are separated by “:”. It has no default value. The register server address can be an IP address or a domain name. When a domain name is used, it is required to activate DNS service and configure DNS server parameters on the page of configuring network parameters. For example: “201.30.170.38:5060”, “register.com: 5060”.
Proxy server	Configure the IP address and port number of SIP proxy server, and the address and port number are separated by “:”. It has no default value. The proxy server address can be set to an IP address or a domain name. When a domain name is used, it is required to activate DNS service and configure DNS server parameters on the page of configuring network parameters. Examples of complete and effective configuration: "201.30.170.38:5060", "softswitch.com: 5060".
Backup proxy server	By specifying the corresponding IP addresses, the gateway can be configured to have multiple soft switches as backup proxy servers. Make sure that the IP addresses must be in their full format. Eg. “202.202.2.202:2727”. The proxy and register severs must be identical. Conditions for falling over to the backup proxy server (any): 1）Gateway register is timeout; 2）No response to master server calls is timeout;
User agent domain name	This domain name will be used in INVITE messages. If it is not set here, the gateways will use the IP address or domain name of proxy server as user agent domain name. It has no default value. It is recommended that subscribers not use LAN IP address to set domain name parameter.
Authentication mode	The gateway supports three registration shemes: register per line, register per gateway and Line Reg/GW Auth. The default value is register by line. l Register by line: authentication and register per line; l Register by gateway: authentication and register per gateway; l Line Reg/GW Auth: register per line, but authentication per gateway.
User name	Configure the user name as part of the account for registration, and it has no default value. Note: If “Register by gateway” or “Line Reg/GW Auth”, is selected, the user name must be entered here. If “register by line” is selected the user name should be set on “Line > Feature” page (Refer to “Feature”).
Password	Password as part of account information is used for authentication by platform. It has no default value. It is formed with either numbers or characters, and case sensitive. Note: If “Register by gateway” or “Line Reg/GW Auth”, is selected, the password must be entered here. If “register by line” is selected the password should be set on “Line > Feature” page (Refer to “Feature”).
Registration period	Valid time of SIP re-registration in second.

2.3.4 MGCP Configuration

The gateways use SIP protocol by default. When the gateways need to interface with MGCP protocol -based softswitch platform, users should set relevant parameters here.

After login, click “Basic > MGCP” tab to open the configuration interface.

Figure 2-5 MGCP Configuration Interface

Table 2-7 MGCP Configuration Parameters

Name	Description
Signaling port	Configure the UDP port for transmitting and receiving MGCP messages, and default value is 2427. Note: The signaling port number can be set in the range of 1-9999, but cannot conflict with the other port numbers used by the equipment.
Proxy server	Configure the IP address and port number of MGCP proxy server, separated by “:”, and it has no default value. The address can be set to an IP address or a domain name according to the subscribers’ requirements. When a domain name is used, it is required to activate DNS service and configure DNS server on the page of configuring network parameters. Examples of complete and effective configuration: “202.202.2.202:2727”, “callagent.com: 2727”.
User agent domain name	It is the gateway logo when the gateways register with proxy server, and it has no default value. Example: test.RealTone.com, [192.168.2.100]. Note: if IP address is written in this way, like “[192.168.2.100]”, “[]” should be added.
Default event package	List all the types of default event packages supported by gateways, and multiple package names are separated by“,”. The default value is L, D, G • L: Line Package; • D: DTMF Package; • G: Generic Media Package.

Name	Description
Persistent line event	List the event types that the gateway can report, and persistent line events are separated by “,”. When gateways process the events listed here, they will report to the call agent. Note: This parameter must be set since there is no default value. The factory setting is L/HD, L/HU: l L/HD: Offhook; l L/HU: Onhook.
FXO event package	Handset Package Line Pakage
Wildcard	Select whether a wildcard with prefix is allowed when a gateway registers to the proxy server. The default value is “not allowed”. l Partially allowed: Gateways will use a wildcard with fixed prefix (eg. aaln / *) when registering. For example, when configuring telephone numbers, if line 1 is set to “aaln/1”, line 2 is set to “aaln/2” and line 3 is set to “aaln/3”, the gateways will register to the call agent in “aaln/*” without the need of registering the lines individually. l Allowed: the gateways will use a wildcard in registering without prefix.
Compatibility Configuration	
CR for End-of-Line	Select whether CR is used as the end of line in the MGCP messages. Default not selected.
Quarantine default to loop	Select the Qurantine handle of gateways making a request to the outside, and default not selected. l Selected: Quarantine using loop mode, the gateways will continually Notify all events as requested after receiving a request.
Enable first digit timer	Select the processing mode when there is no timeout parameter in the outside request received by the gateways, and default not selected. l Selected: the gateways will report timeout in terms of its own timeout setting (the time interval set in non-dial timeout of configuration system parameters) when subscribers hasn’t dialed up in time after offhook.
Using configured digit map	Select whether to activate the digit map configured by local gateway, and default value is not selected.
Using notify instead of 401/402	Set whethr the gateways report “offhook events” to replace 401 messages in NTFY or report “onhook events” to replace 402 messages in NTFY when responding to messages sent by the proxy server. Default: not selected. l Selected: The gateways will use NTFY message to replace 401 and 402 messages.
No name in default package	Select if a package name is included when the gateways reply to the default package, and default not selected.
Keep connection when on-hook	Select if the gateways actively cancel connection disconnect when subscribers hook on, and default not selected.

2.3.5 FoIP

After login, click the label of “Basic > FoIP” to open this interface.

Figure 2-6 fax configuration interface

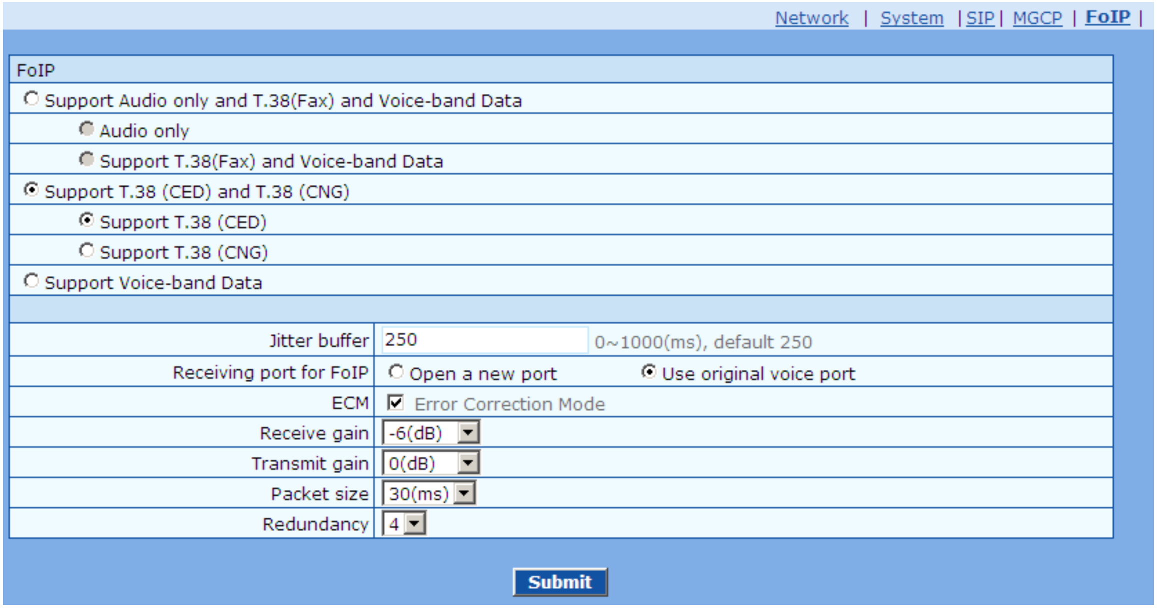


Table 2-8 fax configuration parameters

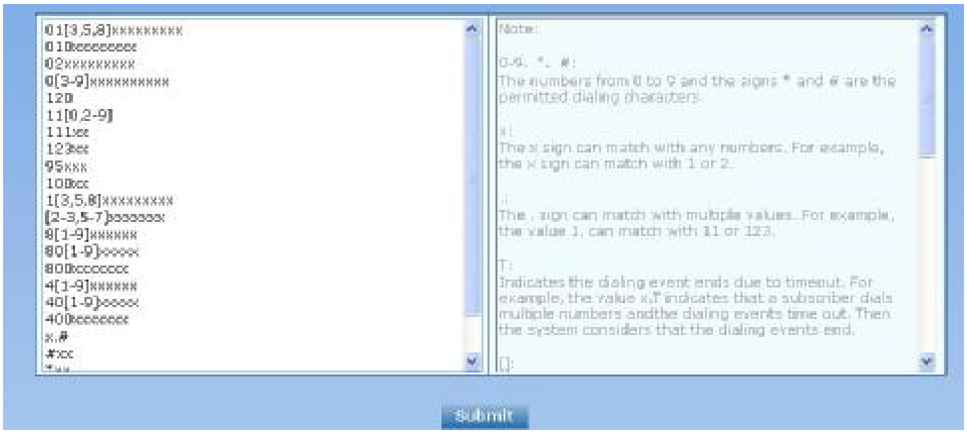
Title	Description
T.30, POS, MODEM only	Audio only Support T.38 (Fax) and voice-band data
T.38 only	Support T.38 (CED) Support T.38 (CNG)
The following are configurable parameters when T.38 activated	
Jitter buffer	Set the extent of T.38 jitter buffer, and the default is 250. The valid range is 40~1000 in milliseconds.
Receiving port for FoIP	Set whether to open a new port when the gateway is switching to T.38 mode, and by default, original voice port will be usd. Open a new port: use the new RTP port. Use original voice port: use the original RTP port that created on call set.
ECM	Determine whether to use corrective mode of fax. By default, it is not selected.
Receive gain	Set the receiving gain of T.38 fax, with the default of 6dB.
Transmit gain	Set the transmission gain of T.38 fax, with the default of 0dB.
Packet size	Set the packet size of T.38. 30 miliseconds is the default value.
Redundancy	Set the number of the redundant frames in T.38 date package, default is 4.

2.4 **Routing**

2.4.1 Digit Map

After login, click “Routing> Digit Map” tab to open the dialing rules interface as shown in Figure 2-7.

Figure 2-7 Configuration Interface for Digit Map



Dialing rules are used to effectively judge if the received number sequence is completed, for the purpose of ending up receiving numbers and sending out the received numbers.The proper use of dialing rules can help to reduce the connection time of telephone calls.

The maximum number of rules that can be stored in gateways is 60. Each rule can hold up to 32 numbers and 38 characters. The total length of dialing rules table (the total length of all dialing rules) can be up to 2280 bytes.

The following provides a description of tipical rules:

Table 2-9 Description of Digit map

Digit map	Description
“x”	Represents any number between 0-9.
“.”	Represents more than one digit between 0-9.
“##”	End after receiving two-digit dialing “##”. “##” is a special dialing for users to receive gateway IP address and version number of firmware by default.
“x.T”	The gateways will detect any length of telephone number starting with any number between 0-9. The gateways will send the detected number when it has exceeded the dialing end time set in system parameter configuration and hasn’t received a new number.
“x.#”	Any length of telephone number starting with any number between 0-9. If subscribers press # key after dial-up, the gateways will immediately end up receiving numbers and send all the numbers before # key.
“*xx”	End after receiving * and any two-digit number. "* xx" is primarily used to activate function keys for supplementary services,such as CRBT, Call Transfer, Do not Disturb, etc.
“#xx”	End after receiving # and any two-digit number. “#xx” is primarily used to stop function keys for supplementary services, such as CRBT, Call Transfer, Do not Disturb, etc.

Digit map	Description
[2-8]xxxxxx	A 7-digit number starting with of any number between 2-8, used to end the dialing.
02xxxxxxxxx	A 11-digit number starting with 02, used to end the long-distance dialings starting with “02”.
013xxxxxxxxx	A 12-digit number starting with 013, used to end long-distance calls
13xxxxxxxxx	A 11-digit number starting with 13, used to end the dialings.
11x	A 3-digit number starting with 11, used to end the emergency calls.
9xxxx	A 5-digit number starting with 9, used to end special service calls.
17911 (eg.)	Send away when the set number, like 17911, is received.

Dial rules by default as follows:

01[3,5,8]xxxxxxxxx

010xxxxxxxx

02xxxxxxxxx

0[3-9]xxxxxxxxxx

11[0,2-9]

111xx

9[5,6]xxx

100xx

10[1-9]

12[0-2,4-9]

1[3,5,8]xxxxxxxxx

[2-3,5-7]xxxxxxx

[4,8][1-9]xxxxxx

[4,8]0[1-9]xxxxx

[4,8]00xxxxxxx

x.T

x.#

#xx

*xx

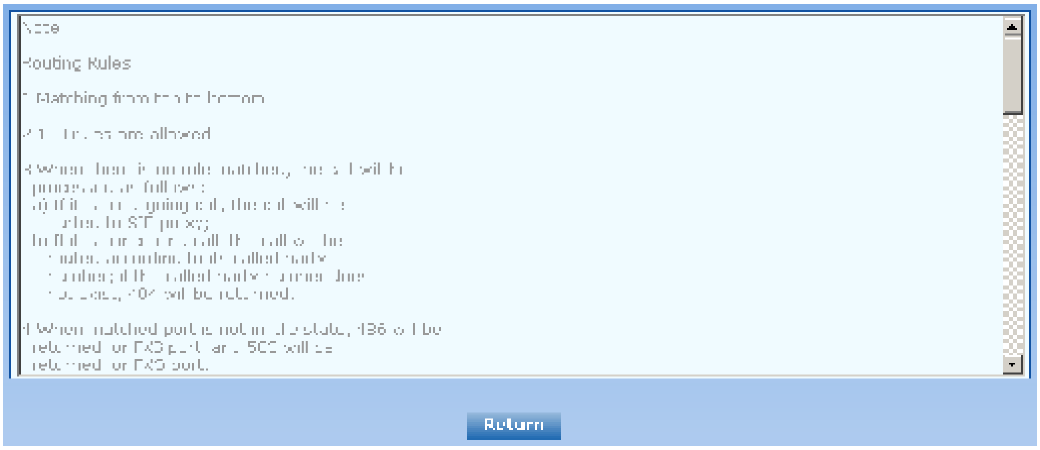
##

2.4.2 Routing Table

After login, click “Routing> “Routing Table” tab to open the configuration interface.

Figure 2-8 Configuration Interface for Routing Table

Click “Help” to open the illustrative interface for routing configuration



The routing table with 500 rules in capacity provides two functions including digit transformation and call routing assignment. Here are the general rules applied by gateways when executing the routing table:

CAUTION

Rules must be filled out without any blank at the beginning of each line; otherwise the data can’t be validated even if the system prompts successful submital.

The routing table is empty by default. The gateways will point a call to the SIP proxy server when there is no matched rule for the call.

The format of number transformation is

Source Number Replacement Method

For example: “FXS 0755 REMOVE 4” means removing the prefix 0755 of the called number for calls from FXS port, where “FXS” is source, “0755” is number, and “REMOVE 4” indicates the method of number transformation.

The format of routing rules is

Source Number ROUTE Routing Destination

For example: “IP 800[0-3] ROUTE FXO 1,2,3,4” means routing calls from IP with called number between 8000~8003 to FXO port in a sequential selecting order of 1, 2, 3, 4. Namely, FOX Port 2 is selected when FXO Port 1 is busy and so on.

Detailed definitions of source and number, number transformation moethods and routing destination are shown below.

Table 2-10 Routing Table Format

Name	Description
Source	There are three types of source: IP, FXS and FXO. Among them, IP source can be any IP address and is denoted by “IP”; “IP [xxx.xxx.xxx.xxx]” is used to denote specific IP address; “IP [xxx.xxx.xxx.xxx: port]” is used to denote specific IP address with port number. FXS and FXO ports can be any port, represented with “FXS” or “FXO”; special lines can be represented with FXS or FXO + port number, eg. FXS1, FXO2 or FXS [1-2], etc.
Number	It chould be a calling number with the form of CPN + number or a called number with the form of number. The number may be denoted with digit 0-9,"*",".","#"," x ", etc., and uses the same regular expression as that of dialing rules. Here are some of the form of number: - Designate a specific number: eg.114，83501950; - Designate a number matching a prefix: such as 61xxxxxx. Note: the matching effect of 61xxxxxx is different from that of 61x or 61. Number matching follows the principle of “minimum priorary matching ” - Specify a number scope. For example, 268[0-1，3-9] specifies any 4-digit number starting with 268 and followed by a digit between 0-1or 3-9; Note: Number matching follows the principle of “minimum matching ”. For example: x matches any number with at least one digit; xx matches any number with with at least two-digit; 12x matches any number with at least 3-digit starting with 12.

Table 2-11 Number Transformations

Processing Mode	Description and Example
KEEP	Keep number. The positive number behind KEEP means to keep several digits in front of the number; the negative number means to keep several digits at the end of the number. Example: FXS 075580501950 KEEP -8 Keep the last 8 digits of the called number 075583501950 for calls from FXS. The transformed called number is 83501950.
REMOVE	Remove number. The positive number behind REMOVE means to remove the first several digits of the number; the negative number means to remove the later several digits of the number. For example: FXS 0755 REMOVE 4 Remove 0755 of the called number beginning with 0755 for calls from FXS.

Processing Mode	Description and Example
ADD	Add prefix or suffix to number. The positive number behind ADD is the prefix; the negative number is suffix. Example 1: FXS1 CPNX ADD 0755 FXS2 CPNX ADD 010 Add 0755 in front of calling numbers for calls from FXS port 1; add 010 in front of calling numbers for calls from FXS port 2. Example 2: FXS CPN6120 ADD -8888 Add 8888 at the end of the calling number starting with 6120 for calls from FXS port.
REPLACE	Number replacement. The replaced number is behind REPLACE. Example: FXS CPN88 REPLACE 2682000 Replace the calling number beginning with 88 for calls from FXS port to 2682000.
REPLACE	Other use of REPLACE is to replace the specific number based on other number associated with the call. For example, replace the calling number according to the called number. Examples: FXS 12345 REPLACE CPN-1/8621 FXS CPN13 REPLACE CDPN0/0 For calls from FXS ports with called party number of 1234, removing one digit at the rear of the calling number and add 8621; for calls from FXS ports with calling party number starting with 13, add 0 in front of the called number.
END or ROUTE	End of number transformation. From top to bottom, number transformation will be stopped when END or ROUTE is encountered; the gateways will route the call to the default routing after meeting EDN, or route the call to the designed routing after meeting ROUTE. Example 1: FXS 12345 ADD -8001 FXS 12345 REMOVE 4 FXS 12345 END Add suffix 8001 to the called number starting with 12345 for calls from FXS ports, then remove four digits in front of the number to end number transformation. Example 2: IP[222.34.55.1] CPNX. REPLACE 2680000 IP[222.34.55.1] CPNX. ROUTE FXS 2 For calls from IP address 222.34.55.1, calling party number is replaced by 2680000, and then the call is routed to FXS port 2 with the new calling party number.
CODEC	Designate the use of codec, such as PCMU/20/16, where PCMU denotes G.711, /20 denotes RTP package interval of 20 milliseconds, and /16 denotes echo cancellation with 16 milliseconds window. PCMU/20/0 should be used if echo cancellation is not required to activate. Example: IP 6120 CODEC PCMU/20/16 PCMU/20/16 codec will be applied to calls from IP with called party number starting with 6120.

Processing Mode	Description and Example
RELAY	Insert prefix of called party number when calling out. The inserted prefix number follows behind REPLAY. Example: IP 010 RELAY 17909 For calls from IP with called party number starting with 010, digit stream 17909 will be outpulsed before the original called party number being sending out.

Table 2-12 Routing Destination

Destination	Description and Example
ROUTE NONE	Calling barring. Example: IP CPN[1,3-5] ROUTE NONE Bar all calls from IP, of which the calling numbers start with 1, 3, 4, 5.
ROUTE FXS	Route a call to FXS ports. Example 2: IP 800[0-3] ROUTE FXS 1 Point this call to FXS port 1. Example 3: IP 800[0-3] ROUTE FXS 1,2,3,4/g For terminating the call from IP, ring FXS port 1, 2, 3, 4 simultaneousely.
ROUTE FXO	Route a call to FXO port. Example: IP x ROUTE FXO 1,2,3,4/r Select the outgoing call FXO port in a round robin way.
ROUTE IP	Route a call to the IP platform. Example: FXS 021 ROUTE IP 228.167.22.34:5060 228.167.22.34:5060 is the IP address of the platform.

2.4.3 Application Examples of Routing Table

Some typical functions that can be realized by the routing table are provided in this section:

1) One Phone with Double Numbers
2) Hunting Group
3) Outbound Call Barring
4) FXO Port Hunting for Outbound Call

One Phone with Double Numbers

The hand set connected to gateway can be configured with two numbers through One Phone with Double Numbers. For example, port FXS1 is set with PSTN number 83501950 and extention number 1001 for internal calling

Routing Setting

FXS 1001 ROUTE IP 127.0.0.1:5060

IP 1001 ROUTE FXS 1

Description:

1) Send the call with the called number starting with 1001 from FXS port to port 5060 of gateway's local IP;

2) Send the call with the called number starting with 1001 and from any IP to the FXS port 1.

Configuration number of FXS1 itself is 83501950, so the call of this number is not required to write specialized routing.

Hunting Group

A hunting group can be associated with a set of FXS ports, and an inbound call from IP or FXO ports can be routed to a hunting group. There are three circuit selection algorithms available: 1) sequential selection, 2) circular selection, and 3) simultaneous ringing.

Routing Setting:

Take WSS8-4S/4 gateway as an example. Send the inbound call from IP trunk and analog line in a circular way to the phone set on the 2nd or 3 rd FXS port.

FXO x ROUTE IP 127.0.0.1:5060

IP x ROUTE FXS 2, 3

Description:

1) Send all calls from FXO port to port 5060 of gateway's local IP;

2) Send all inbound calls from any IP (inside and outside) to the 2nd or 3rd FXS port in sequence. Namely, the 2nd FXS port is selected firstly when it is free, otherwise the 3rd port is selected.

Outbound Call Barring

Restrict users to dial certain telephone numbers, such as an international call. Examples are as follows:

Routing Setting	Description
FXS[1] 0 ROUTE NONE	A calling starting with 0 is barred to dial using the phone set at FXS1 port.
FXS[1-4] 00 ROUTE NONE	A calling starting with 00 is barred to dial at 1-4 FXS ports.
FXS CPN2 ROUTE NONE	The telephone whose calling number starts with 2 at FXS port is barred to call out.

FXO Port Hunting with Outbound Calls

Routing Setting (take WSS8-4S/4 as an example):

FXS x ROUTE IP 127.0.0.1:5060

IP x ROUTE FXO 1,2,3,4/r

Description:

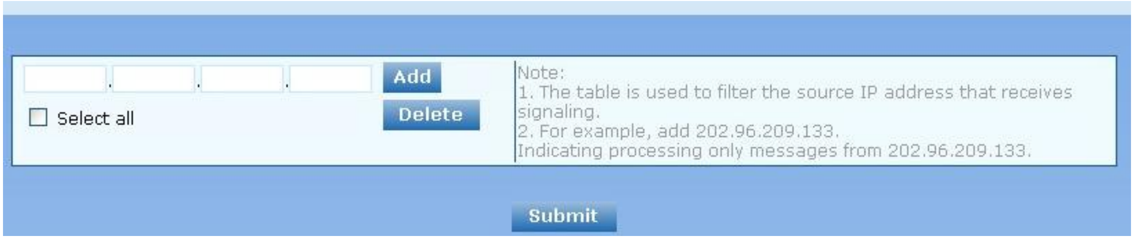
1) Send all calls from FXS port to port 5060 of gateway's local IP (this port must be consistent with the local port in "Configuring SIP");

2) Send all calls from any IP to FXO port for round selection in an order;

2.4.4 IP Table

After login, click "Routing> "IP Table" tab to open the configuration interface.

Figure 2-9 Configuration Interface for IP Table



This table is designed to ensure the safe use of gateways. Administrators can add the authorized IP addresses to this table, and the gateways will only process the information from authorized IP addresses. If the IP table is empty, the gateways will not perform IP address-based message filtering.

For example: the gateway will only process the messages from 202.96.209.133 after adding 202.96.209.133 to its IP table.

2.5 Line Configuration

2.5.1 FXS Phone Number

After login, click “Line > FXS phone number” tab to open the configuration interface.

Figure 2-10 Configuration Interface for Telephone Number

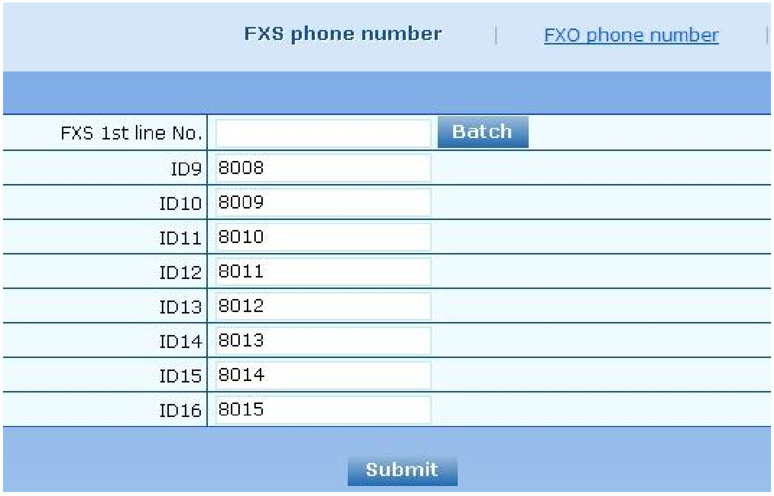


Table 2-13 Configuration Parameters of Telephone Number

Name	Description
FXS 1st line No.	This number is used for the batch setup of consecutive number of subscriber line. Click “Batch ”after filling in initial number, the number of Line 1 adopts initial number; that of Line 2 increases 1 progressively based on that of Line 1, and so on. You needn’t fill in if you do not use batch configuration or the number is not consecutive.
ID n	Fill in the telephone number associated with the subscriber line n (FXS port). This should be manually performed if Batch mode is not used.

2.5.2 FXO Phone Number

After login, click “Line > FXS phone number” tab to open the configuration interface.

Figure 2-10 Configuration Interface for Telephone Number

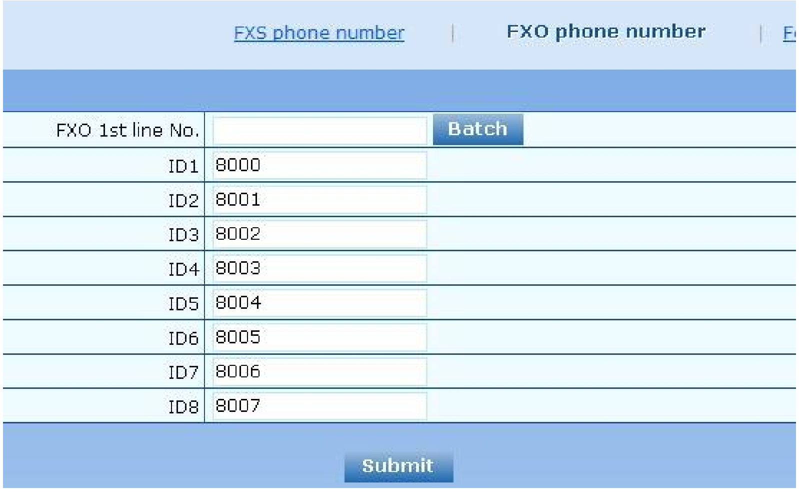


Table 2-13 Configuration Parameters of Telephone Numbers

FXO 1st line No.	This number is used for the fast setup of consecutive number of trunk line. Click "Batch " after filling in initial number, the number of Line 1 adopts initial number; that of Line 2 increases 1 progressively based on that of Line 1, and so on. You needn’t fill in if you do not use batch configuration or the number is not consecutive.
ID n	Fill in the telephone number associated with the trunk n (FXO port). This should be manually performed if Batch mode is not used.

2.5.3 Subscriber Line Features

This page is only used for configuring gateways with subscriber lines (FXS port).

After login, click “Line > Feature” tab to open the configuration interface.

Figure 2-11 Configuration Interface for Subscriber Line Features

Table 2-14 Configuration Parameters of Subscriber Line Features

Name	Description
Line ID	Select a subscriber line required to configure. “FXS -n” corresponds to the “Line > FXS phone Number > ID n”. Copy the configuration of “FXS -n” for selected line to “FXS -n+1”~“FXS - m” by clicking “Batch”, where n indicates the current selected subscriber line number and m indicates the total number of subscriber lines.
Phone number	Display the Telephone Number of this line set in “Line Configuration > FXS phone Number”. Users can input or change the telephone number here.
Registration	Setelect if this line is required to register to softswitch. This is selected as default.
Password	If the “Registration” is selected, users need to enter the authentication password for register of this line here.
Note: The following features are valid only in SIP protocol. When the gateways use MGCP protocol, features are controled by the proxy server without the need of setting on gateways.	
Hot line	Select if the gateways are required to automatically dial out the hotline number after offhook. By default, hot line is disabled. l Disable hot line: Close this feature. l Hot line: Automatically dial out the hotline number after offhook. l Delayed hot line: Automatically dial out the hotline number when the offhook is timeout with a time delay of 5 seconds.
Hot line number	After the hotline function is activated on this line, the hotline number must be entered here.

Name	Description
CRBT	CRBT stands for Color Ring Back Tone. Set if CRBT is activated, that is, provide prepared audio package as ringback tone. Note: it is required to set the CRBT file download platform. This is not selected by default. WSS8 support two CRBT storage modes: on-system (stored in a flash memory) and run-time download (from FTP server). The capacity of both modes are described as follows: On-sytem: - WSS8: No more than 20 seconds in G.729 format (fring1.dat) Run-time download: - WSS8: Up to 20 tone files, a maximum of 10000 seconds in G.729 format or 1250 seconds in G.711 format. Note:Tone files are stored in the flash memory and the gateways automatically download the tone files from FTP server after power on.
CRBT ID	Set the CRBT number with a valid rang of 0~255, where 0 indicates disabling CRBT. The default value is 0.
Speed dials	Select if the Speed dials is activated on this line. By default, this is not selected.
Speed dial list	If the Speed dials is activated on this line, enter the speed dials list. The abbreviated number consists of max 30 pairs of "abbreviated number-real number" with an minus sign between them; "abbreviated number-real number" pairs are separated by "/"; the value range of abbreviated number is 20 ~ 49. For example: 20-83501950/23-13952475822/30-96961. Users can set the list on a telephone set and display it here.
Call forwarding	Select if Call forwarding is activated on this line. By default, it is not selected.
CFU	If it is required to forward all incoming calls unconditionally, enter the new destination number here.
CFNR	If it is required to forward an incoming call when there is no answer, enter the new destination number here.
CFB	If it is required to forward an incoming call when it is busy, enter the new destination number here.
Forking	Select if the Forking is activated. Forking allows the gateway to initiate a call to another telephone terminal while ringing on this line terminal, and the answer by either terminal will end up with ringing of the other terminal.
Forking number	If forking of this line is activated, set a number for the second ringing terminal here. The ringing terminal can be another port of gateways or an external terminal such as mobile phone.
Release control by caller	Select if the call release is controlled by the caller. By default, this is not selected. - Selected: The gateway will immediately release the call upon caller hanging up; the gateway will not release the call as long as the caller is still off until timeout (60 seconds by default); - Unselected: The gateway will immediately release the call upon either party hanging up the call.
Call waiting	Select if Call waiting is activated on this line. By default this is not selected.
CID on call waiting	Select if Caller ID Display is activated on this line during call waiting. By default, this is not selected.

Name	Description
Call hold	Select if Call Hold is activated on this line. By default this is not selected. Note: If this function is activated, the gateways will automatically activate Call Transfer (Either party may transfer the current call to a third party).
Caller Transfer	Select if Caller Transfer is activated on this line. By default, this is not selected. When A calls to B, B picks up the call and transfers the call to C,. Note: The call hold must be activated before caller transfer.
Caller ID display	Set whether Caller ID display is activated on this line. By default, this is selected. Note: In addition to number display, the Caller ID can display the names of incoming calls as long as terminal equipments support.
Caller ID restriction	Set whether the number of this telephone is sent to the called party with support from platform. By default this is not selected.
Outgoing call barring	Select if outgoing calls are barred on this line. By default, this is not selected.
DND	Select if “Do Not Disturb” is activated on this line. By default, this is not selected.
Direct Dialing in (DDI)	Set whether DDI (Direct Dialing In) is activated, By default this is not selected. Different from FXS, DDI is only used for incoming calls, and the gateways will not send dial tone after off-hook (calling in) on user side. Note: Reverse polarity signal must be activated on the gateways when DDI is used.
Maintenance	Select if the line is set to maintenance status, namely, stop to supply of power for the line port. By default, this is not selected.
Polarity reversed signal	Select if reverse polarity signal is activated on this line. By default, this is not selected. Note: The gateways will provide reverse polarity signal when the phone is connected after this feature is activated.
Subscribe MWI	Select if voice mail service is activated, and by default this is not selected. (Used with “Advanced > SIP” Interface “MWI subscription” Configuration)

2.5.4 Trunk Line Features

This page is only used for configuring gateways with trunks (FXO port).

After login, click “Line > Trunk” tab to open the configuration interface.

Figure 2-12 Configuration Interface for Trunk Line Features

Table 2-15 Configuration Parameters of Trunks

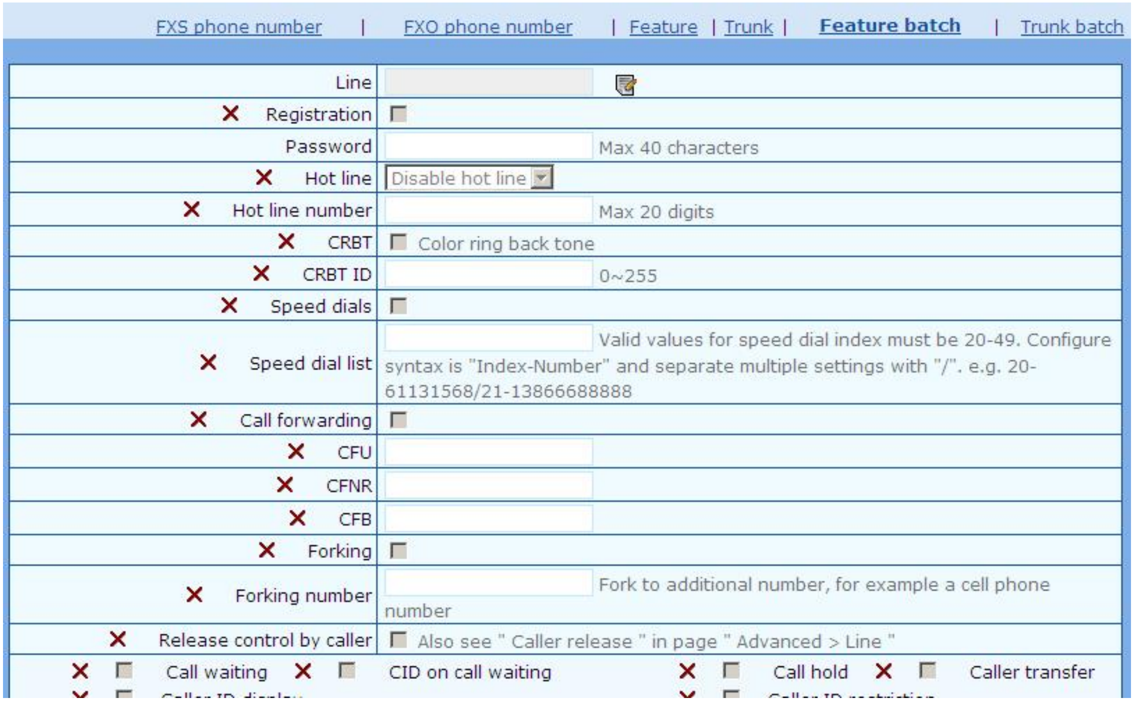
Name	Description
Trunk ID	Select a trunk line required to configure. “FXO-n” corresponds to the “Line > FXO phone Number ID n”. Copy the configuration of “FXO-n” for selected line to “FXO-n+1”~“FXO- m” by clicking “Batch”, where n indicates the current selected trunk number and m indicates the total number of trunks.
Phone number	Display phone number associated with the trunk set in “ Line > FXO phone Number”
Registration	Select if this trunk registers with the SIP registeration server. By default, this is selected.
Password	If the “Registration” is selected, the authentication password for register of this line must be entered here.
Note: The following features are valid only in SIP protocol. When the gateways use MGCP protocol, the control of all call services is provided by the proxy server without the need of setting.	
Inbound handle	The gateways provide two scenarios for handling incoming calls of FXO port: l “Second stage dialing”: When a telephone call comes to the FXO port, the gateways will provide the second dial tone and route the call according to the extension number pressed in. Note: dialing tone or voice prompt file can be changed by user. l “Binding”: When a telephone call comes to the FXO port, the gateways will route the call to a FXS port according to the DID number bound with the port. Note: Setting a number to be bound is required or this setting is invalid.
Polarity reversed signal detection	If a PSTN line supports reverse polarity, make a selection here. Or this setting is invalid. By default, this is not selected.
Caller ID detection	Select if the detection function of caller ID for this FXO port is enabled. By default, this is selected.
Outgoing call barring	Select if this FXO port bars outgoing call service to PSTN. By default, this is not selected.

Name	Description
Echo cancellation	Select if echo cancellation is enabled for this FXO line.By default, this is selected.
Connect signal delay	After making an outgoing call from a FXO port, the gateway will send a 200 OK message to the platform with delay if this parameter is selected. If unselected, the system sends a 200 OK message to the platform after off hook on the FXO port. Used with the configuration item of “Answer delay” on the “Advanced > Trunk” interface.

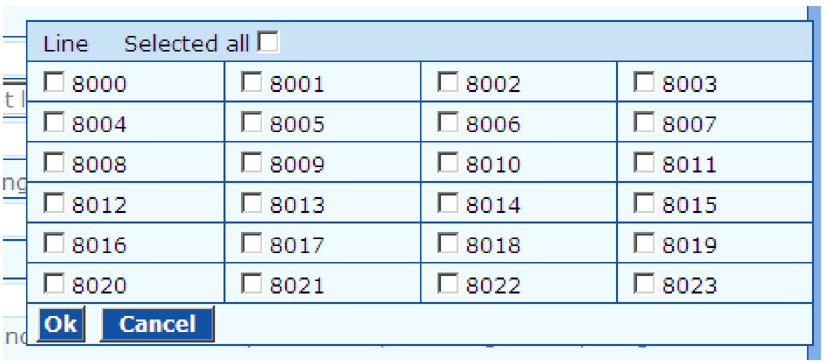
2.5.5 Feature Batch

After login, click “Line > Feature Batch” to open this interface.

Figure 2-13 feature batch configuration interface



Click , the following interface is shown. Choose batch configured features and click “ok”

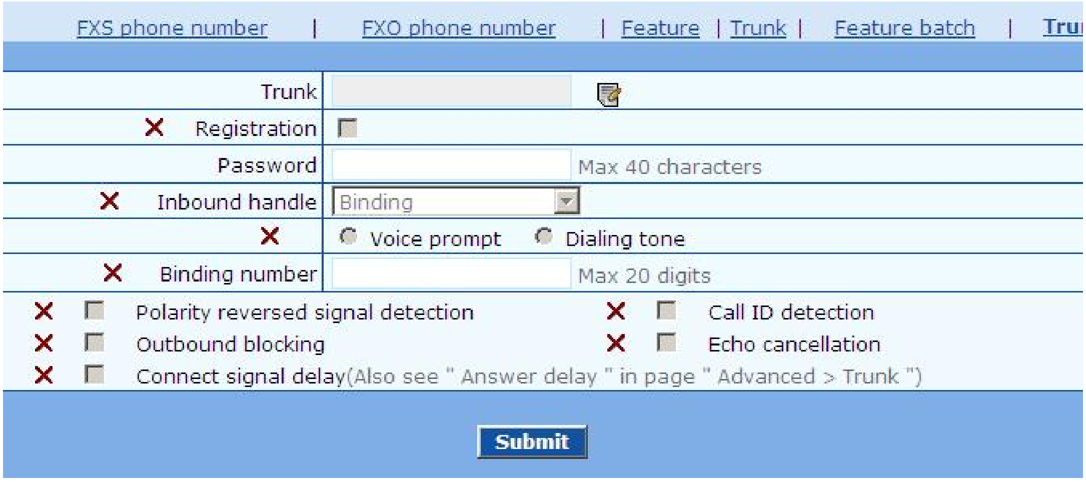


Click to choose whether to activate this function to configurate this parameter. Seen in “Subscriber Line Features”.

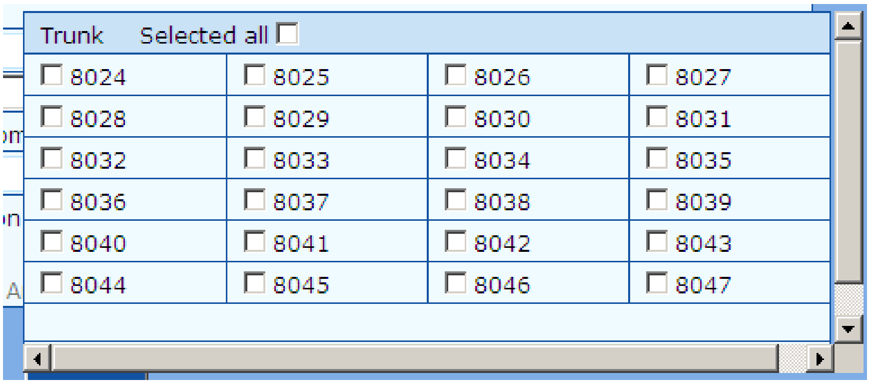
2.5.6 Trunk Batch

After login, click “Line > Trunk Batch” to open this interface.

Figure 2-14 Trunk Batch configuration interface



Click , the following interface is shown. Choose batch configured trunks and click “ok”



Click to choose whether to activate this function to configurate this parameter. Seen in “Trunk Line Features”.

2.6 Advanced Configuration

2.6.1 System

After login, click the label of “Advanced > System” to open this interface.

Figure 2-15 Inferface of system advanced configuraiton

Table 2-16 Parameters of system advanced configuration

Title	Explanation
NAT	
NAT traversal	Gateways support several mechanisms for NAT traversal. Usually, static NAT is used when fixed public IP address is available. It's necessary to perform port mapping or DMZ function on router when choosing dynamic or static NAT.
Refresh period	The refresh time must be filled in here when choosing dynamic NAT or STUN traversal. Besides, refresh time interval shall be determined by giving consideration into the NAT refresh time of the LAN router which the gateway is located. Gateway's NAT holding function and STUN function will carry out periodically operation accoding to this parameter. With second as its unit, default value of 60 seconds.
SDP Address	This parameter determines the IP address used in transmitted SDP. ı WAN IP Address: Apply NAT address into the transmitted SDP; ı Local IP Address: Apply the gateway's IP address into the transmitted SDP. Note: The parameter should come into effect only on condition that gateway successfully obtained NAT address.
NAT IP address	This parameter must be filled when using static NAT traversal, in which gateway works under LAN and the WAN address is fixed. The WAN address should be filled in this field, which will be used in SDP. This parameter can be set in IP address format or hostname format (note: DNS service should be activated when hostname format is used). There is no default value for this field.
STUN server	Set the IP or domain name of STUN server. No default value. If the set is empty, the gateway will adopt the STUN server address configured at factory. When choosing STUN for NAT traversal, the gateway will carry out STUN operation periodically according to the configured interval time of NAT refresh.
RTP Receiving Port	The gateways will send the RTP receiving port selected here to the remote side. ı NAT port: Use NAT mapped port， which is obtained through STUN, for example; ı Local port: Use local SIP and RTP port.

Title	Explanation
Remote management	
Remote management	The gateways support EMS which is a centralized gateway management server provided by Real Tone, and Auto-provision.
EMS	
Server URL	User needs to enter the IP address and port of EMS server for activating EMS service.
Auto provision	
Server URL	Gateways may download software upgrade packages and configuration files automatically through auto-provision server. Once the auto provision is selected, you have to enter the IP address of ACS here.

2.6.2 Media Stream

After login, click the label of "Advanced > Media Stream" to open this interface.

Figure 2-16 Media stream configuration interface

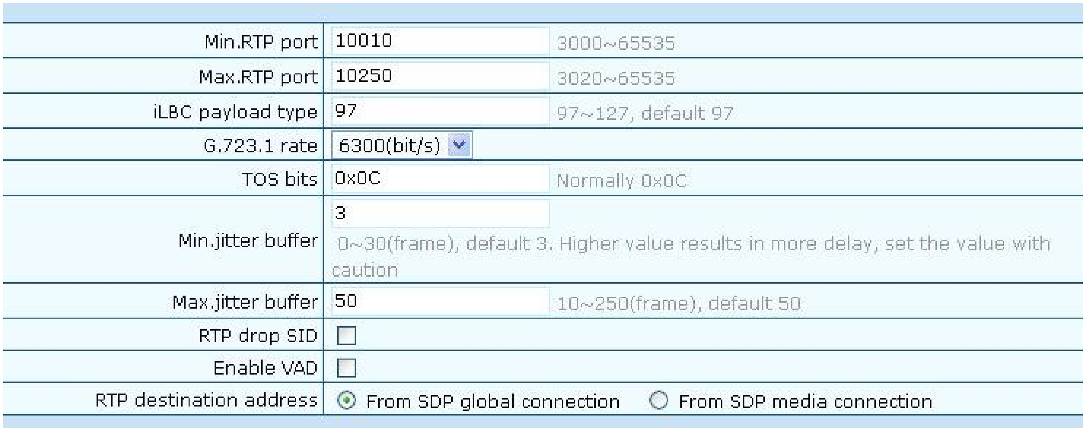


Table 2-17 Media stream configuration parameter

Title	Explanation
Min. RTP port	The minimum value of UDP ports for RTP transmission and receiving, and the parameter must be greater than or equal to 3000. This field must be filled in. Note: each phone call will occupy RTP and RTCP ports. If the gateway is equipped with 4 subscriber lines (or trunk line), then at least 8 UDP ports are needed.
Max. RTP port	The maximum values of UDP ports for RTP's transmission and receiving. This field must be filled in. It's advisable to be greater than or equal to "2× number of lines ＋min. RPT port".
iLBC payload type	Set the RTP payload type of iLBC，and the default value is 97. Accepted value is 97 ~ 127. The parameter shall be configured in conformity to that of platform.

Title	Explanation
G.723.1 rate	Set G.723.1 coding rate，the default value is 6300. The optional parameters are followings: ı 5300: the Bit rate is 5.3k per second; ı 6300: the Bit rate is 6.3k per second
TOS bits	This parameter specifies the quality assurance of services with different priorities. The default value is 0x00. Eg: TOS=0xB8 indicates level 5 that has no reliability requirement.
Min. Jitter buffer	RTP Jitter Buffer is constructed to reduce the influence brought by network jitter. This default value is 3.
Max. Jitter buffer	RTP Jitter Buffer helps to reduce the influence brought by network jitter. The default value is 50.
RTP drop SID	Determine whether to discard received RTP SID voice packets. By default, SID voice packets will not be dropped. Note: RTP SID packets should be dropped only when they are in unconformity to the specifications. Nonstandard RTP SID data could generate noise for calls.
Enable VAD	Only applicable to G.723, GSM, iLBC. In case of selecting this parameter, it will not send any voice packet during mute period. By default, this is selected.
RTP destination address	This parameter determines where to obtain the IP address of the receiving side for RTP packets. By default, the IP address is obtained “From SDP global connection”. ı From SDP global connection: Obtain the IP address from SDP global connection; ı From SDP media connection: Obtain the IP address from SDP Media Description.

2.6.3 SIP related configuration

The SIP messages consist of request message and response message. Both include SIP message header field and SIP message body field. SIP message header maily describes the message sender and receiver; SIP message body mainly describes the specific implementation method of the dialog.

Message of request: the SIP message sent by a client to the server, for the purpose of activating the given operation, including INVITE, ACK, BYE, CANCEL, OPTION and UPDATE etc.

Message of response: the SIP message sent by a server to the client as response to the request, including 1xx, 2xx, 3xx, 4xx, 5xx, and 6xx responses.

Message header: Call-id.

Parameter line: Via, From, To, Contact, Csq, Content-length, Max-forward, Content-type, White Space, and SDP etc.

WSS gateways provide good flexibility in content setting in order to improve the compatibility with the platform.

After login, click the label of “Advanced > SIP” to open this interface.

Figure 2-17 SIP related configuration interface

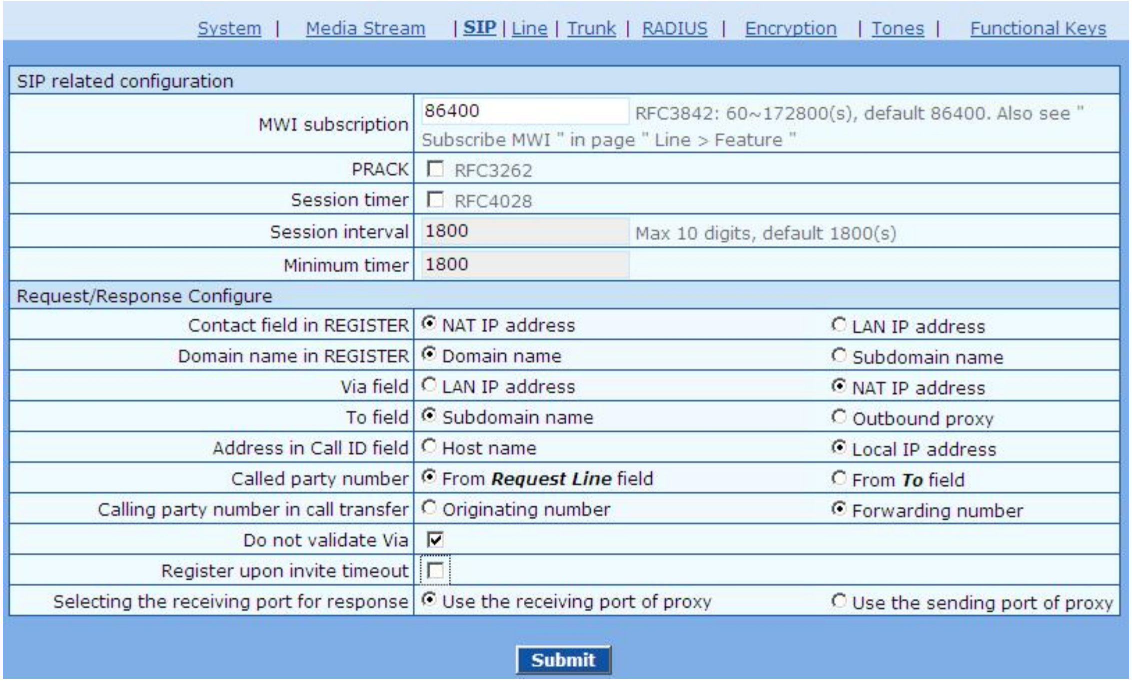


Table 2-18 SIP related configuration parameter

Title	Explanation
SIP related configuration	
MWI subscription	The default is 86400 seconds. The gateway will send platform a message to confirm that has subscribed MWI service at intervals of the time period set here. This parameter should be used in conjuection with voice mail subscription on the page of subscriber line.
PRACK	Determine whether to activate Reliable Provisional Responses. (RFC 3262)
Session timer	Choose to activate session refresh (Session Timer, RFC 4028). By default, session timer is not activated.
Session interval	Set the session refresh interval, the gateway will enclose the value of Session-Expires into INVITE or UPDATE messages. Default value is 1800 in second.
Minimum timer	Set the minimum value of session refresh interval.
Request/Response Configure	
Contact field in REGISTER	Choose the registration mode of gateway under LAN traversal circumstance, the default is “NAT IP Address”. ı LAN IP address: Keep original content of “Contact” when register; ı NAT IP address: Use the NAT information returned by registration server.

Title	Explanation
Domain name in REGISTER	The default is “Domain name”. ı Domain name: Complete domain name used for registration (for example: 8801@registrar.RealTone.com); ı Subdomain name: Only use the common part of the name of domain (for example: 8801@RealTone.com).
Via field	Choose whether to use NAT IP address or LAN IP address for “Via” header field value, the default is “NAT IP address”.
To field	Choose whether to apply Domain name or Outbound proxy to “To” header field, the default is “Domain name”.
Call ID field	Choose whether to fill Call ID field with host name or local IP, the default is “local IP address”.
Called party number	Choose whether the gateway acquires the called number from Request Line header field or To header field. The default is “from Request Line”.
Calling party number in call transfer	Under call forwarding, the calling party number sent can be choosed from Originating number or Forwarding number being set for sending, the default is “Forwarding number”. For example: the subscriber line 2551111 on the gateway activates call forwarding feature and set the destination to 3224422. When caller with 13055553333 calls 2551111, the call will be forwarded to 3224422: ı if choose“Originating number”，the number 13055553333 will be sent to 3224422 as calling party number； ı if choose“Forwarding number”，the number 2551111 will be sent to 3224422 as calling party number；
Do not validate Via	Set whether to ignore Via field, By default, Via is ignored.
Register upon INVITE timeout	Set whether to activate registration when SIP message of INVITE is failed or time expired, and by default, re-registration is not selected.
Selecting the receiving port for response	Use the receiving port of proxy or use the sending port of proxy

2.6.4 Characteristics of subscriber line

After login, click the label of “Advanced > line” to open this interface.

Figure 2-18 Subscriber-line characteristics configuration interface

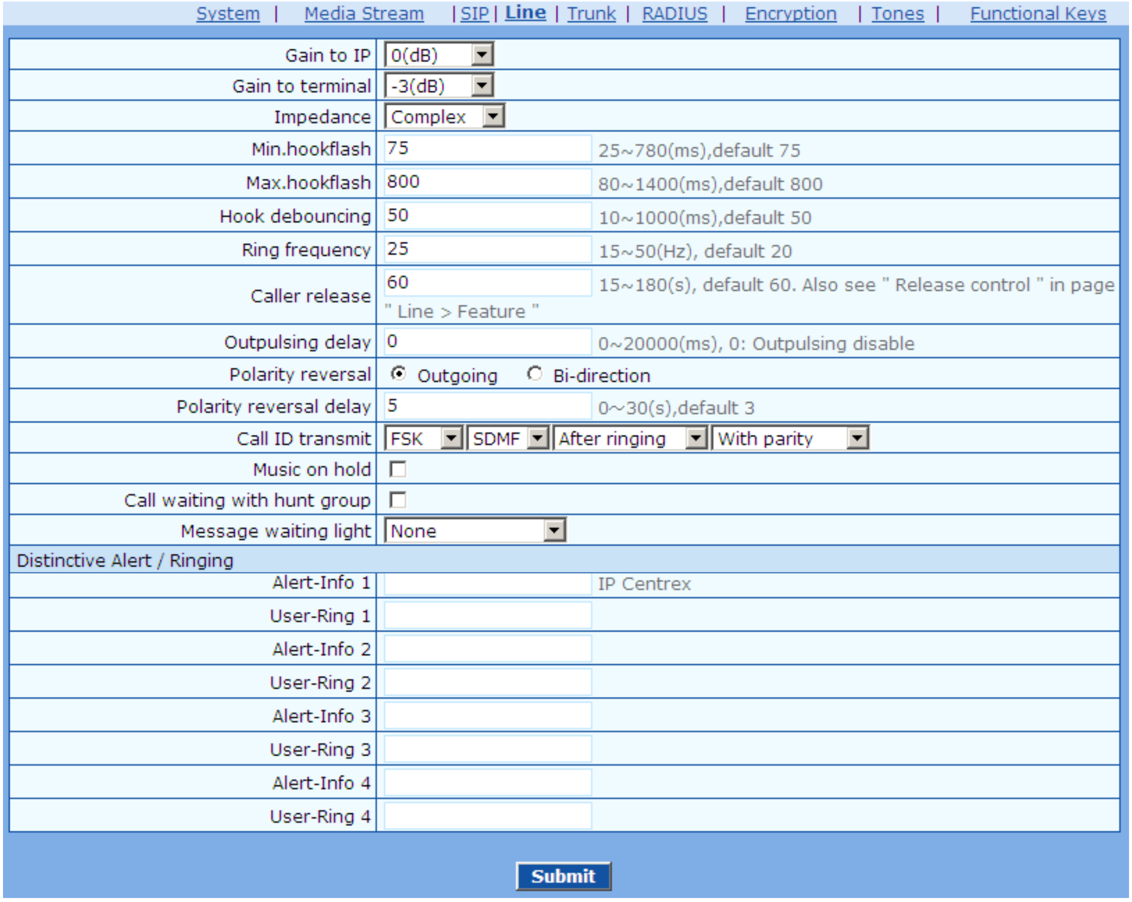


Table 2-19 Subscriber-line characteristics configuration parameter

Title	Explanation
Gain to IP	Set the voice volume gain towarding IP side, the default is 0. Taking decibel as the unit, setting range is -3 ~ +3 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Gain to terminal	Set the voice volume gain towarding FXS port side, the default is -3. Taking decibel as the unit, setting range is -6 ~ +3 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Impedance	Select the parameter of FXS port line impedance, and the default value is 600 ohm. The optional values as below: ı Complex ı 600（ohm） ı 900（ohm）
Min.hookflash	Used by gateway to detect Hook Flash event, the default is 75 milliseconds. The gateway will ignore any flash that fall short of the shortest flash time. Generally, this value should not be less than 75 milliseconds.

Title	Explanation
Max.hookflash	Used by gateway to detect hook flash, the default is 800 milliseconds. The gateway will regard the flash duration between “Min.hookflash” and “Max.hookflash” as effective flash. Any flash lasting over the longest time will be considered by gateway as hang up. Generally, this value should not be less than 800 milliseconds.
Hook debouncing	Used by gateway to avoid the glitch of the phone status, with default of 50 milliseconds. When the duration from hang-up to off-hook falls short of this value, the gateway will ignore the status variation, and consider the phone remains hang-up status. In case of vice versa, the gateway will ignore the status variation, and consider the phone remains off hook status. Effective range of setting is 10~1000 milliseconds.
Ring frequency	Set the ringing frequency to be transmitted by gateway to the phone, ranging from 15 to 50 Hz, with default of 20 Hz.
Caller release	Set the delay release time of line as caller control method, with default of 60 seconds. Effective range of setting is 15~180 seconds.
Outpulsing delay	Used when gateways’ FXS port is connected with the trunk interface of PBXs. For calls from gateway to PBX, gateways will relay the extensions to PBX after the delay set here. Setting of “0” means no extension number relay.The default is 0 millisecond.
Polarity reversal	Set the trigger for polarity reversal the default is “Outgoing”. l Outgoing:Transmit reverse polarity signal only when the outbound is connected; l Bi-direction: Transmit reverse polarity signal for the connection of both inbound and out bound calls.
Polarity reversal delay	The delay time from call being answereed to the transmission of reverse polarity signa. The default value is 3 in seconds. Effective range of setting is 0 ~ 30 seconds.
Call ID transmit	Select transmission mode of Caller ID signal from the FXS port to the phone. l FSK or DTMF; l SDMF or MDMF; l Sending Caller ID data before or after ringing; l Sending Caller ID data with or without parity.
Music on hold	Choose whether to play the background music while call waiting, and the default is not to play.
Call waiting with hunt group	Choose whether to activate hunt group feature for call waiting, Default not selected.
Message waiting light	Choose the lighting method of message waiting indicator of voice mail here: None, Polarity reversed, FSK. Message waiting indicator refers to the special LED on a phone, working with voice mail function. When user gets the latest mail, the gateway will light this lamp upon receiving the notice from platform; the light goes off when there’s no unheard mail. It’s essential to understand whether the phone supports the indicators and lighting method when selecting the lighting method.
Distinctive ring	Apply to enterprise customers
Alert info 1	To match with “User ring 1”. Four patterns of user ring are offered. When the Alert-info value of INVITE message matches with this parameter, “User ring 1” is activated.

Title	Explanation
User ring 1	Configure user ring 1. Eg 1: if the user ring is set “2,500,500,1000,3000”, the ringing effection will display as 0.5s ringing, 0.5s pause; 1s ringing, 3s pause. Eg 2: if the user ring is set “2000,4000”, the ringing effection will display as 2s ringing, 4s pause.
Alert info 2	To match with “user ring 2”
User ring 2	Configure user ring 2
Alert info 3	To match with “user ring 3”
User ring 3	Configure user ring 3
Alert info 4	To match with “user ring 4”
User ring 4	Configure user ring 4

2.6.5 Characteristics of trunk line

After login, click the label of “Advanced > trunk” to open this interface.

Figure 2-19 Trunk line characteristics configuraiton interface

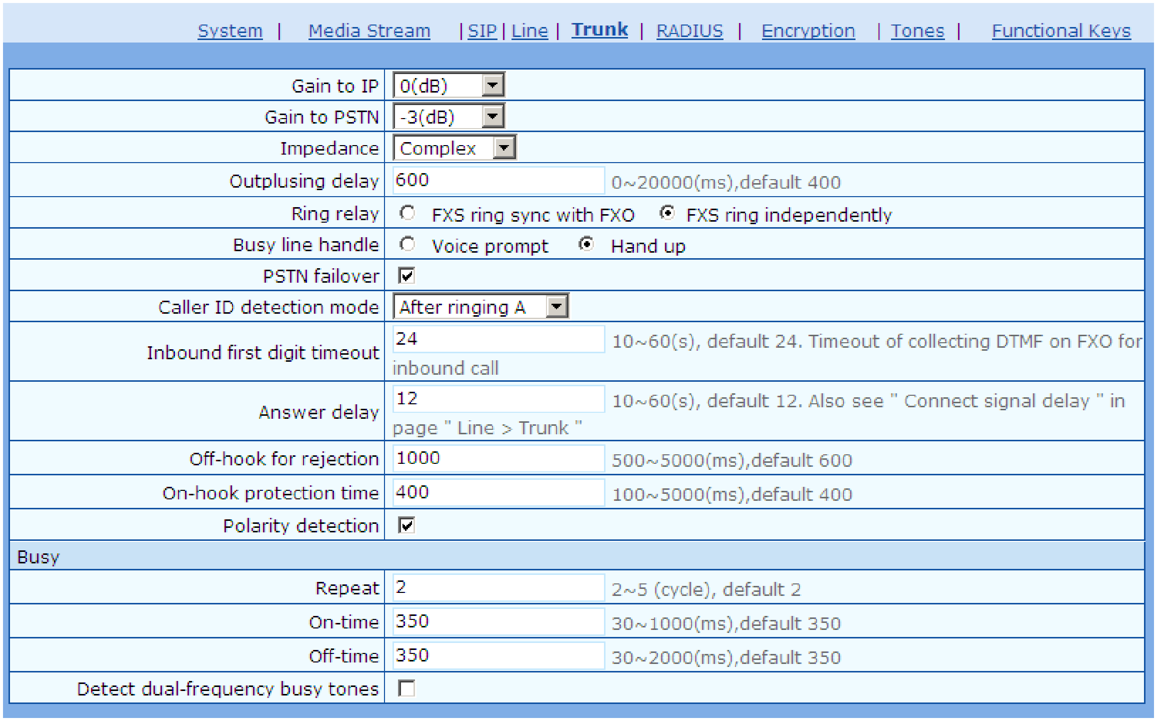


Table 2-20 Configuration parameter of trunk line characteristics

Title	Explanation
Gain to IP	Set the voice volumn gain towarding IP side, the default is 0. Taking decibel as the unit, setting range is -3 ~ +9 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Gain to PSTN	Set the voice volumn gain towarding PSTN side, the default is -3. Taking decibel as the unit, setting range is -6 ~ +9 decibels.

Title	Explanation
Impedance	Set the parameter of FXO line impedance, with the default of 600 ohm. The optional settings as below: l Complex l 600（ohm） l 900（ohm）
Outplusing delay	Set the time interval between FXO going off-hook and starting outpulsing the first digit to PSTN. The default is 400 in milliseconds.
Ring relay	Whether to relay the ring of inbound call to the FXS port when applying to DID. The default is “FXS ring independently”.
Busy line handle	Either a voice prompt or hanging up can be applied to FXO port when an incoming call goes to the FXS port which is in busy. This only applies to DID feature.
PSTN failover	Whether to route a call to PSTN through FXO port when the IP network faults or no response to the call request. Default selected.
Caller ID detection mode.	l After ring A； l After ring B； l Before ring A；； l Before ring B；
Inbound first digit timeout	Set the timeout of calling DTMF on FXO port for inbound calls, ranging from 10-60 seconds, with default of 24 seconds.
Answer delay	Set the delay time of outbound connection ranging from 10-60 seconds, with default of 12 seconds. Working with “Line >Trunk” interface and “Connect signal delay” configuration.
Off-hook for rejection	Used for binding a FXO port with a FXS port. For inbound calls to a FXO port, if the FXS port which binging with the FXO port is in the state of busy line, the gateway will hang up after hook off according to the time set by the parameter, so as to refuse the upcoming call. The duration of off hook is 500~5000 milliseconds, with default of 600 milliseconds.
On-hook protection time	Protection period following hang up of FXO port. During this period, gateway ignores any voltage variation of line. Value range is 100~5000 milliseconds, the default is 400 in milliseconds.
Polarity detection.	Choose whether to activate the detection of reverse polarity signal of FXO port inlet. Note the detection will work only when the trunk supports polarity reversal.
Busy Detection	
Repeat	Gateways will regard the busy tone signal with the repeat times specified here as hang-up signal. Default is 2, effective range is 2 ~ 50.
On-time	Set duration of busy tone signal, the default is 350 in milliseconds.
Off-time	Set the interval time of busy tone, the default is 350 in milliseconds.

2.6.6 Radius call logs

After login, click the label of “Advanced > RADIUS” to open this interface.

Figure 2-20 Configuration interface of Radius call logs

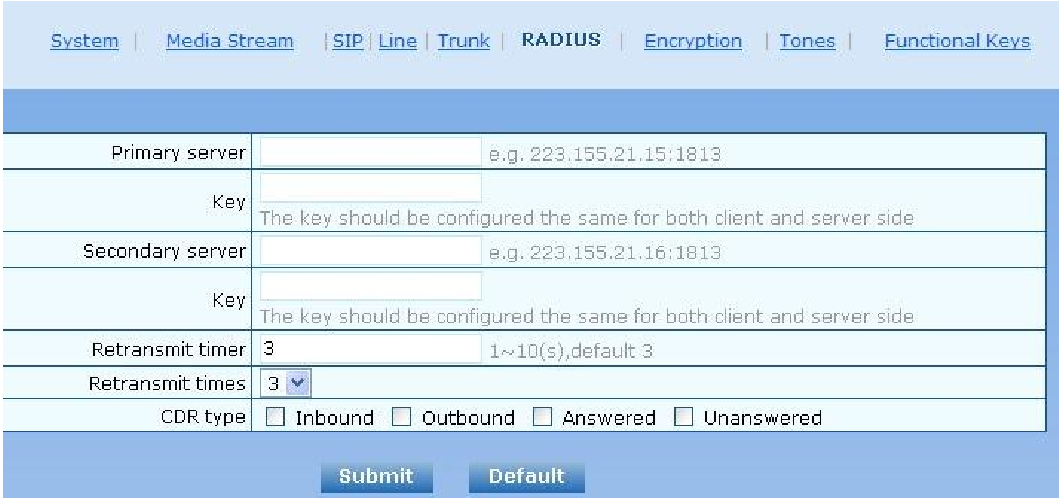


Table 2-21 Configuration parameter of Radius call logs

Title	Explanation
Primary server	Set IP address and port number of preferred Radius server. Note: if the port number is not configured yet, please use Radius default port number of 1813.
Key	Set the share key to be used for encrypted communications between Radius client and server. Note: the share key should be configured the same for both client and server side
Secondary server	Set the IP address and port number of standby Radius server. When the fault appears in communications between gateway and preferred Radius server, the gateway will automatically activate standby Radius server. Note: in case of no configuration of port number, use default port number of 1813.
Key	The share key for communications between Radius client and standby Radius server. Note: the key should be configured the same for both client and server side
Retransmit timer	Set the amount of overtime on response after transmission of Radius message, the default is 3 seconds. The retransmission will be performed If no response is given after the timeout.
Retransmit times	Set the times of retransmission of Radius message when no response is received default is 3 times.
CDR type	ı Outbound: Set whether to send RADIUS charge message for outbound calls; ı Inbound: Set whether to send RADIUS charge message for inbound calls; ı Answered: Set whether to send RADIUS charge message when calls are connected; ı Unanswered: Set whether to send RADIUS charge message for unanswered calls.

2.6.7 Encryption

After login, click the label of “Advanced > Encryption” to open this interface.

Figure 2-21 Encryption configuration interface

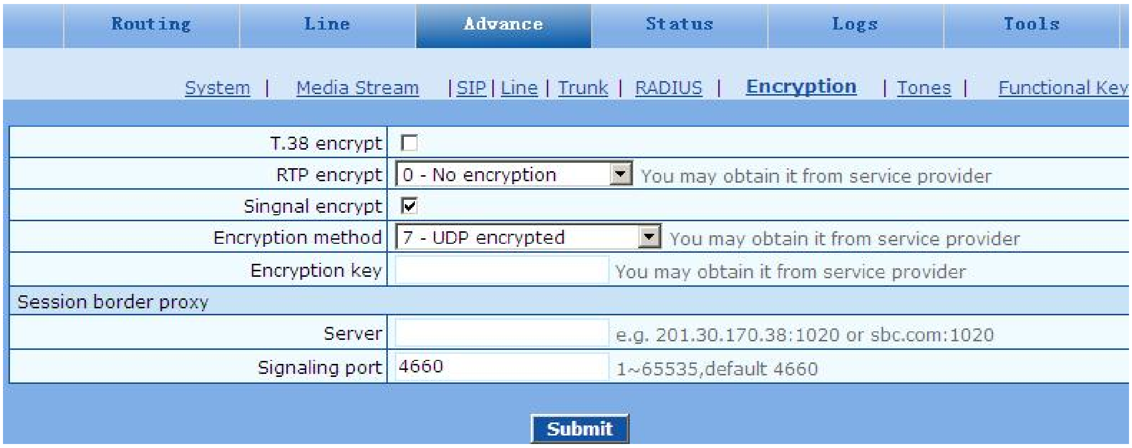


Table 2-22 Encryption configuration parameters

Title	Explanation
Singnaling encrypt	Choose whether to encrypt signaling. By default, this is not selected.
T.38 encrypt	Choose whether to encrypt T38 data. By default, this is not selected.
RTP encrypt	Choose whether to encrypt RTP voice pack, the default is “0” ı 0: No encryption； ı 1: Entire message encryption； ı 2: only encrypt RTP header； ı 3: only encrypt RTP body；
Encryption mode	Set the gateway encryption method, default is 7. The optional parameters as below: ı 2: TCP Not Encrypted； ı 3: TCP Encrypted； ı 6: UDP Not Encrypted； ı 7: UDP Encrypted（Real Tone）； ı 8: Using key words, coordinate with platform； ı 10: RC4； ı 13: Encrypt13, coordinate with platform； ı 14: Encrypt14（Real Tone）； ı 16: Word Reverse（263）； ı 17: Word Exchange（263）； ı 18: Byte Reverse（263）； ı 19: Byte Exchange（263）； ı 20: Word Exchange (VOS).
Encryption key	You may obtain it from service provider

Title	Explanation
Session Border Proxy	
Server	Set the IP address and port number of session border proxy server. The character of “:” must be used between IP address and port number. Server address could be set into IP address or domain name. When domain name is used, “DNS service” must be activated as shown in the page of “configure network parameter”, and “DNS server” must be configured. Example: “201.30.170.38:5060” and “softswitch.com:5060”
Signaling port	Signaling port value of the gateway, the default value is 4660. Signaling port number could be set at will, but can not conflict with other ports of equipment.

2.6.8 Call progress tone plan

After login, click the label of “Advanced > Tones” to open this interface.

Figure 2-22 Call progress tone configuration interface

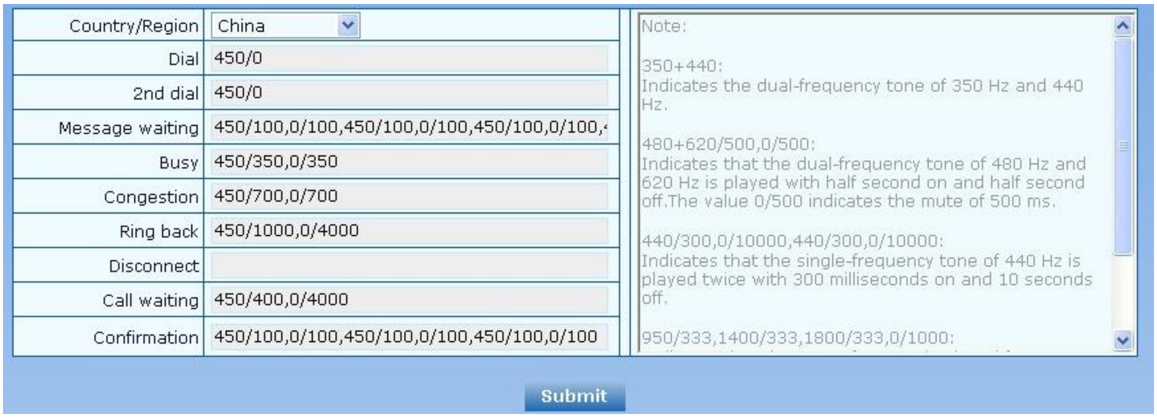


Table 2-23 Call progress tone configuration parameters

Title	Explanation
Country/region	There are progress tone plans for several countries and regions which are pre-programmed in gateways. Users may also specify the tone plan according to the national standard. Gateways provide tone plans for the following countries and regions: China; the United States; France; Italy; Germany; Mexico; Chile; Russia; Japan; South Korea; Hong Kong; Taiwan; India; Sudan; Iran; Algeria; Pakistan; Philippines; Kazakhstan;
Dial	Prompt tone of off-hook dialup
2nd dial	Used for the second stage dialup
Message waiting	Used for prompt of voice mail, or when the subscriber line is set with “Don’t Disturb Service and Call Transfer”.
Busy	Used for busy line prompt
Congestion	Used for notification of call set up failure due to resource limit
Ring back	The prompt tone sent to caller when ring
Disconnect	Used for reminding the subscriber of off-hook and no dialup status of the phone

Title	Explanation
Call waiting	Used for notification in call waiting
Confirmation	Used for confirming function keys being entered.

Here are examples which illustrate the rules of defining call progress tone.

- 350+440

  Indicates the dual–frequency tone consisting of 350 and 440 Hz
- 480+620/500,0/500

  Indicates the dual–frequency tone consisting of 480 and 620 Hz, repeated playing with 500 milliseconds on and 500 milliseconds off. Note: 0/500 indicates 500 milliseconds mute.
- 440/300,0/10000,440/300,0/10000

  Indicates 440 Hz single frequency tone, repeated twice in terms of 300 milliseconds on and 10 seconds off.
- 950/333,1400/333,1800/333,0/1000

  Indicates repeated playing 333 milliseconds of 950 Hz, 333 milliseconds of 1400 Hz, 333 milliseconds of 1800 Hz, and mute of 1 second

2.6.9 Functional keys

The function key consists of system function key and service function key. The system function key is used for acquiring gateway information, and the later is used for users to activate and inactivate supplementary services.

After login, click the label of "Advanced > Functional Keys" to open this interface.

The following are the examples of the dialing rule for the function key:

a) Using *xx (dial * and 2 digits number) to activate a service;

b) Using #xx (dial # and 2 digits number) to cancel a service.

Illustrate with following defaults of various parameters, which may be modified according to requirements.

Figure 2-23 Functional keys configuration interface

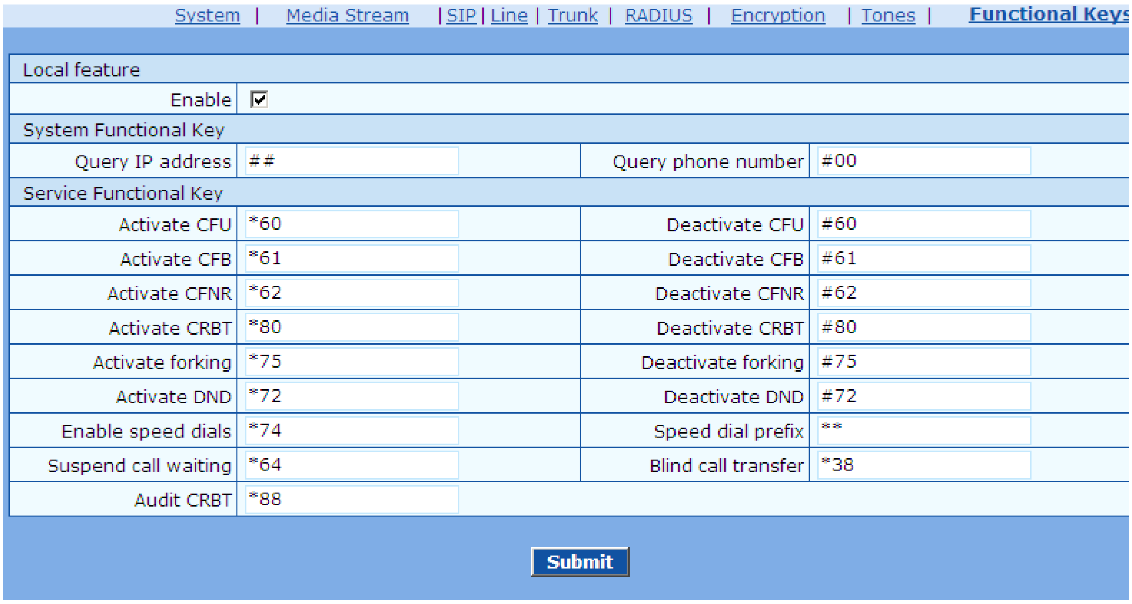


Table 2-24 Functional keys configuration parameter

Title	Explanation
Signaling functional keys	
Activate	Activate: subscriber line number matches with functional keys listed on this page, applied to local equipment Non: all dialed functional keys are sent to proxy server.
System Functional Key	
Query IP address	The function key for inquiring the IP address of gateway, with default of ##. Dialing this key, users can hear gateway broadcasting IP address and system software version number. Narrative: if the gateway is only equipped with FXO port, connect FXO port through PBX extension line or PSTN direct line, and dial the number of this line accordingly, press "##" immediately after hearing the second dial tone, users may thus hear IP address and system software version number of the gateway.
Query phone number	The function key for inquiring the phone number of this subscriber line, with default of #00. Dialing this key may hear the phone number of the subscriber line broadcasted by gateway.
Service Functional Key	
Activate CFU	The function key for activating unconditional call forwarding, with default of *60. Dialing this key may activate unconditional call forward of the line, and set the destination number for call forwarding. User operation: Off hook → press *60 →enter the destination number. Users can inquire the latest destination number set by dialing " *60* ". Note: it's required to enable call forwarding service before using this function (please see the instructions on relevant configuration of "subscriber line").
Deactivate CFU	The function key for deactivating unconditional call forwarding, with default of #60. User operation: Off hook → press #60 → hang up.
Activate CFB	The function key for activating call forwarding on busy， with default of *61. Dialing this key may activate CFB, and specify the destination number. Note: it's required to enable call forwarding on busy service before using this function (please see the instructions on relevant configuration of "subscriber line").
Deactivate CFB	The function key for deactivating call forwarding on busy, with default of #61. User operation: Off hook → press #61 → hang up.
Activate CFNR	The function key for activating call forwarding on no answer, with default of *62. Dialing the function key may activate call forwarding on no answer and specify destination number. Note: it's required to enable call forwarding on no answer ervice before using this function (please see the instructions on relevant configuration of "subscriber line").
Deactivate CFNR	The function key for deactivating call forwarding on no answer, with default of #62.

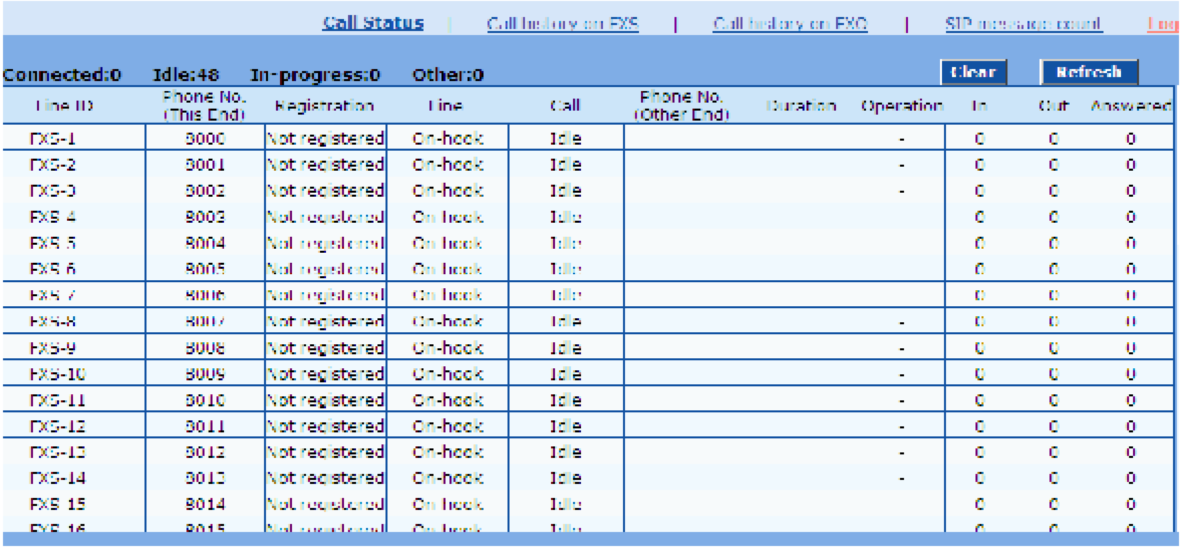
Title	Explanation
Activate CRBT	The function key for activating color ring, with default of *80. The subscribers may select their favorite color rings by using the key. Note: it's required to start color ring service before using this function (please see the instructions on relevant configuration of "subscriber line"). User operation: Upon off hook, the subscriber may press the function key (like *80), then, input two digit index numbers of color ring; "*80* " is used for hearing and inquiring the color ring that have been set already.
Deactivate CRBT	The function key for deactivating the color ring, with default of #80. The subscriber may use such key to recover the normal ring of phone. User operation: Off hook → press #80 → hang up.
Activate forking	The function key for activating double ringing feature, with default of *75.
Deactivate forking	The function key for deactivating the feature, with default of #75.
Activate DND	Activating "Don't Disturb Service", with default of *72. After dialing up, the gateway will reject all coming calls by sending busy tone to the caller. Note: it's required to start "Don't Disturb Service" before using this function (please see the instructions on relevant configuration of "subscriber line").
Deactivate DND	The function key to cancel "Don't Disturb Service", with default of #72. Dialing the function key may recover normal ringing upon the arrival of incoming calls.
Enable speed dials	Define the function key of dial, with default of *74. Dialing of this function key may build a table of 2 digits（20~49）of abbreviated numbers, which corresponding to the real numbers. Note: It's necessary to get the dial-up service under way before applying this function (please refers to the instructions about "subscriber line". User operation: Upon dialing the function key (such as " *74 ") set hereof, the subscriber may save the corresponding relationship into gateway following dialing 2 digits of abbreviated number and corresponding number with # as ending..
Speed dial prefix	The prefix number for applying abbreviated dialing, with default of "**". The said prefix should be added ahead of abbreviated dialing numbers when using abbreviated dialing. User operation: off hook → dial the prefix number of abbreviated dialing（**）and dial abbreviated dialing number（20）。
Audit CRBT	The function key for hearing the color ring， with default of *88. User operation: Off hook → press *88 → input color ring number.
Blind call transfer	Function key of blind call transfer， with default of *38. User operation: During the call, tap the phone hook switch or press R butto n→ dial *38→ dial the called number and then hang up.
Suspend call waiting	The function key for cancelling the call waiting for next call, with default of *64. Dialing this function key may temporarily shield the call waiting for next call, avoiding the possible intervention. Note: the function key works only for single cancel, if to cancel the call waiting completely, please refer to the instructions on relevant configuration of "subscriber line".

2.7 Call status and statistics

2.7.1 Call status

After login, click the label of “Status > Call Status” to open this interface.

Figure 2-24 Interface of call status



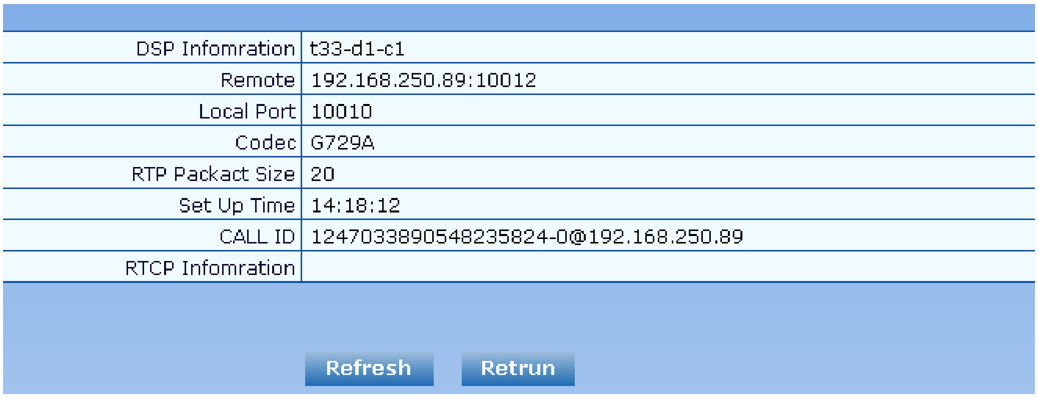
Line ID	Phone No. (This End)	Registration	Line	Call	Phone No. (Other End)	Duration	Operation	In	Out	Answered
FXS-1	8000	Not registered	On-hook	Idle			-	0	0	0
FXS-2	8001	Not registered	On-hook	Idle			-	0	0	0
FXS-3	8002	Not registered	On-hook	Idle			-	0	0	0
FXS-4	8003	Not registered	On-hook	Idle				0	0	0
FXS-5	8004	Not registered	On-hook	Idle				0	0	0
FXS-6	8005	Not registered	On-hook	Idle				0	0	0
FXS-7	8006	Not registered	On-hook	Idle				0	0	0
FXS-8	8007	Not registered	On-hook	Idle			-	0	0	0
FXS-9	8008	Not registered	On-hook	Idle			-	0	0	0
FXS-10	8009	Not registered	On-hook	Idle			-	0	0	0
FXS-11	8010	Not registered	On-hook	Idle			-	0	0	0
FXS-12	8011	Not registered	On-hook	Idle			-	0	0	0
FXS-13	8012	Not registered	On-hook	Idle			-	0	0	0
FXS-14	8013	Not registered	On-hook	Idle			-	0	0	0
FXS-15	8014	Not registered	On-hook	Idle				0	0	0
FXS-16	8015	Not registered	On-hook	Idle				0	0	0

Table 2-25 Parameters of call state

Title	Explanation
Line	There are six types of line statuses, including On-hook, Off-hook, Ringing, Maintenance, Disconnect, Parallel line in-use.
Call	The call state includes Idle, Ooutpusling, Ring, Entering number, In progress, Ring back, Talk, Near end hung up, Far end hung up, Timeout.

Click the label of “check detail” to open detail interface.

Figure 2-25 Details for the call



DSP Infomration	t33-d1-c1
Remote	192.168.250.89:10012
Local Port	10010
Codec	G729A
RTP Packact Size	20
Set Up Time	14:18:12
CALL ID	1247033890548235824-0@192.168.250.89
RTCP Infomration	

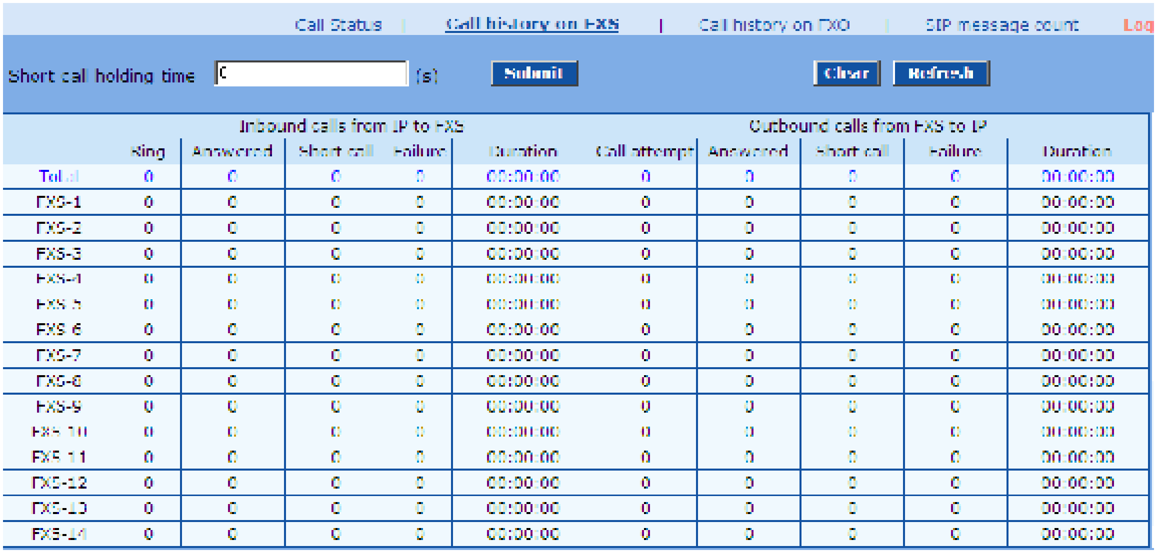
Table 2-26 Details for the call

Title	Explanation
DSP Information	This indicates the DSP chip information used for the call, in which “t” indicates time slot, “d” indicates the DSP chip, “c” refers to the channel on the chip.
Remote	The IP address of the equipment at the far end, followed with RTP port number.
Local port	Local RTP port number of the call.
Codec	The codec for this call.
RTP Packet Size （millisecond）	Packet length of the RTP of the call.
Set Up time	The time at which the call is answered.
CALL ID	Call ID in SIP message.
RTCP Information	The latest RTCP statistics report received by this call.

2.7.2 Call history on FXS

After login, click the label of “status > Call history on FXS” to open this interface.

Figure 2-26 Interface of call on FXS

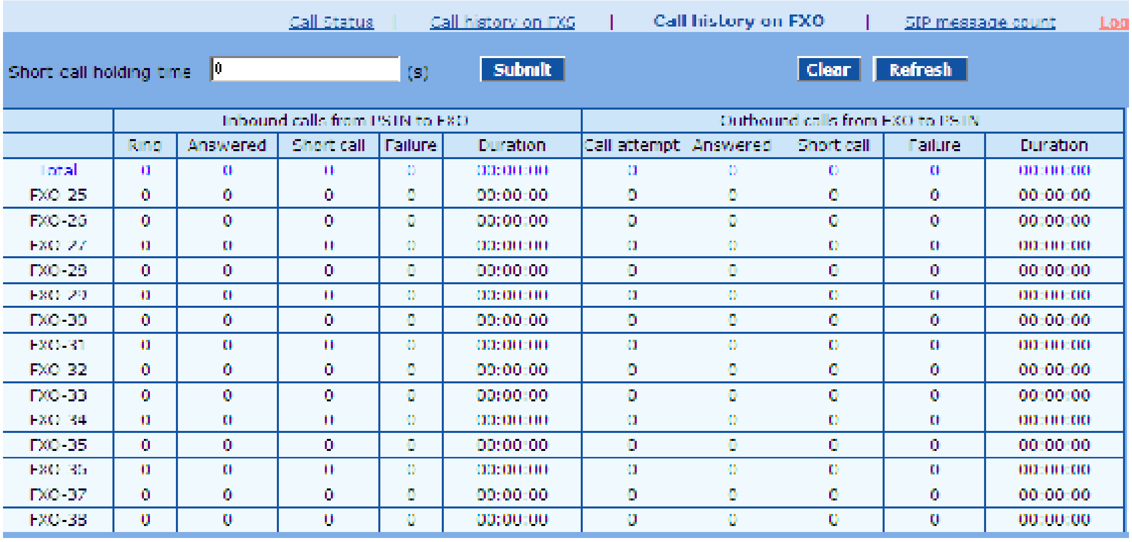


	Inbound calls from IP to FXS					Outbound calls from FXS to IP				
	Ring	Answered	Short call	Failure	Duration	Call attempt	Answered	Short call	Failure	Duration
Total	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-1	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-2	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-3	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-4	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-5	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-6	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-7	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-8	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-9	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-10	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-11	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-12	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-13	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-14	0	0	0	0	00:00:00	0	0	0	0	00:00:00

2.7.3 Call history on FXO

After login, click the label of “status > Call history on FXO” to open this interface.

Figure 2-27 Interface of call on FXO

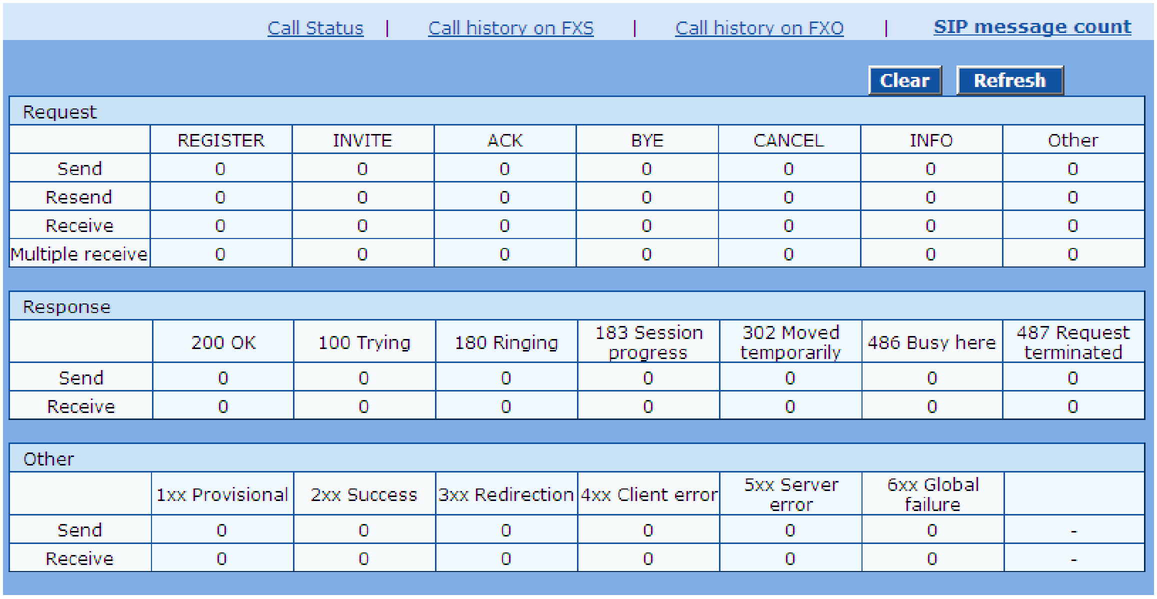


	Inbound calls from PSTN to FXO					Outbound calls from FXO to PSTN				
	Ring	Answered	Short call	Failure	Duration	Call attempt	Answered	Short call	Failure	Duration
Total	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-25	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-26	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-27	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-28	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-29	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-30	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-31	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-32	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-33	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-34	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-35	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-36	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-37	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-38	0	0	0	0	00:00:00	0	0	0	0	00:00:00

2.7.4 SIP message count

After login, click "status > SIP message count" to open this interface.

Figure 2-28 Figure 2-29 interface of SIP message count



Request	REGISTER	INVITE	ACK	BYE	CANCEL	INFO	Other
Send	0	0	0	0	0	0	0
Resend	0	0	0	0	0	0	0
Receive	0	0	0	0	0	0	0
Multiple receive	0	0	0	0	0	0	0

Response	200 OK	100 Trying	180 Ringing	183 Session progress	302 Moved temporarily	486 Busy here	487 Request terminated
Send	0	0	0	0	0	0	0
Receive	0	0	0	0	0	0	0

Other	1xx Provisional	2xx Success	3xx Redirection	4xx Client error	5xx Server error	6xx Global failure	
Send	0	0	0	0	0	0	-
Receive	0	0	0	0	0	0	-

2.8 Log management

2.8.1 System status

Critical runtime information of gateways can be obtained in this interface, including:

1) The information about login interface (including IP address and jurisdiction of the user);
2) SIP registration status;
3) Call related signaling and media (RTP) information;

After login, click the label of “Logs > System Status” to open this interface.

Table 2-27 Parameters of system status

Title	Explanation

Login User Info	Show the IP address and jurisdiction of login user. The numbers following the IP address show the online jurisdiction of the user: 1- administrator; 2 - operator; 3 – viewer. The viewer can only read the configuration, but is not allowed to modify it.
	When more than one administrator log in at the same time, the first login’s jurisdiction is 1, others are 3; also, when more than one operators log in at the same time, the first one’s jurisdiction is 2, others are 3.
	For example:
	`Login User Info >>>>>`
	`1） 192.168.2.247  1`

SIP Registration Info	Show registration status:
	ı Not enabled: The registeration server’s address is not entered yet;
	ı Latest response: The latest response message for the registration. 200 means registered successfully;
	ı No response: No response from registeration server. The cause may contribute to 1）incorrect address for the registration server; 2) IP network fault; or, 3) the registration server is not reachable.
	For example:
	`SIP Registration Info >>>>>`
	`---- Not enabled ----`
	`SIP Registration Info >>>>>`
	`Contact: <sip:2681403@220.218.77.70:1003; user=phone>`
	`latest response: 200 （timeout-555）`
	`Contact: <sip:2681402@220.218.77.70:1003; user=phone>`
	`latest response: 200 （timeout-555）`

Call Context Info	Show the call status.

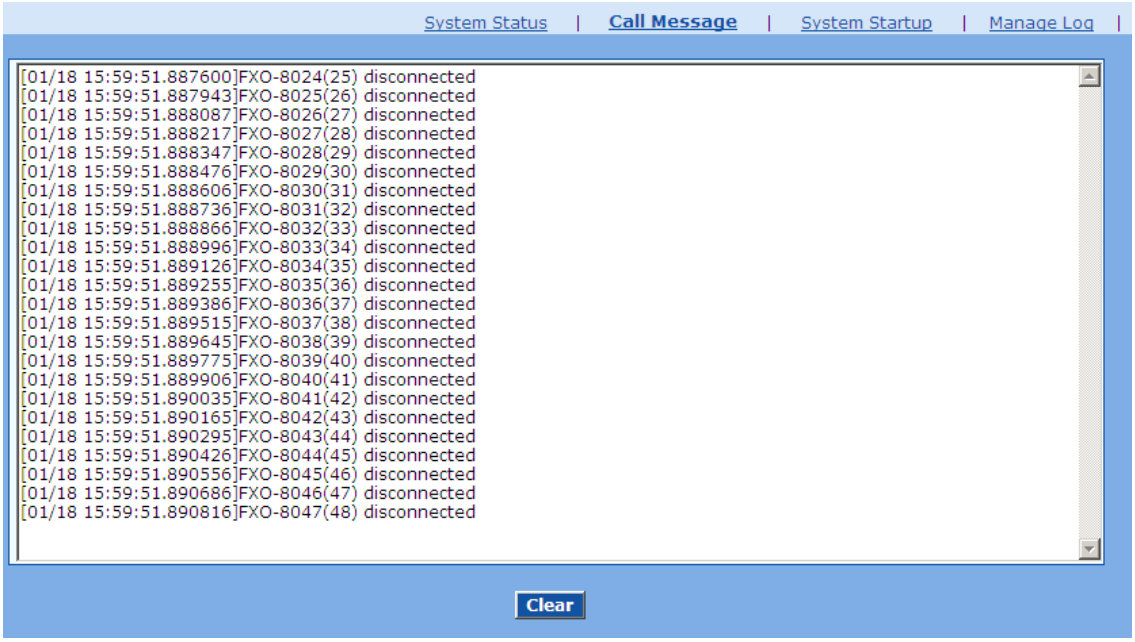
Title	Explanation

Rtp Context Info	Show the voice channel related to the calls.
	For example:
	`Rtp  Context Info >>>>>`
	`3） created， call =e011`

2.8.2 Call message

After login, click the label of “Logs > Call Message” to open this interface.

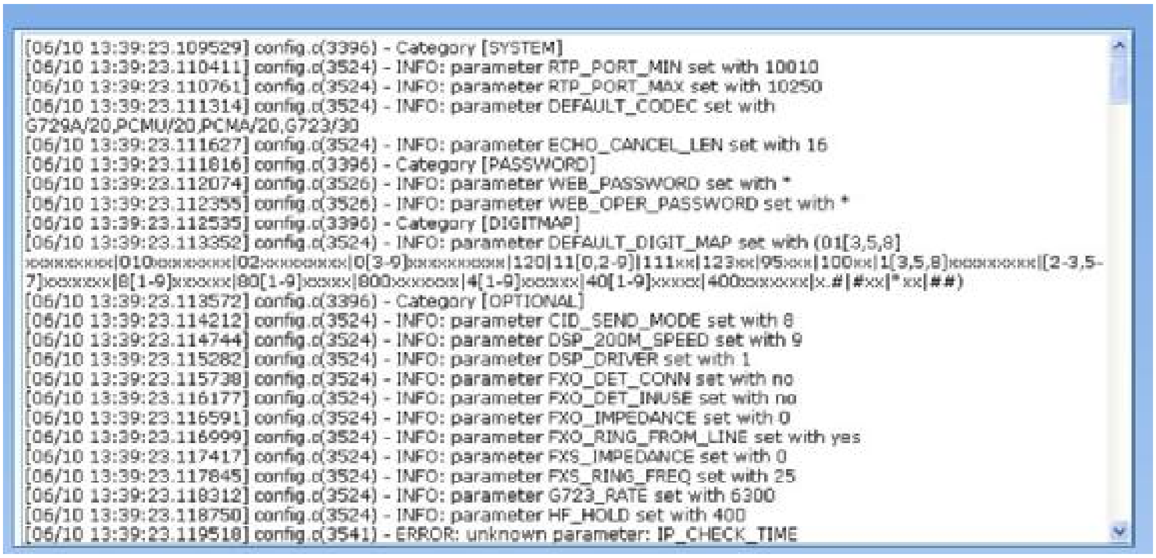
Figure 2-29 Call message interface



2.8.3 System Startup

After login, click the label of “Logs > System Startup” to open this interface. The gateway boot up information is available in this page, including the hardware configuration.

Figure 2-30 Interface of system startup



2.8.4 Manage log

After login, click the label of "Logs > Manage Log" to open this interface. Log files can be downloaded through this interface.

Figure 2-31 Interface of debugging log management

Table 2-28 Configuration parameters of debugging log management

Title	Explanation
Log download	See the description below.
System log server	Set the IP address of system log server.
Log server	IP address of debugging log server.
Log level	Select the log file level of gateway, default is 3. The setting range is 1 ~ 5, the higher the level goes, the more details the log file will be. Note: log level should be set to be 3 or lower when gateway is used in normal operation, avoiding influencing the system performance.

Procedure of downloading the debugging log:

Step 1: Click "download", the gateway starts pack the logs.

Step 2: After few seconds, the interface of log saving will appear.

Step 3: click "Save", and select path to save.

Step 4: The user may review the log from the server concerned.

WARNING

The procedure of downloading log files described hereof is only applicable to release 1.9.x.238 of WSS series or updated version of software.

2.9 System tool

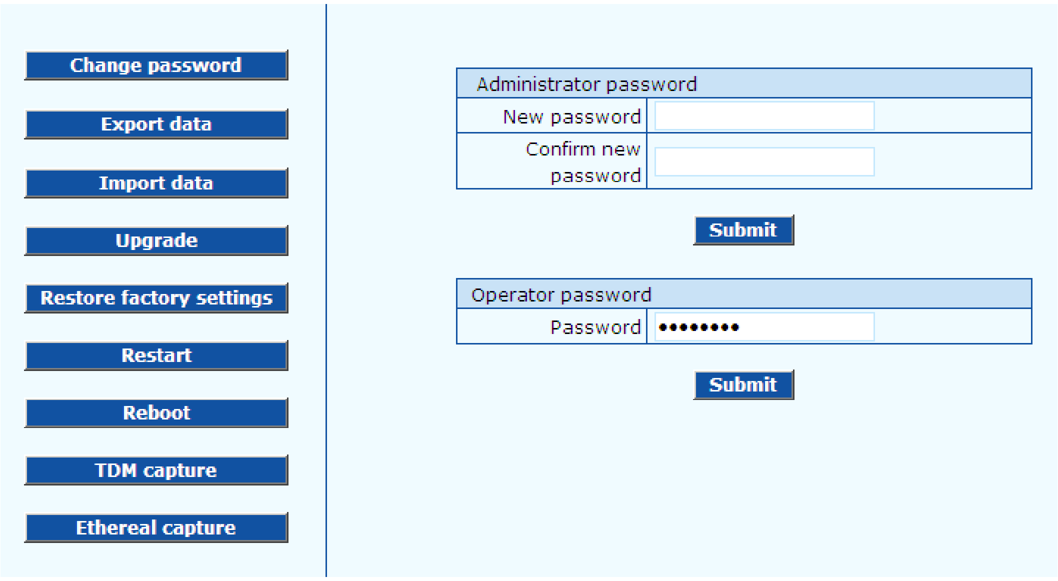
2.9.1 Change password

After login, click the label of “Tools” to open this interface. Only administrator is entitled to change the password of login.

For changing administrator password, it’s required to enter new password into “New password” field and “Confirm new password” field, then click “Submit”.

The password being used by operator will be displayed as hidden codes, which could be changed by administrator at any time. The administrator is allowed to change the operator’s password by entering new password into “Operator password>password”.

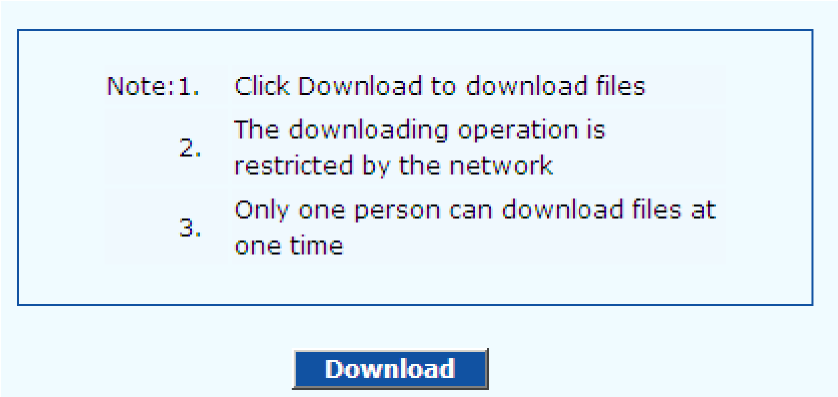
Figure 2-32 Interface of password changing



2.9.2 Configuration export

After login, click “Tools >Export of configuration” to open this interface. The downloading procedure is similar to the downloading procedure of log files..

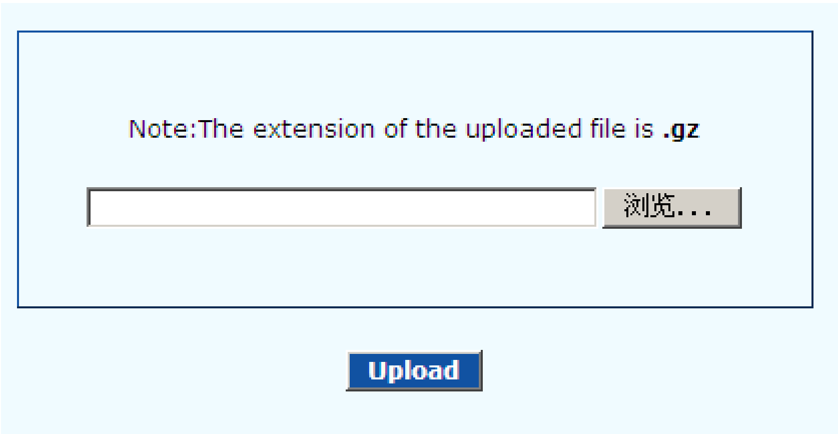
Figure 2-33 Interface of export data



2.9.3 Configuration import

After login, click “Tools>Import data” to open this interface. Operating procedure is the same as that of “software upgrade”.

Figure 2-34 Interface of import data



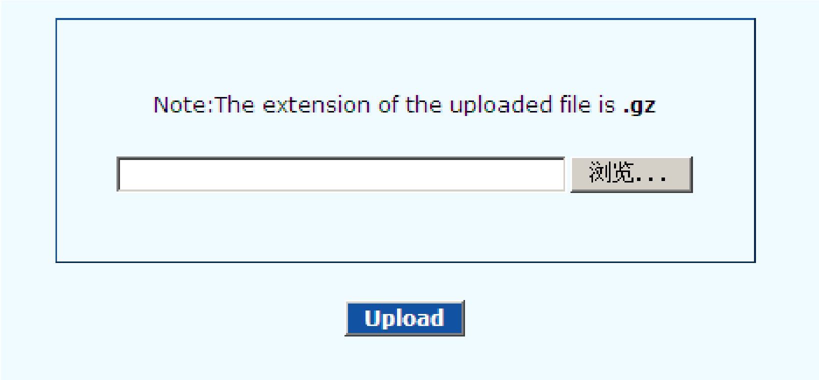
2.9.4 Software upgrade

After login, click “Tools > Upgrade” to open this interface. The software upgrading procedure is presented as below:

Step 1: Obtain the upgrade files (tar.gz file), and save the file onto a local computer.

Step 2: Click “System tool > software upgrade” to access to the page of software upgrade.

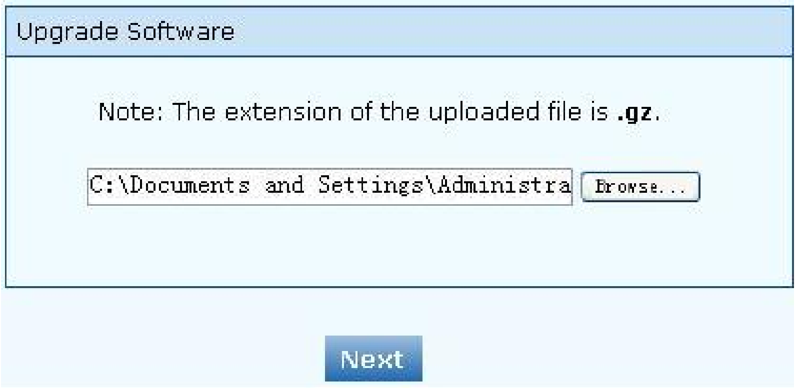
Figure 2-35 Interface of software upgrade



Step 3: Click “Browse” to select the upgrade files and click “Open”.

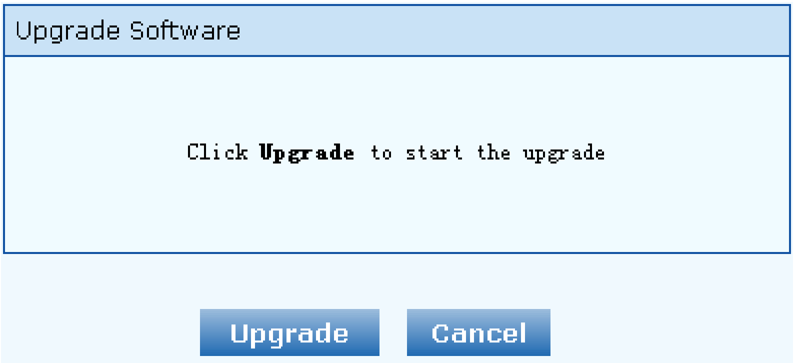
Step 4: Click “Next” when the following interface appears, and start uploading the upgrade files to the gateway.

Figure 2-36 Interface of file upload



Step 5: Uploading will be completed in about 30 seconds, and click “Upgrade” on following dialog.

Figure 2-37 Upgrade interface



Step 6: The following prompt appears during the upgrade.

Figure 2-38 Prompt of upgrade process

WARNING
A few minutes are needed to upgrade the gateway. Don’t operate the gateway during this period.

Step 7: After success in upgrade, the following dialog will appear, click “Confirm”.

Figure 2-39 Interface of successful upgrade

Step 8: The gateway is on the progress of reboot when the interface cannot be displayed.

Step 9: Wait for about 2 minutes, and access to the interface of gateway management system, click “Info” and check the software version.

WARNING
For WSS100 and WSS120 gateways, the software upgrade operation must be conducted on an 100M Ethernet port.

2.9.5 Restore factory settings

After login, click “Tools > Restore factory settings” to restore the parameters of gateway into the factory settings.

The factory settings are designed based on common applications, and therefore, no need to modify them in many deployment situations.

2.9.6 Software restart

After login, click “Tools > Restart” to restart the gateway, making modified configuration come into effect.

CAUTION
In most cases, ‘there is no need to reset the gateway, and the modified parameters will come into effect upon confirming the “submit”.

2.9.7 System reboot

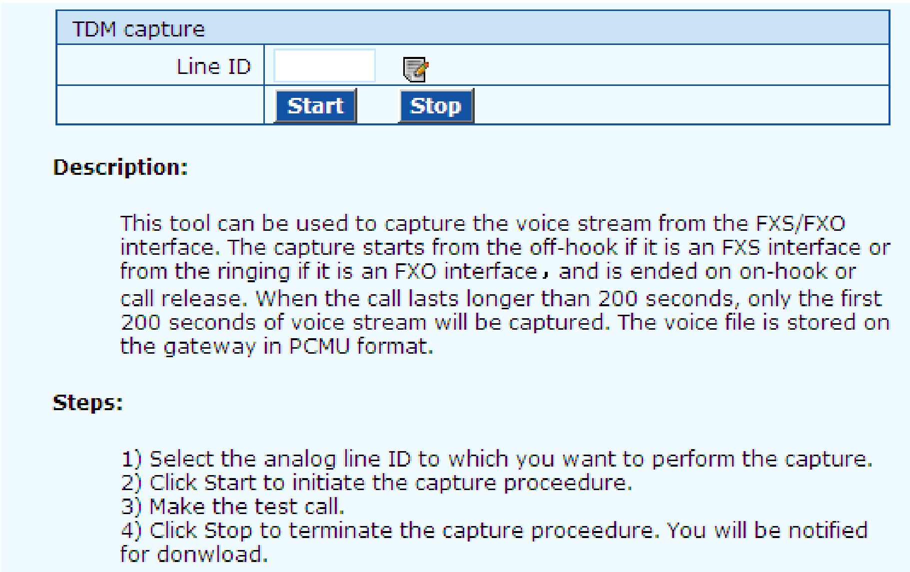
After login, click “Tools >Reboot” to restart the gateway. As this is a system wide reset, it takes longer time.

CAUTION

Generally, it’s sufficient to restart software when the gateway confirms to reset; the system reboot will be required only when network settings of the gateway are changed.

2.9.8 TDM Capture

After login, click “Tools > TDM Capture” to open this interface. This tool can be used to capture the voice stream from the FXS/FXO interface. The capture starts from the off-hook if it is an FXS interface or from the ringing if it is an FXO interface， and is ended on on-hook or call release. When the call lasts longer than 200 seconds, only the first 200 seconds of voice stream will be captured. The voice file is stored on the gateway in PCMU format.



Steps:

1) Select the analog line ID to which you want to perform the capture.

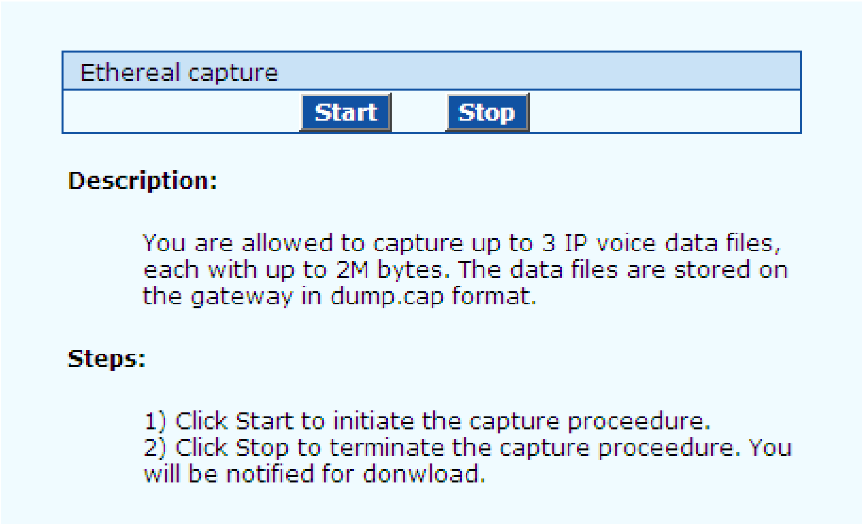
2) Click Start to initiate the capture proceedure.

3) Make the test call.

4) Click Stop to terminate the capture proceedure. You will be notified for donwload.

2.9.9 Ethereal Capture

After login, click “Tools > Ethereal Capture” to open this interface. You are allowed to capture up to 3 IP voice data files, each with up to 2M bytes. The data files are stored on the gateway in dump.cap format under catalog “/var/log”.



Steps:

1) Click Start to initiate the capture proceedure.

2) Click Stop to terminate the capture proceedure. You will be notified for donwload.

2.10 Version information

After login, click "Info" to view the gateway hardware and software version information.

Software version	Rev 1.9.82.303
Hardware version	Rev 1.0.1 MI120-24S/24-C
Kernel version	Kernel 1.1.8 (F)
DSP version	Rev 1.8.195

2.11 Logout

After login, click the "Logout" at top right to exit the gateway management system and return to the login interface.

3 Appendix

3.1 WSS120 system operation state

Table 3-1 WSS120 system operation state

Glittery letter	Status meaning
"C"	The IP address of gateway conflicts with that of other equipment in LAN. Please settle this problem before the gateway can be operated normally.
"D"	Internal failures have been entountered during gateway start up procedure. Please contact your local distributor for further diagnosis.
"P"	The gateway is in progress of system software upgrade. Please guarantee stable power supply and do not conduct other operations during this period.
"T"	The application software of gateway has been exited. If it can not be restored by rebooting the system, please contact your local distributor for further diagnosis.

If you have any other problems please send mail to support@realtonetech.com

Thanks!

WSS Analog Voice Gateway Series

User Configuration Guide

Amendent Records

Document Rev.2.1 (Jan 12, 2011)

Chapter 1

The configuration of WSS8-2S/2O and WSS8-6S/2O is added.

Document Rev.2.0 (Sep 30, 2010)

Chapter 1

1.3.3 Description of WSS60

Chapter 2

Modifications were made to the terms accuracy of description, etc. in the document. GUI version: 1.9.81.300.9

The following interfaces were added: fax; batch configuration of subscriber/trunk lines; call history on FXS/FXO; SIP message count; TDM Capture; Ethereal Capture.

Document Rev. 1.2 (Aug 31, 2010)

2.4.2 IP Table: modification of Example 3

2.6.9 Functional keys, datasheet of VoIP Gateway

Document Rev. 1.1 (May 10, 2009)

Modifications were made to the terms, accuracy of description, etc. in the document.

Chapter 1

The equipment structure of WSS120 was added.

Chapter 2

The content was modified according to the release of new Web interface.

Document Rev. 01 (June 16, 2005)

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1 Overview

1.1 Product Introduction

WSS Series intelligent VoIP Gateways (hereinafter called “WSS Gateways” or simply “Gateways”) are designed for bridging the traditional telecom terminal devics into IP networks through SIP or MGCP protocols. The main applications include:

- For carriers and value-added service providers to provide telephone, fax and voice-band data services to subscribers using IP access methods such as FTTB, HFC, and ADSL;
- Used to bridge the traditional telecom terminal equipments, such as PBXs, to the IP core networks of carriers;
- Connected with PBX of enterprises to provide IP-based voice private network solutions for institutions, enterprises and schools;
- Used as remote acces equipments for IP-PBXs in call center deployment.

WSS Gateways are suitable for placement on office desktops or installation on walls in the corridor and racks in the equipment room.

WSS Series includes WSS8, WSS60 and WSS120 subseries. Their features are similar with the main differences as follows:

Table 1-1 Differences Between WSS Gateway Series

	Capacity	Chassis	Subscriber Line Board Card	Installat ion	Power
WSS8	2-8 FXS/FXO Ports	Plastic Casing	Built-in	Desktop	5-9 VDC
WSS60	16-48 FXS/FXO Ports	19’’ wide and 1U High	Built-in	Rack	100-240 VAC
WSS120	24-48 FXS/FXO Ports	19” Wide and 1U High	Pluggable	Rack	100-240 VAC, -48 VDC (Optional)
WSS120	48-96 FXS/FXO Ports	19” Wide and 2U High	Pluggable	Rack	100-240 VAC, -48 VDC (Optional)

WSS Gateways use Freescale® PowerQUICC communications processors as main control processors (including 50MHz MPC852T, 200MHz MPC8250 and 300MHz MPC8247) and TI’s TMS320VC5509A high-performance digital signal processing chips as processors for voice and fax processing (equipped with 1-12 DSP chips based on the need of concurrent call capacity), and are integrated with 32MB-64MB SDRAM as system memory, 4MB-16MB FLASH as permanent file system. The powerful processing capability and sufficient hardware configuration ensure that all products of WSS Series can provide concurrent calls of full capacity and maintain good call quality.

All WSS Gateways run on stable and reliable embedded Linux operating system. On top of Linux OS, the driver layer handles hardware specific control in different product platforms. This makes single source application software running cross the full range of WSS product series, and ensures the consisten functions and stable performance in different WSS product lines.

WSS Gateways support SIP and MCGP protocols. They can provide

- PBX functions such as hunting group, second stage dialing, internal communications, caller ID (FSK/DTMF), call transfer, call waiting, call hold, call barring, caller ID restriction, hotline, corporate CRBT, three-way calling, ring group, fax and etc;
- FXO related functions such as PSTN failover, gain control, busy tone detection, voice prompt in inbound calls, polarity reversal detection;
- Media stream processing functions such as RTP redundancy, packet loss compensation, G.711/G.729A/G.723.1/iLBC/GSM voice codec, echo cancellation, and etc.

WSS Gateways support local and remote, distributed and centralized management modes, including Web access management，command line configuration based on Linux OS, auto-provision for firmware upgrade and configuration management based on TFTP/FTP/HTTP, SNMPv2, TR069 based ACS.

1.2 Functions and Features

- Connect analog telephone, PBX, facsimile machine and POS machine to the IP core network, or PSTN;
- Work with service platform to provide various telephone supplementary services;
- Support protocols: SIP, MGCP;
- Flexible configuration of FXS/FXO interfaces;
- Support static IP address configuration or dynamically obtain an IP address through DHCP and PPPoE;
- Support G.711, G.729A, G.723.1, GSM, iLBC;
- Support echo cancellation;
- Up to 500 routing rules can be stored in gateways;
- Support digitmap;
- Support T.30/T.38 fax mode;
- Support multiple local and remote maintenance & management modes such as Web, Telnet, auto-provision, and TR069/TR104/TR106 clinet;.
- security strategy: IP filter, encryption
- Support call progress tones for various countries and regions;
- Support FXO second stage dialing or voice prompt;
- Support PSTN failover through FXO ports;
- Support High Capacity SD Card (optional, only for WSS60)
- Support polarity inverse detection and busy tone detection
- Support three-way calling
- Compatible with unified communication solutions, such as CallManager, OCS and Asterisk

1.3 Equipment Structure

1.3.1 WSS8

WSS8 is the product with smallest capacity in WSS Gateway Series. Designed with small plastic structure for desktop placement, WSS8 can provide up to 8 analog line interfaces. WSS8 supports the following types of configuration:

Table 1-2 Common Configuration Combination of WSS8

Models	Number of FXS Ports	Number of FXO Ports
WSS8-2S/2	2	2
WSS8-6S/2	6	2
WSS8-4S	4	0
WSS8-8S	8	0
WSS8-4FXO	0	4
WSS8-8FXO	0	8
WSS8-4S/4	4	4

Figure 1-1 WSS8 Front Panel

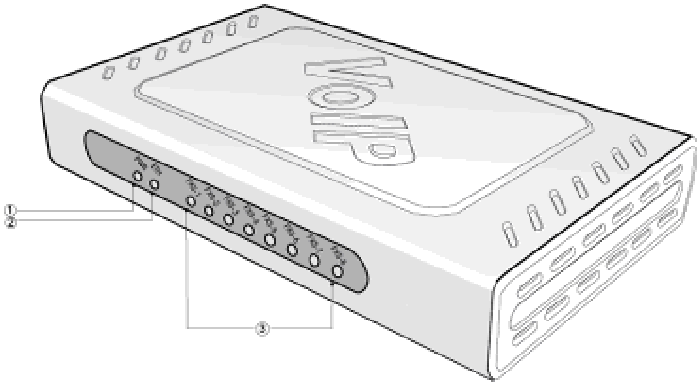


Table 1-3 Description of WSS8 Front Panel

#	Description
①	Power indicator (PWR), Light-on indicates that it has been powered.
②	Steady on indicates valid Ethernet link, flashing indicates Ethernet activities (receiving and/or transmitting)
③	Analog subscriber line (FXS) or analog trunk (FXO) interface indicator, Light-on indicates that it is in use.

Figure 1-2 WSS8 Back Panel

Table 1-4 Description of WSS8 Back Panel

#	Description
①	Power interface, 5-9 VDC input
②	10/100 M Ethernet Interface, RJ45
③	Analog subscriber line (FXS) or analog trunk (FXO) interface

Table 1-5 Configuration Description of Analog Line Interfaces for All WSS8 Models

WSS8 Models	RJ11 Interface Configuration							
	1	2	3	4	5	6	7	8
WSS8-2S/2	Trunk Line 1	Trunk Line 2	Subscriber Line 1	Subscriber Line 2	NA	NA	NA	NA
WSS8-6S/2	Trunk Line 1	Trunk Line 2	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Subscriber Line 5	Subscriber Line 6
WSS8-4S	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	NA	NA	NA	NA
WSS8-8S	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Subscriber Line 5	Subscriber Line 6	Subscriber Line 7	Subscriber Line 8
WSS8-4FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	NA	NA	NA	NA
WSS8-8FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	Trunk Line 5	Trunk Line 6	Trunk Line 7	Trunk Line 8
WSS8-4S/4	Subscriber Line 1	Subscriber Line 2	Subscriber Line 3	Subscriber Line 4	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4

1.3.2 WSS60

Designed with a 1U high and 19’’ wide compact chassis, WSS60 is suitable for installation in a standard cabinet. The interface card of WSS60 uses a RJ-45 socket and is connected to the distribution panel in equipment room using CAT-5 cables supplied with the unit. It has a built-in 110-220V power module. WSS60 offers up to 48 interfaces of FXS/FXO. WSS60 supports the following types of configuration.

Table 1-6 Configuration combination of WSS60

Models	Numbers of FXS Ports	Numbers of FXO Ports
WSS60-16S	16	0
WSS60-24S	24	0
WSS60-32S	32	0
WSS60-48S	48	0
WSS60-8S/8	8	8
WSS60-24S/8	24	8
WSS60-40S/8	40	8
WSS60-16S/16	16	16
WSS60-32S/16	32	16
WSS60-24S/24	24	24

Figure 1-3 WSS60 Front Panel

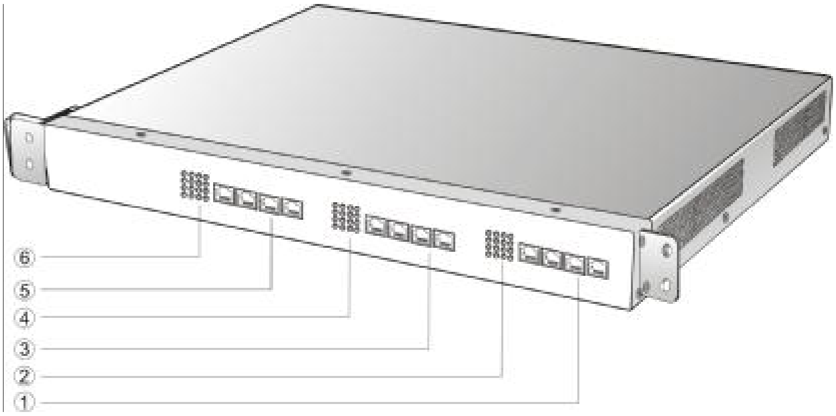


Table 1-7 Description of WSS60 Front Panel

#	Description
①③⑤	Three interface slots; each can correspond with four RJ45 sockets; each RJ45socket can correspond with four pairs of analog lines. Note: numbers of interface slots vary from different configuration.
②④⑥	Matrix of 4 x 4 LED status indicator on interface cardl

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation. Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-8 Pin Specifications for WSS60 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-4 Schematic Diagram of WSS60 Subscriber Line Connection

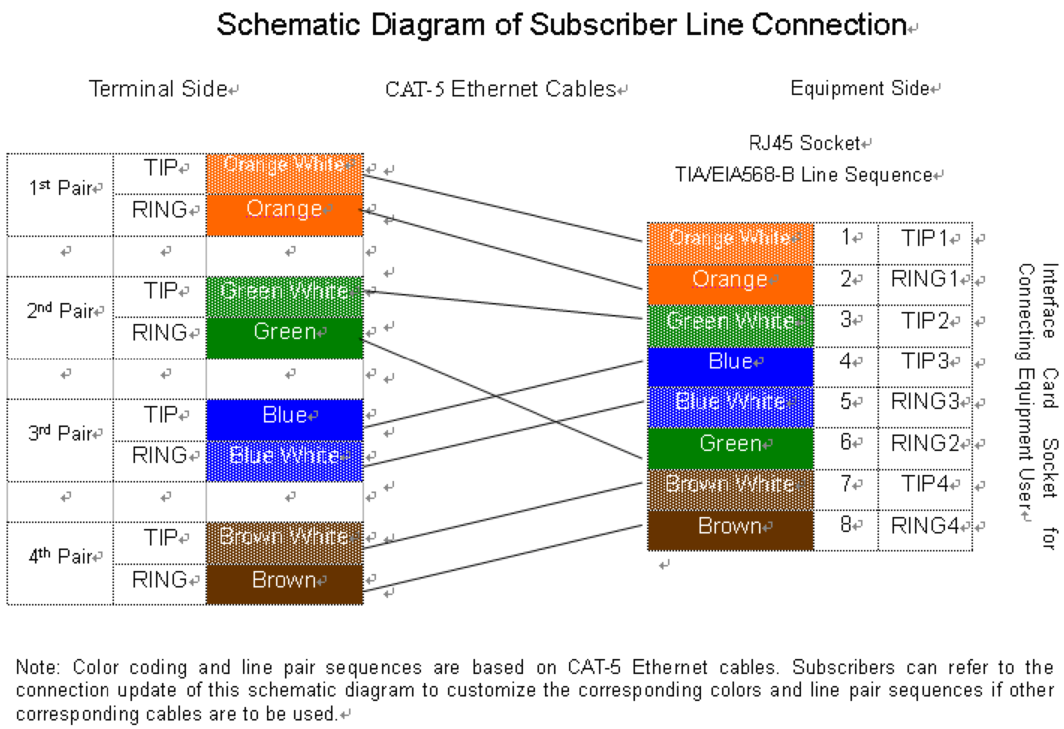


Figure 1-5 WSS60 Back Panel

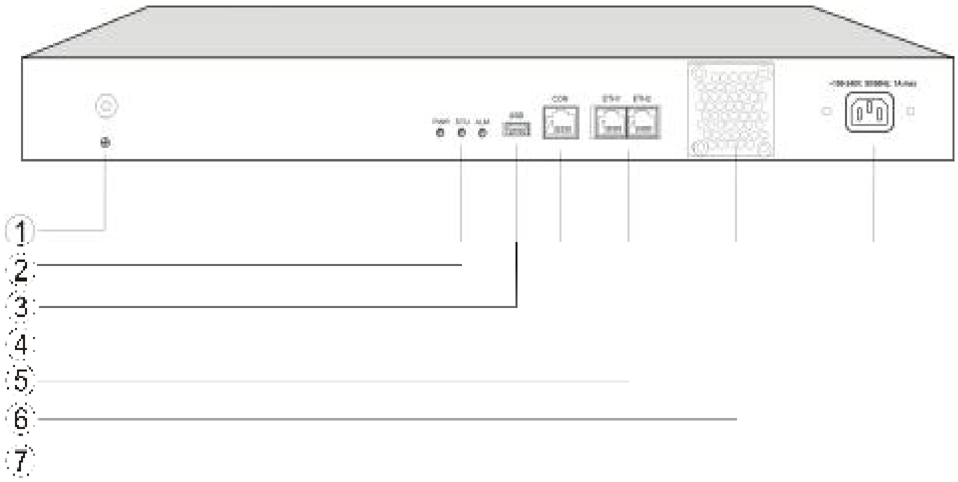


Table 1-9 Description of WSS60 Back Panel

#	Description
①	Ground Pole
②	Indicator, see Table 1-17 for description
③	USB Interface, reserved for future use
④	Configuration interface (CON), Ethnet lines used for local management and debugging

#	Description
⑤	Two Ethernet interfaces: one IP address
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input

Table 1-10 Meanings of WSS60 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Off	No alarms
		Red Flash	New alarms occurred but not confirmed.
		Red Constant	System in the process of powerup and not in the normal operation mode
		Red	Alarms existed and all alarm information confirmed.

1.3.3 WSS120 1U

Designed with 1U high and 19’’ wide compact chassis and a swappable modular structure, WSS120 can offer up to 48 analog lines. The interface card of WSS120 uses a RJ45 socket and is connected to the distribution panel in equipment room using CAT-5 cables supplied with the unit.

The device of WSS120 1U can hold two interface cards which enable to flexibly configure FXS and FXO ports. And each card equips up to 24 ports. It supports the following configurations:

Table 1-11 Configuration Combination of WSS120:

Models	Number of FXS Ports	Number of FXO Ports
WSS120-24FXO	0	24
WSS120-32FXO	0	32
WSS120-48FXO	0	48
WSS120-40S/8	40	8
WSS120-36S/12	36	12
WSS120-32S/16	32	16
WSS120-28S/20	28	20
WSS120-24S/24	24	24

Figure 1-6 WSS120 Front Panel

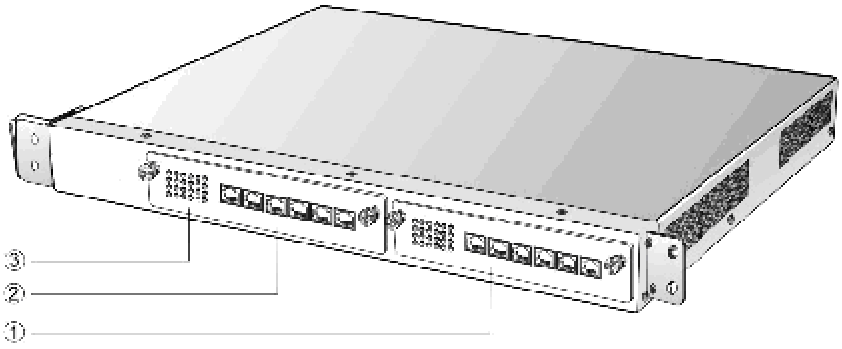


Table 1-12 Description of WSS120 Front Panel

#	Description
① and ②	Two interface slots; each can contain one 24-port interface card.
③	Matrix of 6×4 LED status indicator on interface card

WARNING

Do not plug and remove the interface cards of WSS120 when equipment is powered on.

Numbering definition of system interface slots: On the left side of main chassis is #1 slot (marked with No.1 to 24), on the right side of main chassis is #2 slot (marked with No.25 to 48).

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation. Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-13 Pin Specifications for WSS120 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-7 Schematic Diagram of WSS120 Subscriber Line Connection

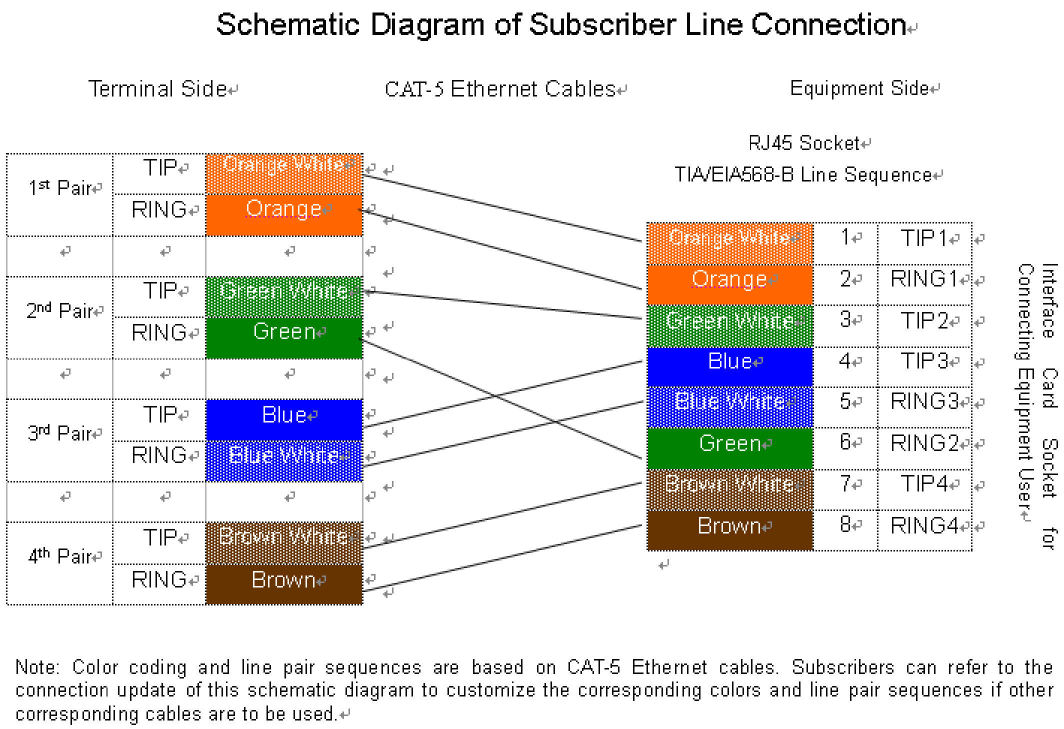


Table 1-14 Corresponding Relation Between WSS120 RJ45 Socket and Line Number

RJ45 Socket No. (From Left to Right)	1	2	3	4	5	6
Line No. of This Card	1 ~ 4	5 ~ 8	9 ~ 12	13 ~ 16	17 ~ 20	21 ~ 24

There is a 6 × 4 LED indicator matrixes on the left side of interface board. Each row of LED indicator matrixes matches four telephone lines on a RJ45. The first row on the left matches Line 1-4 respectively from top to bottom, the first row on the right matches Line 21-24 respectively from top to bottom, and the middle rows in the same manner.

LED indicators are used for multiple purposes as follows

- Line status indication: This is the most common mode during normal use of equipment. In this mode, if a line is idle, the indicator corresponding to it goes off; if a line is in call or in use status (such as ringing, offhook and caller ID transmission of FXS interface, ringing, offhook and caller ID detection of FXO interface) the indicator corresponding to it goes on.
- Line type indication: This is the mode for cable wiring check when installing the equipment. This mode can be entered by disconnecting Ethernet cables (Both WAN and LAN ports must be disconnected) at installation stage. After entering this mode, steady on LED indicates that the corresponding line is equipped as analog subscriber line type, flashing LED indicates that the corresponding line is equipped as analog foreign exchange line type, off LED indicates that the corresponding line is not equipped or not ready for use.
- System operation status indication: This is the mode for displaying information on system operation of equipment in specific conditions. Usually, this mode is entered when some prompts are required to give operator during equipment startup, diagnosis or operation. In this mode, LED flashes to display numbers, letters or other patterns in matrix. Please refer to the Appendix: Check List for Operation Status Indication of WSS120 System.

Figure 1-8 WSS120 Back Panel

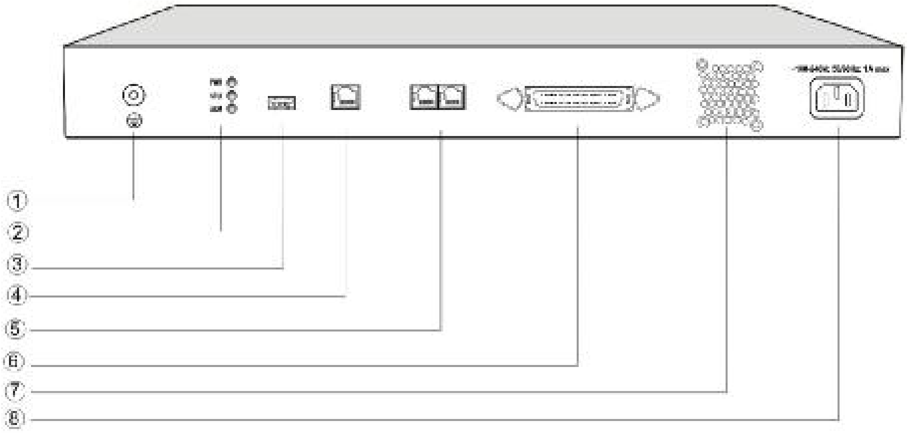


Table 1-15 WSS120 Back Panel

#	Description
①	Ground Pole
②	Indicator, see Table 1-16 for description.
③	USB interface, reserved for future use.
④	Configuration interface (CON), used for local management and debugging.
⑤	Two Ethernet interfaces: ETH1 and ETH2, only ETH1 has been set when the equipment is delivered from factory, default IP address: 192.168.2.240
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input.

Table 1-16 Meanings of WSS120 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Green	No alarms
		Red Flash	New alarms occurred but not confirmed
		Red	Alarms existed and all alarm information confirmed

1.3.4 WSS120 2U

The device of WSS120 2U can hold four interface cards which enable to flexibly configure FXS and FXO ports. And each card equips up to 24 prots. WSS120 2U can provide up to 96 ports. It supports the following configurations:

Table 1-17 Configuration Combination of WSS120 2U:

Models	Number of FXS Ports	Number of FXO Ports
WSS120-72S	72	0
WSS120-96S	96	0
WSS120-72FXO	0	72
WSS120-96FXO	0	96
WSS120-64S/8	64	8
WSS120-88S/8	88	8
WSS120-60S/12	60	12
WSS120-84S/12	84	12
WSS120-56S/16	56	16
WSS120-80S/16	80	16
WSS120-52S/20	52	20
WSS120-76S/20	76	20
WSS120-48S/24	48	24
WSS120-72S/24	72	24
WSS120-44S/28	44	28
WSS120-68S/28	68	28
WSS120-40S/32	40	32
WSS120-64S/32	64	32
WSS120-36S/36	36	36
WSS120-60S/36	60	36

Figure 1-9 WSS120 2U Front Panel

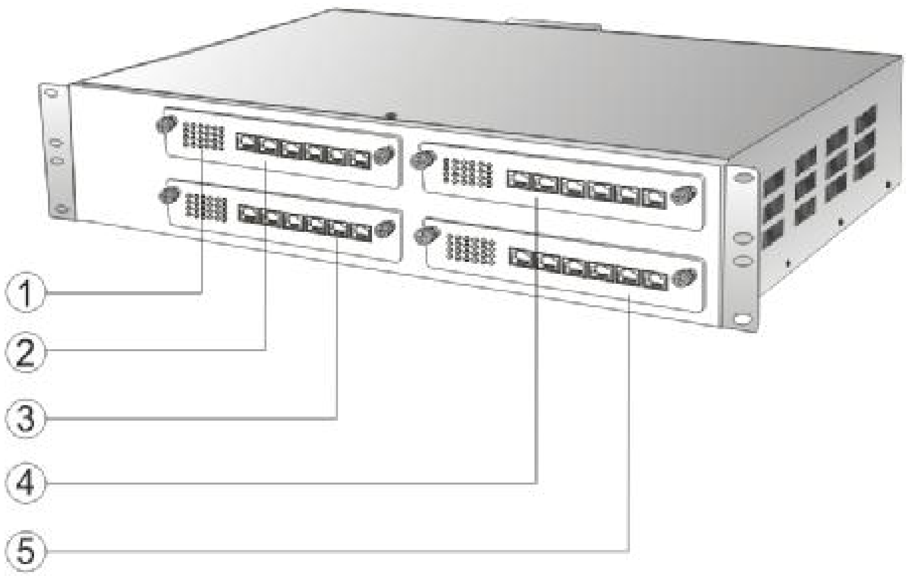


Table 1-18 Description of WSS120 Front Panel

#	Description
①	Matrix of 6×4 LED status indicator on interface card
②③④⑤	Four interface slots; each can contain one 24-port interface card.

WARNING

Do not plug and remove the interface cards of WSS120 when equipment is powered on.

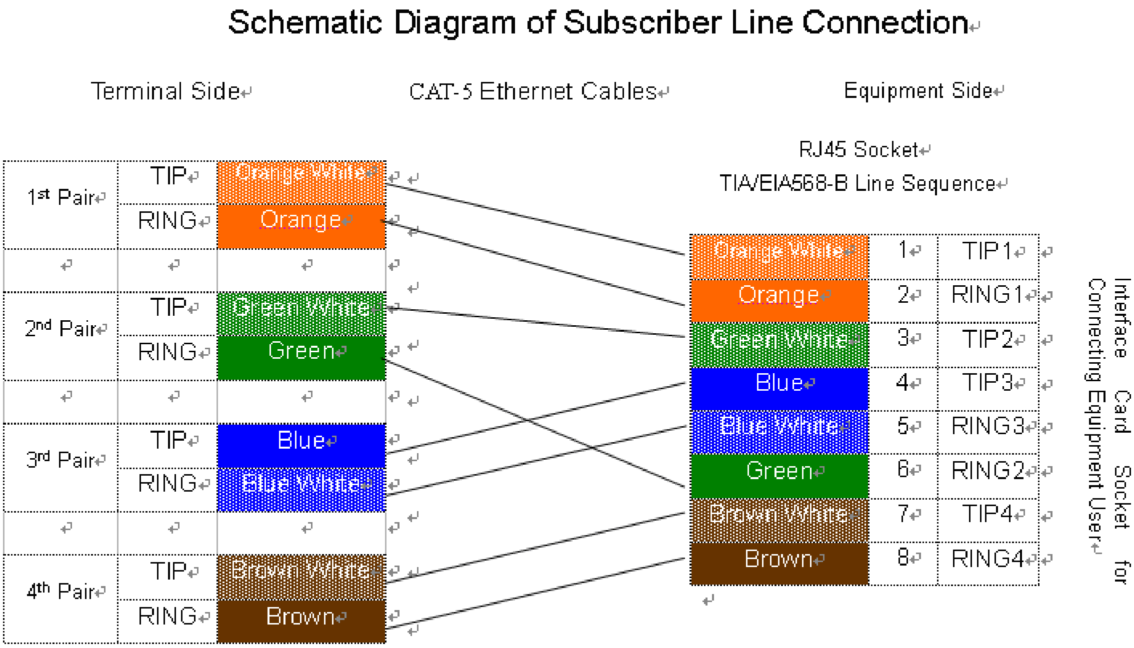
Numbering definition of system interface slots: On the low-left side of chassis is #1 slot (marked with No.1 to 24), on the low-right side of chassis is #2 slot (marked with No.25 to 48), on the up-left side of chassis is #3 slot (marked with No.49 to 72), and on the up-right side of chassis is #4 slot (marked with No.73 to 96).

Each RJ45 socket has 8 pins leading out 4 pairs of analog telephone or trunk lines in agreement with the pair specifications for Ethernet interfaces, whose corresponding relations can be seen in the table below. CAT-5 cables are used to connect the interface card and distribution panel in equipment installation. Standard RJ11 telephone lines can be used to plug in a RJ45 socket. The telephone/trunk lines are connected to the 3rd pair of pins for simple call test.

Table 1-19 Pin Specifications for WSS120 RJ45 Socket Port

RJ45 Pin Number	1	2	3	4	5	6	7	8
Analog line pair	1st Pair		2nd Pair	3rd Pair		2nd Pair	4th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	Orange white	Orange	Green white	Blue	Blue white	Green	Brown white	Brown

Figure 1-10 Schematic Diagram of WSS120 Subscriber Line Connection



Note: Color coding and line pair sequences are based on CAT-5 Ethernet cables. Subscribers can refer to the connection update of this schematic diagram to customize the corresponding colors and line pair sequences if other corresponding cables are to be used.

Table 1-20 Corresponding Relation Between WSS120 RJ45 Socket and Line Number

RJ45 Socket No. (From Left to Right)	1	2	3	4	5	6
Line No. of This Card	1 ~ 4	5 ~ 8	9 ~ 12	13 ~ 16	17 ~ 20	21 ~ 24

There is a 6 × 4 LED indicator matrixes on the left side of interface board. Each row of LED indicator matrixes matches four telephone lines on a RJ45. The first row on the left matches Line 1-4 respectively from top to bottom, the first row on the right matches Line 21-24 respectively from top to bottom, and the middle rows in the same manner.

LED indicators are used for multiple purposes as follows

- Line status indication: This is the most common mode during normal use of equipment. In this mode, if a line is idle, the indicator corresponding to it goes off; if a line is in call or in use status (such as ringing, offhook and caller ID transmission of FXS interface, ringing, offhook and caller ID detection of FXO interface) the indicator corresponding to it goes on.
- Line type indication: This is the mode for cable wiring check when installing the equipment. This mode can be entered by disconnecting Ethernet cables (Both WAN and LAN ports must be disconnected) at installation stage. After entering this mode, steady on LED indicates that the corresponding line is equipped as analog subscriber line type, flashing LED indicates that the corresponding line is equipped as analog foreign exchange line type, off LED indicates that the corresponding line is not equpped or not ready for use.
- System operation status indication: This is the mode for displaying information on system operation of equipment in specific conditions. Usually, this mode is entered when some prompts are required to give operator during equipment startup, diagnosis or operation. In this mode, LED flashes to display numbers, letters or other patterns in matrix. Please refer to the Appendix: Check List for Operation Status Indication of WSS120 System.

Figure 1-11 WSS120 2U Back Panel

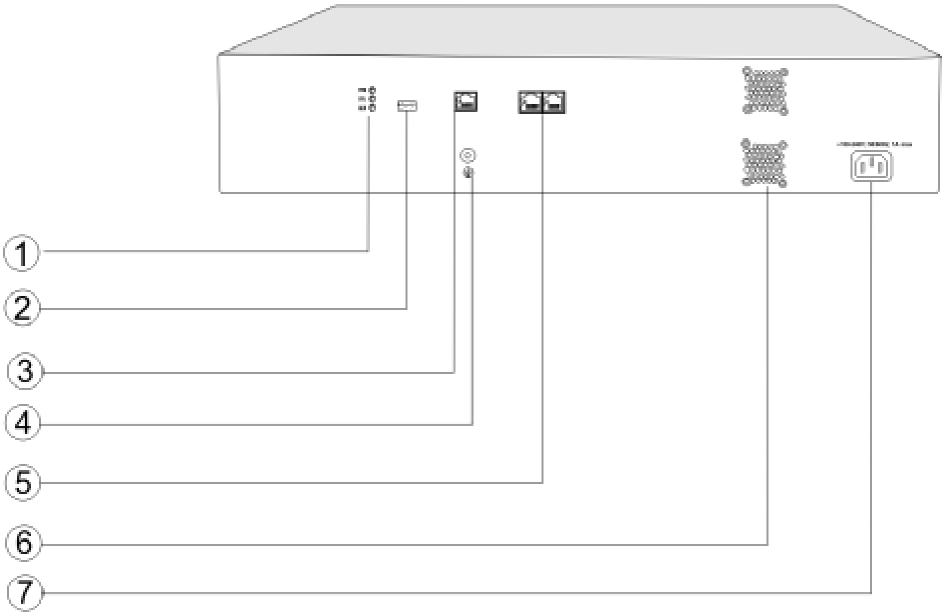


Table 1-21 WSS120 Back Panel

#	Description
①	Indicator, see Table 1-28 for description.
②	USB interface, reserved for future use.
③	Configuration interface (CON), used for local management and debugging.
④	Ground Pole
⑤	Two Ethernet interfaces: ETH1 and ETH2, only ETH1 has been set when the equipment is delivered from factory, default IP address: 192.168.2.240
⑥	Cooling fan
⑦	AC power socket, 100-240 VAC voltage input.

Table 1-22 Meanings of WSS120 Indicators

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STU	Status Indication	Off	System locked and inactive
		Green Flash	Normal operation
ALM	Alarm Indication	Green	No alarms
		Red Flash	New alarms occurred but not confirmed
		Red	Alarms existed and all alarm information confirmed

2 Parameter Setting

2.1 Login

2.1.1 Obtain Gateway IP Address

WSS8 Gateways start DHCP service by default, and automatically obtain an IP address on the LAN; users can use the factory default gateway IP address if it is unable to be obtained (e.g. when connected directly with a computer).

WSS60 and WSS120 Gateways use a static IP address by default.

Table 2-1 Default IP Address of Gateway

Type	Default DHCP Service	Default IP Address	Default Subnet Mask
WSS8	Enabled	192.168.2.218	255.255.0.0
WSS60	Disabled	192.168.2.240	255.255.0.0
WSS120	Diasabled	192.168.2.240	255.255.0.0

- DHCP Used in Network

  Users can dial "# #"to obtain the current gateway IP address and version information of firmware using the telephone connected to the subscriber line (FXS interface) after the equipment is powered on.

  If the gateways are only configured with FXO ports for analog trunks without FXS ports for subscriber lines (e.g. WSS8-4FXO), users can dial into the gateway by connecting a PBX extension line or PSTN POTS line to a FXO port, and press "# #" to obtain the current gateway IP address and version information of firmware after receiving the second dial tone.

- Fixed IP Address Used
  - If the DHCP service on the network is not available or the gateway is directly connected with a computer, the gateways will use the factory default IP address.
  - A user could fail to log in with the default IP address if the IP address of user's computer and the default gateway IP address are not at the same network segment. It is recommended that the IP address of user's computer is changed to be identical with the same network segment of gateway. For example, if the gateway IP address is 192.168.2.240, it is recommended to set the computer's IP address to any address at the network segment of 192.168.2.XXX).

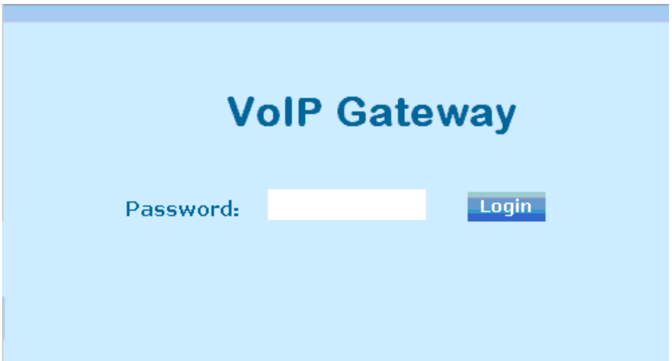
- PPPoE Used

  In "Basic Configuration> Network Configuration", the gateways will automatically obtain the WAN address returned by access network after PPPoE service is started and user name and password are set. Users can dial "# #" on the gateways to receive the IP address and version information of firmware the gateways has obtained.

2.1.2 Log on Gateway

Double-click the icon to open IE browser, and enter the gateway IP address in the browser address bar (eg. 192.168.2.218); you can enter the login interface for gateway configuration by entering a password on the login interface.

Figure 2-1 Login Interface for WSS Gateway Configuration



Both Chinse and English version of WEB are offered.

2.1.3 Permission of Gateway Administrator

Logon users are classified into “administrator” and “operator”. The default password is seen Table 2-2. The password is shown in a cipher for safety.

Table 2-2 Default Passwords of Gateway

Type	Default Administrator Passwords (lowercase letters required)	Default Operator Password
WSS8	voip	operator
WSS60	voip	operator
WSS120	voip	operator

- The administrator can browse and modify all configuration parameters, and modify login passwords.
- The operator can browse and modify part of configuration parameters.

The gateways allow multiple users to log in:

- Ø The administrator has permission for modification and the operator has permission for browsing;
- Ø When multiple users with same level of permission log in, the first has permission for modification, while the others only have permission for browsing.

CAUTION

The system will confirm timeout if users do not conduct any operation within 10 minutes after login. They are required to log in again for continuing operations.

Upon completion of configuration, click "Logout" button to return to the login page, so as not to affect the login permission of other users.

2.2 **Buttons Used on Gateway Management Interface**

“Submit” and “Restore Default Configuration” buttons are at the bottom of configuration interface.

- Submit” Button: Submit configuration information. Users click “Submit” button after completion of parameter configuration on a page. A success prompt will appear if configuration information is accepted by the system; if a “The configuration takes effect after the system is restarted”dialog box appears, it means that the parameters are valid only after system restart; it is recommended that users press the "Restart" button on the "Tool" page to validate the configuration after changing all parameters to be modified.

- "Default" Button: Click this button to use default configuration of gateway. A success prompt will appear on the interface after the system restores parameters on the configuration page to default configuration. For part of parameters, it is required to restart the software to validate the default configuration, and in this case "The configuration takes effect after the system is restarted" will appear on the interface. Subscribers can click "Restart" on the "Tool" page to restart.

2.3 Basic Configuration

2.3.1 Network Configuration

After login, click "Basic > Network" tab to open the configuration interface.

Figure 2-2 Network Configuration Interface

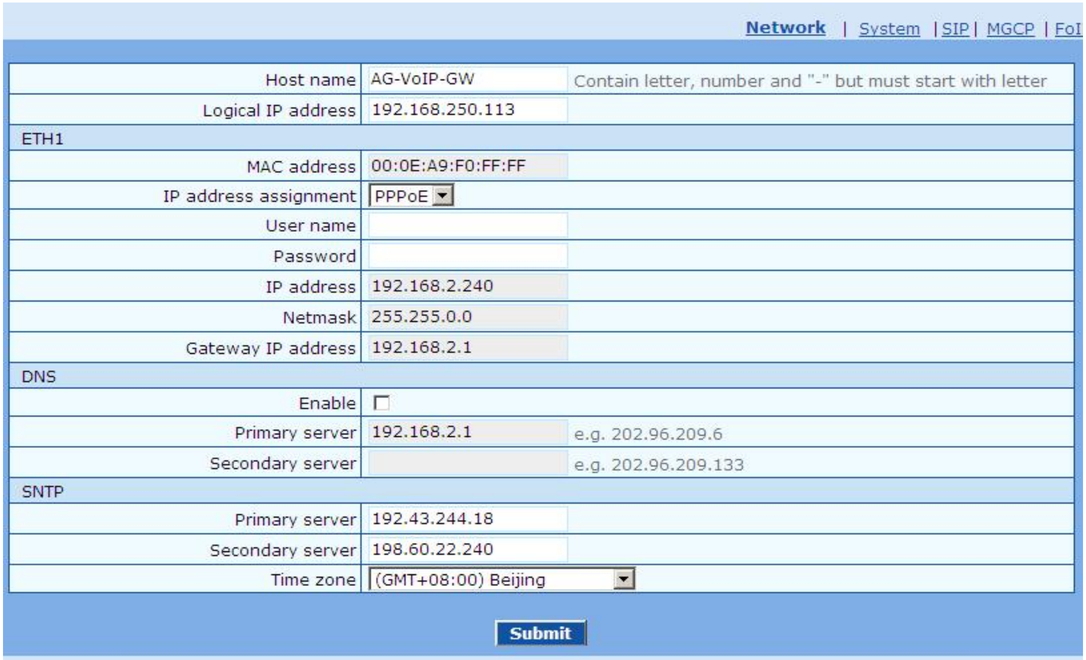


Table 2-3 Network Configuration Parameters

Name	Description
Host name	This is the equipment name of a configuration gateway. The default values of WSS8, WSS60 and WSS120 are WSS8-VoIP-AG, WSS60-VoIP-AG and WSS120-VoIP-AG respectively. Users can set a different name for each gateway to distinguish from each other according to the deployment plan. A host name can be a maximum of 48 characters, either letters (A-Z or a-z), numbers (0-9) and minus sign (-). It may not be null or space, and it must start with a letter.
Logical IP address	This parameter only exists in WSS100-TG, used to display the actual gateway IP address in use.
ETHn	
MAC address	Display the MAC address of gateway.

Name	Description
IP address assignment	Methods for obtaining an IP address ı static: Static IP address is used; ı DHCP: Activate DHCP service and use the dynamic host configuration protocol (DHCP) to allocate IP addresses and other network parameters; ı PPPoE: PPPoE service is used.
User name	Enter an authentication user name if PPPoE service is selected, and there is no default value.
Password	Enter an authentication password if PPPoE service is selected, and there is no default value.
IP address	If “Static” or “DHCP” is selected for the network type but an address fails to be obtained, the gateways will use the IP address filled in here. If the gateways obtain an IP address through DHCP, the system will display the current IP address automatically obtained from DHCP by the gateways. This parameter must be set due to no default value.
Netmask	The subnet mask is used with an IP address. When the gateways use a static IP address, this parameter must be entered; when an IP address is automatically obtained through DHCP, the system will display the subnet mask automatically obtained by DHCP. This parameter must be set due to no default value.
Gateway IP address	LAN gateway IP address where the gateways are located. When the gateways obtain an IP address through DHCP, the system will display the LAN gateway address automatically obtained through DHCP. This parameter must be set due to no default value.
ETH1	Only apply to WSS100-TG
MAC address	Display the MAC address of gateway
IP address	Fill in IP address of ETH1
Netmask	The subnet mask is used with an ETH1 IP address.
DNS	
Enable	Activate DNS service.
Primary Server	If DNS service is activated, the network IP address of preferred DNS server must be entered, and there is no default value.
Secondary Server	If DNS service is activated, the network IP address of standby DNS server can be entered here. It is optional and there is no default value.
SNTP	
Primary Server	Enter the IP address of preferred time server here. This parameter must be set due to no default value.
Secondary Server	Enter the IP address of standby time server here. This parameter must be set due to no default value.

Name	Description
Time Zone	Select a time zone, and the parameter values include: ı (GMT-11:00) Midway Island ı (GMT-10:00) Honolulu. Hawaii ı (GMT-09:00) Anchorage, Alaska ı (GMT-08:00) Tijuana ı (GMT-06:00) Denver ı (GMT-06:00) Mexico City ı (GMT-05:00) Indianapolis ı (GMT-04:00) Glace_Bay ı (GMT-04:00) South Georgia ı (GMT-03:30) Newfoundland ı (GMT-03:00) Buenos Aires ı (GMT-02:00) Cape_Verde ı (GMT) London ı (GMT+01:00) Amsterdam ı (GMT+02:00) Cairo ı (GMT+03:00) Moscow ı (GMT+03:30) Teheran ı (GMT+04:00) Muscat ı (GMT+04:30) Kabul ı (GMT+05:30) Calcutta ı (GMT+05:00) Karachi ı (GMT+06:00) Almaty ı (GMT+07:00) Bangkok ı (GMT+08:00) Beijing ı (GMT+09:00) Tokyo ı (GMT+10:00) Canberra ı (GMT+10:00) Adelaide ı (GMT+11:00) Magadan ı (GMT+12:00) Auckland

2.3.2 System Configuration

After login, click “Basic > System” tab to open the configuration interface.

Figure 2-3 System Configuration Interface

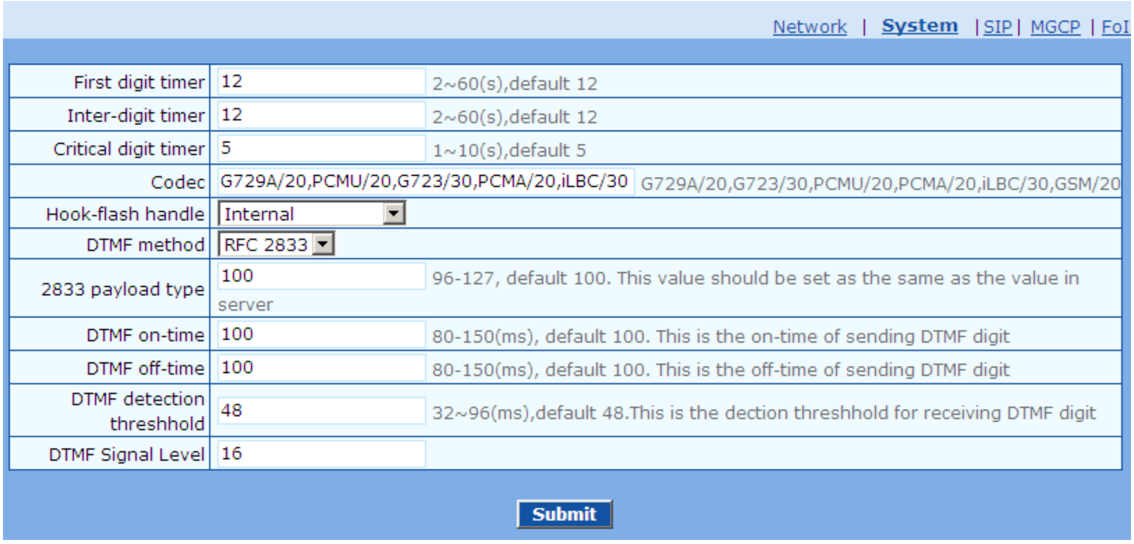


Table 2-4 System Configuration Parameters

Name	Description
First digit timer	If a subscriber hasn't dialed any number within a specified time by this parameter after offhook, the gateways will consider that the subscriber has given up the call and prompt to hang up in busy tone. Unit: second; Default value: 12 seconds.
Inter-digit timer	If a subscriber hasn't dialed the next number key from the time of dialing the last number key to the set time by this parameter, the gateways will consider that the subscriber has ended dial-up and call out the dialed number. Unit: second; Default value: 12 seconds.
Critical digit timer	This parameter is used with the "x.T" rule set in dialing rules. For example, there is "021.T" in the dialing rules table. When a subscriber has dialed 021 and hasn't dialed the next number within a set time by this parameter (eg. 5 seconds), the gateways will consider that the subscriber has ended dial-up and call out the dialed number 021. Input intergers, not decimal fractions Unit: second; Default value: 5 seconds.
Codec	Codecs methods supported by the gateways include G729A/20, G723/30, PCMU/20, PCMA/20, iLBC/30 and GSM/20 (as shown in table 2-5). This parameter must be set due to no default value. Several encoding methods can be configured in this item at the same time, separated with "," in the middle; the gateways will negotiate with the platform in the order from front to back when configuring the codec methods
Hook-flash handle	The gateways provide the following processing modes after detecting hook flash from subscriber terminals: processing the hook flash internally; transmitting the hook flash to platform with RFC 2833, and transmitting the flash-off to platform with SIP INFO.
DTMF method	Transmission modes of DTMF signal supported by the gateways include Audio, RFC 2833 and SIP INFO. The default value is Audio. Audio: DTMF signal is transmitted to the platform with sessions; SIP INFO: Separate DTMF signal from sessions and transmit it to the platform in the form of SIP INFO messages; RFC 2833: Separate DTMF signal from sessions and transmit it to the platform through RTP data package in the format of RFC2833.

Name	Description
2833 payload type	Used with "RFC 2833" in the DTMF transmission modes. The default value of 2833 payload type is 100. The effective range available: 96 ~ 127. This parameter should match the setting of far-end device (eg. platform).
DTMF on-time	This parameter sets the on time (in ms) of DTMF signal sent from FXO port. The default value is 100 ms. Generally, the duration time should be set in the range of 80 ~ 150 ms.
DTMF off-time	This parameter sets the off time (ms) of DTMF signal sent from FXO port. The default value is 100 ms. Generally, the interval time should be set in the range of 80 ~ 150 ms.
DTMF detection threshhold	Minimum duration time of effective DTMF signal. Its effective range is 32-96 ms. The greater the value is set, the more stringent the detection is.

Table 2-5 Codec Methods Supported by Gateways

Codec Supported by WSS	Bit Rate (Kbit/s)	Time Intervals of RTP Package Sending (ms)
iLBC	13.3/15.2	20/30
GSM	13	20
G729A	8	10/20/30/40
G723	5.3/6.3	30/60
PCMU/PCMA	64	10/20/30/40

2.3.3 SIP Configuration

After login, click "Basic> SIP" tab to open the SIP configuration interface.

Figure 2-4 SIP Configuration Interace

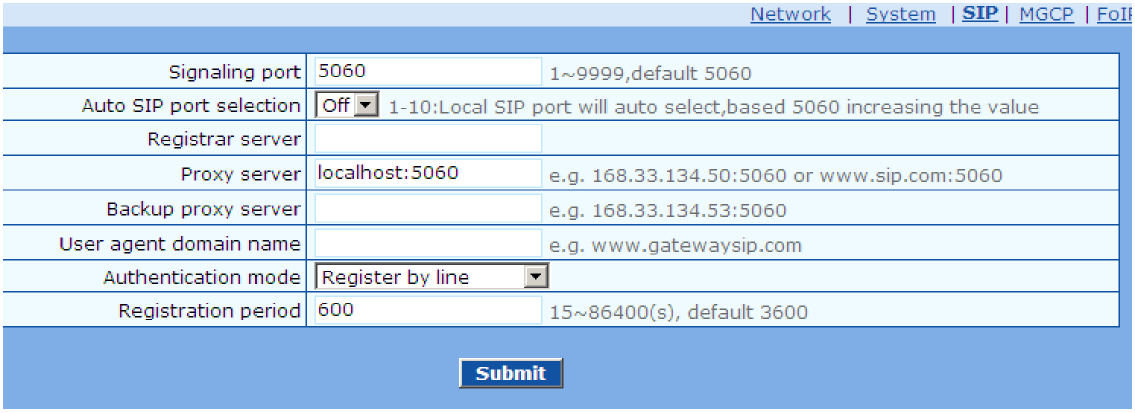


Table 2-6 SIP Configuration Parameters

Name	Description
Signaling port	Configure the UDP port for transmitting and receiving SIP messages, with its default value 5060. Note: The signaling port number can be set in the range of 1-9999, but cannot conflict with the other port numbers used by the equipment.

Name	Description
Change signaling port	If “n”(ranked from 1-10) is chosen, after the failure registration of signaling port’s original configuration, the range of signaling port’s change varies from “original signaling port, original signaling port +n”. Register with the new signaling port value (signaling port +1) until it succeds.
Register server	Configure the address and port number of SIP register server, and the address and port number are separated by “:”. It has no default value. The register server address can be an IP address or a domain name. When a domain name is used, it is required to activate DNS service and configure DNS server parameters on the page of configuring network parameters. For example: “201.30.170.38:5060”, “register.com: 5060”.
Proxy server	Configure the IP address and port number of SIP proxy server, and the address and port number are separated by “:”. It has no default value. The proxy server address can be set to an IP address or a domain name. When a domain name is used, it is required to activate DNS service and configure DNS server parameters on the page of configuring network parameters. Examples of complete and effective configuration: "201.30.170.38:5060", "softswitch.com: 5060".
Backup proxy server	By specifying the corresponding IP addresses, the gateway can be configured to have multiple soft switches as backup proxy servers. Make sure that the IP addresses must be in their full format. Eg. “202.202.2.202:2727”. The proxy and register severs must be identical. Conditions for falling over to the backup proxy server (any): 1）Gateway register is timeout; 2）No response to master server calls is timeout;
User agent domain name	This domain name will be used in INVITE messages. If it is not set here, the gateways will use the IP address or domain name of proxy server as user agent domain name. It has no default value. It is recommended that subscribers not use LAN IP address to set domain name parameter.
Authentication mode	The gateway supports three registration shemes: register per line, register per gateway and Line Reg/GW Auth. The default value is register by line. l Register by line: authentication and register per line; l Register by gateway: authentication and register per gateway; l Line Reg/GW Auth: register per line, but authentication per gateway.
User name	Configure the user name as part of the account for registration, and it has no default value. Note: If “Register by gateway” or “Line Reg/GW Auth”, is selected, the user name must be entered here. If “register by line” is selected the user name should be set on “Line > Feature” page (Refer to “Feature”).
Password	Password as part of account information is used for authentication by platform. It has no default value. It is formed with either numbers or characters, and case sensitive. Note: If “Register by gateway” or “Line Reg/GW Auth”, is selected, the password must be entered here. If “register by line” is selected the password should be set on “Line > Feature” page (Refer to “Feature”).
Registration period	Valid time of SIP re-registration in second.

2.3.4 MGCP Configuration

The gateways use SIP protocol by default. When the gateways need to interface with MGCP protocol -based softswitch platform, users should set relevant parameters here.

After login, click "Basic > MGCP" tab to open the configuration interface.

Figure 2-5 MGCP Configuration Interface

Table 2-7 MGCP Configuration Parameters

Name	Description
Signaling port	Configure the UDP port for transmitting and receiving MGCP messages, and default value is 2427. Note: The signaling port number can be set in the range of 1-9999, but cannot conflict with the other port numbers used by the equipment.
Proxy server	Configure the IP address and port number of MGCP proxy server, separated by ":", and it has no default value. The address can be set to an IP address or a domain name according to the subscribers' requirements. When a domain name is used, it is required to activate DNS service and configure DNS server on the page of configuring network parameters. Examples of complete and effective configuration: "202.202.2.202:2727", "callagent.com: 2727".
User agent domain name	It is the gateway logo when the gateways register with proxy server, and it has no default value. Example: test.RealTone.com, [192.168.2.100]. Note: if IP address is written in this way, like "[192.168.2.100]", "[]" should be added.
Default event package	List all the types of default event packages supported by gateways, and multiple package names are separated by",". The default value is L, D, G ı L: Line Package; ı D: DTMF Package; ı G: Generic Media Package.

Name	Description
Persistent line event	List the event types that the gateway can report, and persistent line events are separated by “,”. When gateways process the events listed here, they will report to the call agent. Note: This parameter must be set since there is no default value. The factory setting is L/HD, L/HU: l L/HD: Offhook; l L/HU: Onhook.
FXO event package	Handset Package Line Pakage
Wildcard	Select whether a wildcard with prefix is allowed when a gateway registers to the proxy server. The default value is “not allowed”. l Partially allowed: Gateways will use a wildcard with fixed prefix (eg. aaln / *) when registering. For example, when configuring telephone numbers, if line 1 is set to “aaln/1”, line 2 is set to “aaln/2” and line 3 is set to “aaln/3”, the gateways will register to the call agent in “aaln/*” without the need of registering the lines individually. l Allowed: the gateways will use a wildcard in registering without prefix.
Compatibility Configuration	
CR for End-of-Line	Select whether CR is used as the end of line in the MGCP messages. Default not selected.
Quarantine default to loop	Select the Qurantine handle of gateways making a request to the outside, and default not selected. l Selected: Quarantine using loop mode, the gateways will continually Notify all events as requested after receiving a request.
Enable first digit timer	Select the processing mode when there is no timeout parameter in the outside request received by the gateways, and default not selected. l Selected: the gateways will report timeout in terms of its own timeout setting (the time interval set in non-dial timeout of configuration system parameters) when subscribers hasn’t dialed up in time after offhook.
Using configured digit map	Select whether to activate the digit map configured by local gateway, and default value is not selected.
Using notify instead of 401/402	Set whethr the gateways report “offhook events” to replace 401 messages in NTFY or report “onhook events” to replace 402 messages in NTFY when responding to messages sent by the proxy server. Default: not selected. l Selected: The gateways will use NTFY message to replace 401 and 402 messages.
No name in default package	Select if a package name is included when the gateways reply to the default package, and default not selected.
Keep connection when on-hook	Select if the gateways actively cancel connection disconnect when subscribers hook on, and default not selected.

2.3.5 FoIP

After login, click the label of “Basic > FoIP” to open this interface.

Figure 2-6 fax configuration interface

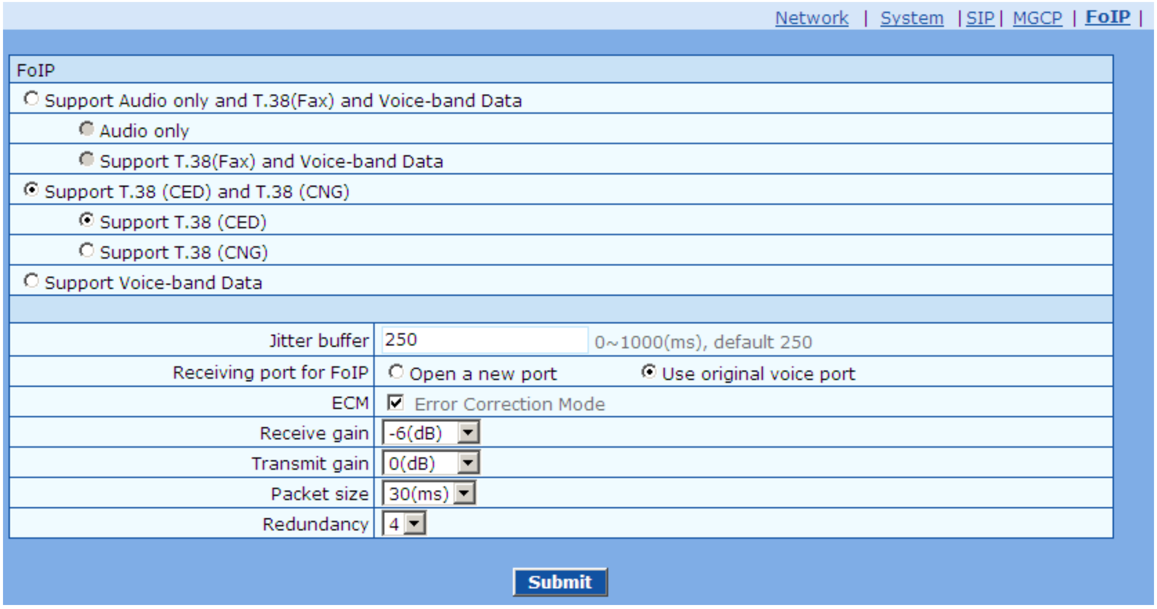


Table 2-8 fax configuration parameters

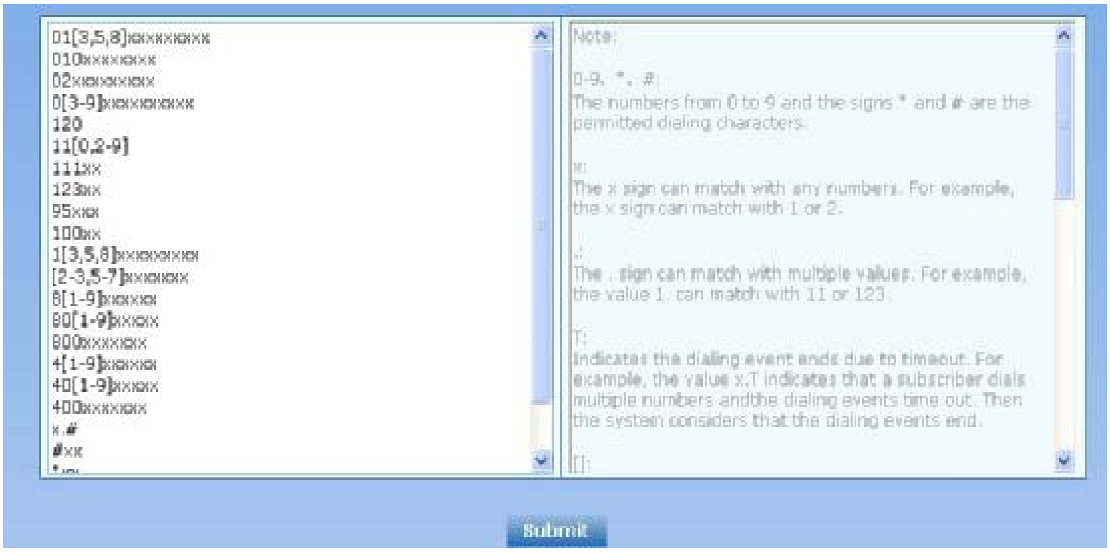
Title	Description
T.30, POS, MODEM only	Audio only Support T.38 (Fax) and voice-band data
T.38 only	Support T.38 (CED) Support T.38 (CNG)
The following are configurable parameters when T.38 activated	
Jitter buffer	Set the extent of T.38 jitter buffer, and the default is 250. The valid range is 40~1000 in milliseconds.
Receiving port for FoIP	Set whether to open a new port when the gateway is switching to T.38 mode, and by default, original voice port will be usd. Open a new port: use the new RTP port. Use original voice port: use the original RTP port that created on call set.
ECM	Determine whether to use corrective mode of fax. By default, it is not selected.
Receive gain	Set the receiving gain of T.38 fax, with the default of 6dB.
Transmit gain	Set the transmission gain of T.38 fax, with the default of 0dB.
Packet size	Set the packet size of T.38. 30 miliseconds is the default value.
Redundancy	Set the number of the redundant frames in T.38 date package, default is 4.

2.4 Routing

2.4.1 Digit Map

After login, click “Routing> Digit Map” tab to open the dialing rules interface as shown in Figure 2-7.

Figure 2-7 Configuration Interface for Digit Map



Dialing rules are used to effectively judge if the received number sequence is completed, for the purpose of ending up receiving numbers and sending out the received numbers.The proper use of dialing rules can help to reduce the connection time of telephone calls.

The maximum number of rules that can be stored in gateways is 60. Each rule can hold up to 32 numbers and 38 characters. The total length of dialing rules table (the total length of all dialing rules) can be up to 2280 bytes.

The following provides a description of tipical rules:

Table 2-9 Description of Digit map

Digit map	Description
“x”	Represents any number between 0-9.
“.”	Represents more than one digit between 0-9.
“##”	End after receiving two-digit dialing “##”. “##” is a special dialing for users to receive gateway IP address and version number of firmware by default.
“x.T”	The gateways will detect any length of telephone number starting with any number between 0-9. The gateways will send the detected number when it has exceeded the dialing end time set in system parameter configuration and hasn’t received a new number.
“x.#”	Any length of telephone number starting with any number between 0-9. If subscribers press # key after dial-up, the gateways will immediately end up receiving numbers and send all the numbers before # key.
“*xx”	End after receiving * and any two-digit number. "* xx" is primarily used to activate function keys for supplementary services,such as CRBT, Call Transfer, Do not Disturb, etc.

Digit map	Description
“#xx”	End after receiving # and any two-digit number. “#xx” is primarily used to stop function keys for supplementary services, such as CRBT, Call Transfer, Do not Disturb, etc.
[2-8]xxxxxx	A 7-digit number starting with of any number between 2-8, used to end the dialing.
02xxxxxxxxx	A 11-digit number starting with 02, used to end the long-distance dialings starting with “02”.
013xxxxxxxxx	A 12-digit number starting with 013, used to end long-distance calls
13xxxxxxxxx	A 11-digit number starting with 13, used to end the dialings.
11x	A 3-digit number starting with 11, used to end the emergency calls.
9xxxx	A 5-digit number starting with 9, used to end special service calls.
17911 (eg.)	Send away when the set number, like 17911, is received.

Dial rules by default as follows:

01[3,5,8]xxxxxxxxx

010xxxxxxxx

02xxxxxxxxx

0[3-9]xxxxxxxxxx

11[0,2-9]

111xx

9[5,6]xxx

100xx

10[1-9]

12[0-2,4-9]

1[3,5,8]xxxxxxxxx

[2-3,5-7]xxxxxxx

[4,8][1-9]xxxxxx

[4,8]0[1-9]xxxxx

[4,8]00xxxxxxx

x.T

x.#

#xx

*xx

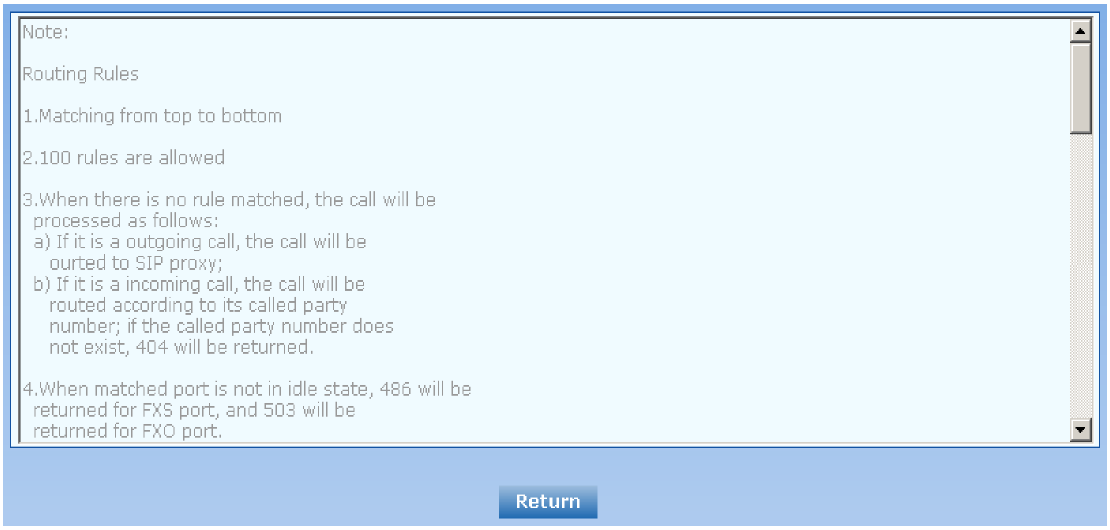
##

2.4.2 Routing Table

After login, click “Routing> “Routing Table” tab to open the configuration interface.

Figure 2-8 Configuration Interface for Routing Table

Click "Help" to open the illustrative interface for routing configuration



The routing table with 500 rules in capacity provides two functions including digit transformation and call routing assignment. Here are the general rules applied by gateways when executing the routing table:

CAUTION

Rules must be filled out without any blank at the beginning of each line; otherwise the data can't be validated even if the system prompts successful submital.

The routing table is empty by default. The gateways will point a call to the SIP proxy server when there is no matched rule for the call.

The format of number transformation is

Source Number Replacement Method

For example: "FXS 0755 REMOVE 4" means removing the prefix 0755 of the called number for calls from FXS port, where "FXS" is source, "0755" is number, and "REMOVE 4" indicates the method of number transformation.

The format of routing rules is

Source Number ROUTE Routing Destination

For example: “IP 800[0-3] ROUTE FXO 1,2,3,4” means routing calls from IP with called number between 8000~8003 to FXO port in a sequential selecting order of 1, 2, 3, 4. Namely, FOX Port 2 is selected when FXO Port 1 is busy and so on.

Detailed definitions of source and number, number transformation moethods and routing destination are shown below.

Table 2-10 Routing Table Format

Name	Description
Source	There are three types of source: IP, FXS and FXO. Among them, IP source can be any IP address and is denoted by “IP”; “IP [xxx.xxx.xxx.xxx]” is used to denote specific IP address; “IP [xxx.xxx.xxx.xxx: port]” is used to denote specific IP address with port number. FXS and FXO ports can be any port, represented with “FXS” or “FXO”; special lines can be represented with FXS or FXO + port number, eg. FXS1, FXO2 or FXS [1-2], etc.
Number	It chould be a calling number with the form of CPN + number or a called number with the form of number. The number may be denoted with digit 0-9,"*",".","#"," x ", etc., and uses the same regular expression as that of dialing rules. Here are some of the form of number: - Designate a specific number: eg.114，83501950; - Designate a number matching a prefix: such as 61xxxxxx. Note: the matching effect of 61xxxxxx is different from that of 61x or 61. Number matching follows the principle of “minimum priorary matching ” - Specify a number scope. For example, 268[0-1，3-9] specifies any 4-digit number starting with 268 and followed by a digit between 0-1or 3-9; Note: Number matching follows the principle of “minimum matching ”. For example: x matches any number with at least one digit; xx matches any number with with at least two-digit; 12x matches any number with at least 3-digit starting with 12.

Table 2-11 Number Transformations

Processing Mode	Description and Example
KEEP	Keep number. The positive number behind KEEP means to keep several digits in front of the number; the negative number means to keep several digits at the end of the number. Example: FXS 075580501950 KEEP -8 Keep the last 8 digits of the called number 075583501950 for calls from FXS. The transformed called number is 83501950.
REMOVE	Remove number. The positive number behind REMOVE means to remove the first several digits of the number; the negative number means to remove the later several digits of the number. For example: FXS 0755 REMOVE 4 Remove 0755 of the called number beginning with 0755 for calls from FXS.

Processing Mode	Description and Example
ADD	Add prefix or suffix to number. The positive number behind ADD is the prefix; the negative number is suffix. Example 1: FXS1 CPNX ADD 0755 FXS2 CPNX ADD 010 Add 0755 in front of calling numbers for calls from FXS port 1; add 010 in front of calling numbers for calls from FXS port 2. Example 2: FXS CPN6120 ADD -8888 Add 8888 at the end of the calling number starting with 6120 for calls from FXS port.
REPLACE	Number replacement. The replaced number is behind REPLACE. Example: FXS CPN88 REPLACE 2682000 Replace the calling number beginning with 88 for calls from FXS port to 2682000.
REPLACE	Other use of REPLACE is to replace the specific number based on other number associated with the call. For example, replace the calling number according to the called number. Examples: FXS 12345 REPLACE CPN-1/8621 FXS CPN13 REPLACE CDPN0/0 For calls from FXS ports with called party number of 1234, removing one digit at the rear of the calling number and add 8621; for calls from FXS ports with calling party number starting with 13, add 0 in front of the called number.
END or ROUTE	End of number transformation. From top to bottom, number transformation will be stopped when END or ROUTE is encountered; the gateways will route the call to the default routing after meeting EDN, or route the call to the designed routing after meeting ROUTE. Example 1: FXS 12345 ADD -8001 FXS 12345 REMOVE 4 FXS 12345 END Add suffix 8001 to the called number starting with 12345 for calls from FXS ports, then remove four digits in front of the number to end number transformation. Example 2: IP[222.34.55.1] CPNX. REPLACE 2680000 IP[222.34.55.1] CPNX. ROUTE FXS 2 For calls from IP address 222.34.55.1, calling party number is replaced by 2680000, and then the call is routed to FXS port 2 with the new calling party number.
CODEC	Designate the use of codec, such as PCMU/20/16, where PCMU denotes G.711, /20 denotes RTP package interval of 20 milliseconds, and /16 denotes echo cancellation with 16 milliseconds window. PCMU/20/0 should be used if echo cancellation is not required to activate. Example: IP 6120 CODEC PCMU/20/16 PCMU/20/16 codec will be applied to calls from IP with called party number starting with 6120.

Processing Mode	Description and Example
RELAY	Insert prefix of called party number when calling out. The inserted prefix number follows behind REPLAY. Example: IP 010 RELAY 17909 For calls from IP with called party number starting with 010, digit stream 17909 will be outpulsed before the original called party number being sending out.

Table 2-12 Routing Destination

Destination	Description and Example
ROUTE NONE	Calling barring. Example: IP CPN[1,3-5] ROUTE NONE Bar all calls from IP, of which the calling numbers start with 1, 3, 4, 5.
ROUTE FXS	Route a call to FXS ports. Example 2: IP 800[0-3] ROUTE FXS 1 Point this call to FXS port 1. Example 3: IP 800[0-3] ROUTE FXS 1,2,3,4/g For terminating the call from IP, ring FXS port 1, 2, 3, 4 simultaneousely.
ROUTE FXO	Route a call to FXO port. Example: IP x ROUTE FXO 1,2,3,4/r Select the outgoing call FXO port in a round robin way.
ROUTE IP	Route a call to the IP platform. Example: FXS 021 ROUTE IP 228.167.22.34:5060 228.167.22.34:5060 is the IP address of the platform.

2.4.3 Application Examples of Routing Table

Some typical functions that can be realized by the routing table are provided in this section:

1) One Phone with Double Numbers
2) Hunting Group
3) Outbound Call Barring
4) FXO Port Hunting for Outbound Call

One Phone with Double Numbers

The hand set connected to gateway can be configured with two numbers through One Phone with Double Numbers. For example, port FXS1 is set with PSTN number 83501950 and extention number 1001 for internal calling

Routing Setting

FXS 1001 ROUTE IP 127.0.0.1:5060

IP 1001 ROUTE FXS 1

Description:

1) Send the call with the called number starting with 1001 from FXS port to port 5060 of gateway's local IP;
2) Send the call with the called number starting with 1001 and from any IP to the FXS port 1.

Configuration number of FXS1 itself is 83501950, so the call of this number is not required to write specialized routing.

Hunting Group

A hunting group can be associated with a set of FXS ports, and an inbound call from IP or FXO ports can be routed to a hunting group. There are three circuit selection algorithms available: 1) sequential selection, 2) circular selection, and 3) simultaneous ringing.

Routing Setting:

Take WSS8-4S/4 gateway as an example. Send the inbound call from IP trunk and analog line in a circular way to the phone set on the 2nd or 3 rd FXS port.

FXO x ROUTE IP 127.0.0.1:5060

IP x ROUTE FXS 2, 3

Description:

1) Send all calls from FXO port to port 5060 of gateway's local IP;
2) Send all inbound calls from any IP (inside and outside) to the 2nd or 3rd FXS port in sequence. Namely, the 2nd FXS port is selected firstly when it is free, otherwise the 3rd port is selected.

Outbound Call Barring

Restrict users to dial certain telephone numbers, such as an international call. Examples are as follows:

Routing Setting	Description
FXS[1] 0 ROUTE NONE	A calling starting with 0 is barred to dial using the phone set at FXS1 port.
FXS[1-4] 00 ROUTE NONE	A calling starting with 00 is barred to dial at 1-4 FXS ports.
FXS CPN2 ROUTE NONE	The telephone whose calling number starts with 2 at FXS port is barred to call out.

FXO Port Hunting with Outbound Calls

Routing Setting (take WSS8-4S/4 as an example):

FXS x ROUTE IP 127.0.0.1:5060

IP x ROUTE FXO 1,2,3,4/r

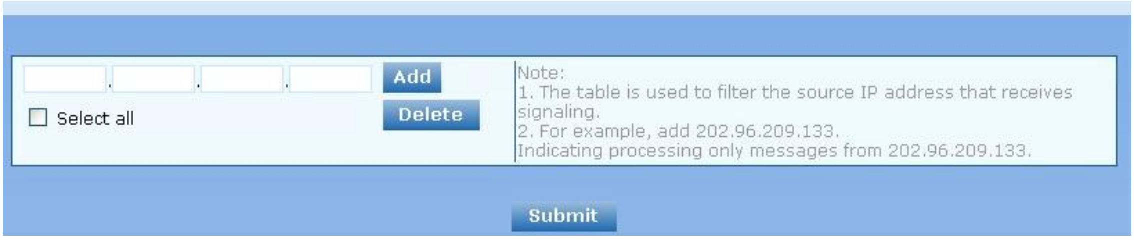
Description:

1) Send all calls from FXS port to port 5060 of gateway's local IP (this port must be consistent with the local port in "Configuring SIP");
2) Send all calls from any IP to FXO port for round selection in an order;

2.4.4 IP Table

After login, click "Routing> "IP Table" tab to open the configuration interface.

Figure 2-9 Configuration Interface for IP Table



This table is designed to ensure the safe use of gateways. Administrators can add the authorized IP addresses to this table, and the gateways will only process the information from authorized IP addresses. If the IP table is empty, the gateways will not perform IP address-based message filtering.

For example: the gateway will only process the messages from 202.96.209.133 after adding 202.96.209.133 to its IP table.

2.5 Line Configuration

2.5.1 FXS Phone Number

After login, click "Line > FXS phone number" tab to open the configuration interface.

Figure 2-10 Configuration Interface for Telephone Number

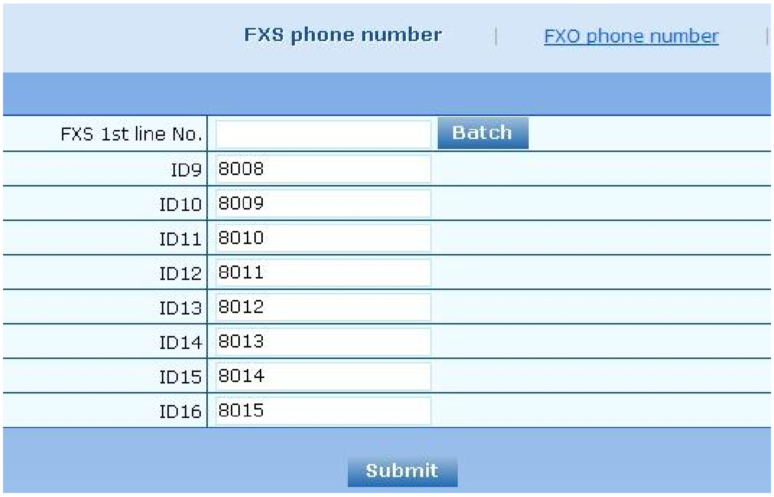


Table 2-13 Configuration Parameters of Telephone Number

Name	Description
FXS 1st line No.	This number is used for the batch setup of consecutive number of subscriber line. Click "Batch "after filling in initial number, the number of Line 1 adopts initial number; that of Line 2 increases 1 progressively based on that of Line 1, and so on. You needn't fill in if you do not use batch configuration or the number is not consecutive.
ID n	Fill in the telephone number associated with the subscriber line n (FXS port). This should be manually performed if Batch mode is not used.

2.5.2 FXO Phone Number

After login, click “Line > FXS phone number” tab to open the configuration interface.

Figure 2-10 Configuration Interface for Telephone Number

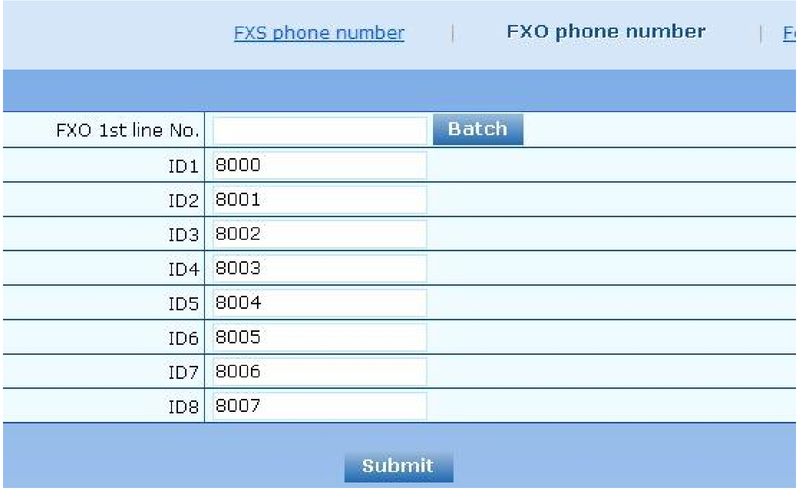


Table 2-13 Configuration Parameters of Telephone Numbers

FXO 1st line No.	This number is used for the fast setup of consecutive number of trunk line. Click "Batch " after filling in initial number, the number of Line 1 adopts initial number; that of Line 2 increases 1 progressively based on that of Line 1, and so on. You needn’t fill in if you do not use batch configuration or the number is not consecutive.
ID n	Fill in the telephone number associated with the trunk n (FXO port). This should be manually performed if Batch mode is not used.

2.5.3 Subscriber Line Features

This page is only used for configuring gateways with subscriber lines (FXS port).

After login, click “Line > Feature” tab to open the configuration interface.

Figure 2-11 Configuration Interface for Subscriber Line Features

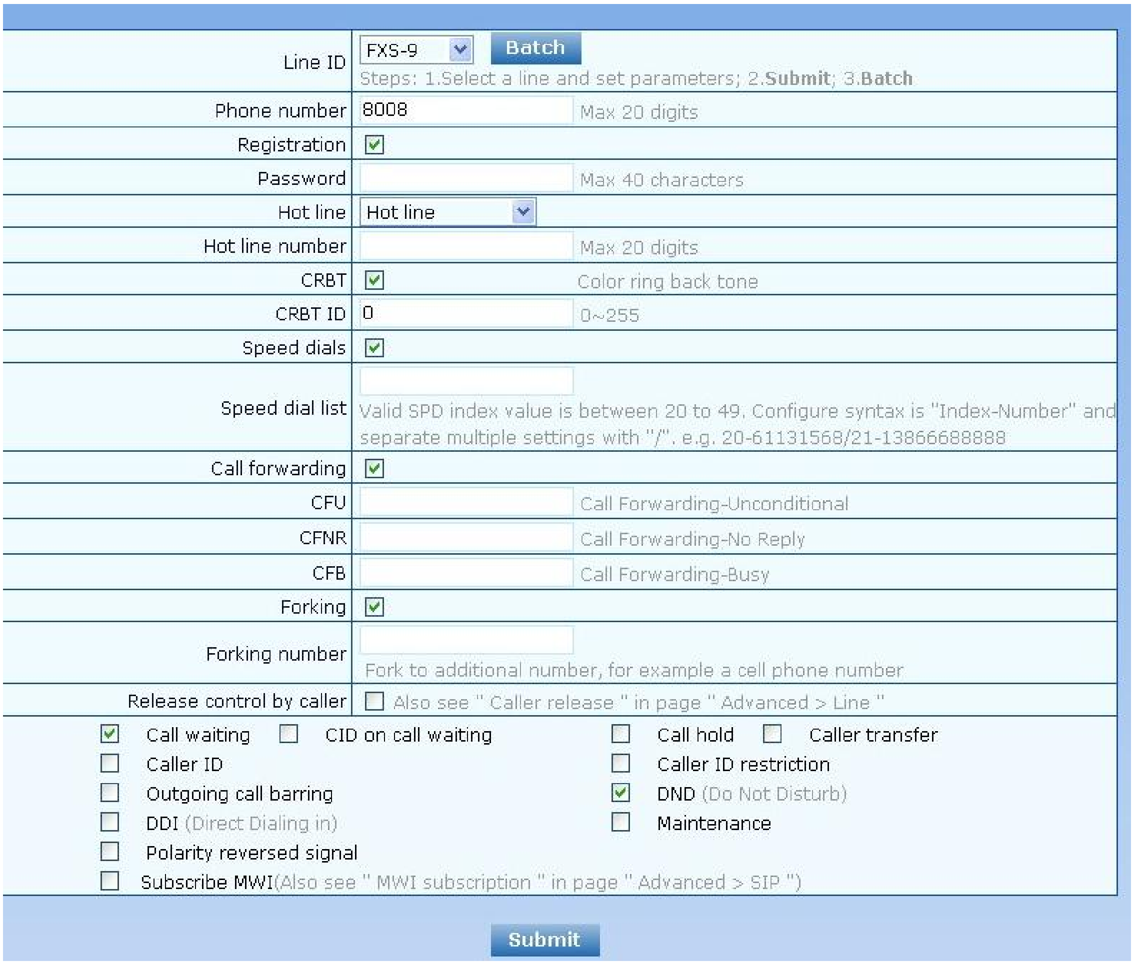


Table 2-14 Configuration Parameters of Subscriber Line Features

Name	Description
Line ID	Select a subscriber line required to configure. “FXS -n” corresponds to the “Line > FXS phone Number > ID n”. Copy the configuration of “FXS -n” for selected line to “FXS -n+1”~“FXS - m” by clicking “Batch”, where n indicates the current selected subscriber line number and m indicates the total number of subscriber lines.
Phone number	Display the Telephone Number of this line set in “Line Configuration > FXS phone Number”. Users can input or change the telephone number here.
Registration	Setelect if this line is required to register to softswitch. This is selected as default.
Password	If the “Registration” is selected, users need to enter the authentication password for register of this line here.

Name	Description
Note: The following features are valid only in SIP protocol. When the gateways use MGCP protocol, features are controled by the proxy server without the need of setting on gateways.	
Hot line	Select if the gateways are required to automatically dial out the hotline number after offhook. By default, hot line is disabled. l Disable hot line: Close this feature. l Hot line: Automatically dial out the hotline number after offhook. l Delayed hot line: Automatically dial out the hotline number when the offhook is timeout with a time delay of 5 seconds.
Hot line number	After the hotline function is activated on this line, the hotline number must be entered here.
CRBT	CRBT stands for Color Ring Back Tone. Set if CRBT is activated, that is, provide prepared audio package as ringback tone. Note: it is required to set the CRBT file download platform. This is not selected by default. WSS8 support two CRBT storage modes: on-system (stored in a flash memory) and run-time download (from FTP server). The capacity of both modes are described as follows: On-sytem: l WSS8: No more than 20 seconds in G.729 format (fring1.dat) Run-time download: l WSS8: Up to 20 tone files, a maximum of 10000 seconds in G.729 format or 1250 seconds in G.711 format. Note:Tone files are stored in the flash memory and the gateways automatically download the tone files from FTP server after power on.
CRBT ID	Set the CRBT number with a valid rang of 0~255, where 0 indicates disabling CRBT. The default value is 0.
Speed dials	Select if the Speed dials is activated on this line. By default, this is not selected.
Speed dial list	If the Speed dials is activated on this line, enter the speed dials list. The abbreviated number consists of max 30 pairs of “abbreviated number-real number” with an minus sign between them; “abbreviated number-real number” pairs are separated by “/”; the value range of abbreviated number is 20 ~ 49. For example: 20-83501950/23-13952475822/30-96961. Users can set the list on a telephone set and display it here.
Call forwarding	Select if Call forwarding is activated on this line. By default, it is not selected.
CFU	If it is required to forward all incoming calls unconditionally, enter the new destination number here.
CFNR	If it is required to forward an incoming call when there is no answer, enter the new destination number here.
CFB	If it is required to forward an incoming call when it is busy, enter the new destination number here.
Forking	Select if the Forking is activated. Forking allows the gateway to initiate a call to another telephone terminal while ringing on this line terminal, and the answer by either terminal will end up with ringing of the other terminal.
Forking number	If forking of this line is activated, set a number for the second ringing terminal here. The ringing terminal can be another port of gateways or an external terminal such as mobile phone.

Name	Description
Release control by caller	Select if the call release is controlled by the caller. By default, this is not selected. ı Selected: The gateway will immediately release the call upon caller hanging up; the gateway will not release the call as long as the caller is still off until timeout (60 seconds by default); ı Unselected: The gateway will immediately release the call upon either party hanging up the call.
Call waiting	Select if Call waiting is activated on this line. By default this is not selected.
CID on call waiting	Select if Caller ID Display is activated on this line during call waiting. By default, this is not selected.
Call hold	Select if Call Hold is activated on this line. By default this is not selected. Note: If this function is activated, the gateways will automatically activate Call Transfer (Either party may transfer the current call to a third party).
Caller Transfer	Select if Caller Transfer is activated on this line. By default, this is not selected. When A calls to B, B picks up the call and transfers the call to C,. Note: The call hold must be activated before caller transfer.
Caller ID display	Set whether Caller ID display is activated on this line. By default, this is selected. Note: In addition to number display, the Caller ID can display the names of incoming calls as long as terminal equipments support.
Caller ID restriction	Set whether the number of this telephone is sent to the called party with support from platform. By default this is not selected.
Outgoing call barring	Select if outgoing calls are barred on this line. By default, this is not selected.
DND	Select if “Do Not Disturb” is activated on this line. By default, this is not selected.
Direct Dialing in (DDI)	Set whether DDI (Direct Dialing In) is activated, By default this is not selected. Different from FXS, DDI is only used for incoming calls, and the gateways will not send dial tone after off-hook (calling in) on user side. Note: Reverse polarity signal must be activated on the gateways when DDI is used.
Maintenance	Select if the line is set to maintenance status, namely, stop to supply of power for the line port. By default, this is not selected.
Polarity reversed signal	Select if reverse polarity signal is activated on this line. By default, this is not selected. Note: The gateways will provide reverse polarity signal when the phone is connected after this feature is activated.
Subscribe MWI	Select if voice mail service is activated, and by default this is not selected. (Used with “Advanced > SIP” Interface “MWI subscription” Configuration)

2.5.4 Trunk Line Features

This page is only used for configuring gateways with trunks (FXO port).

After login, click “Line > Trunk” tab to open the configuration interface.

Figure 2-12 Configuration Interface for Trunk Line Features

Table 2-15 Configuration Parameters of Trunks

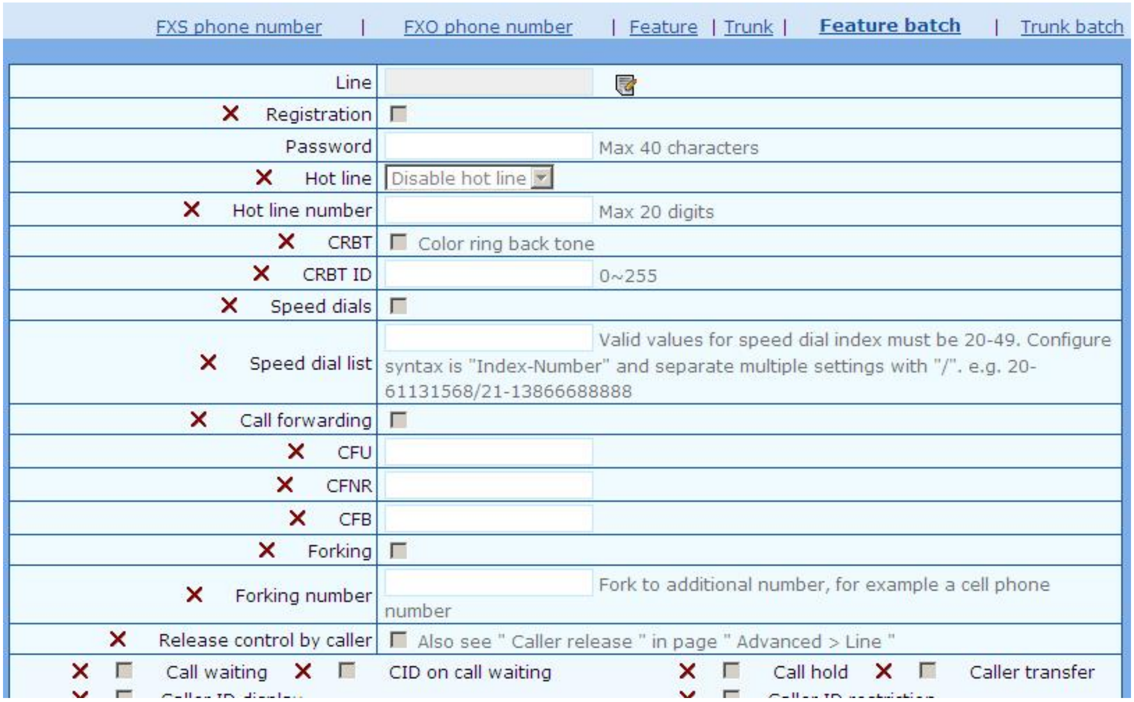
Name	Description
Trunk ID	Select a trunk line required to configure. "FXO-n" corresponds to the "Line > FXO phone Number ID n". Copy the configuration of "FXO-n" for selected line to "FXO-n+1"~"FXO- m" by clicking "Batch", where n indicates the current selected trunk number and m indicates the total number of trunks.
Phone number	Display phone number associated with the trunk set in " Line > FXO phone Number"
Registration	Select if this trunk registers with the SIP registeration server. By default, this is selected.
Password	If the "Registration" is selected, the authentication password for register of this line must be entered here.
Note: The following features are valid only in SIP protocol. When the gateways use MGCP protocol, the control of all call services is provided by the proxy server without the need of setting.	
Inbound handle	The gateways provide two scenarios for handling incoming calls of FXO port: l "Second stage dialing": When a telephone call comes to the FXO port, the gateways will provide the second dial tone and route the call according to the extension number pressed in. Note: dialing tone or voice prompt file can be changed by user. l "Binding": When a telephone call comes to the FXO port, the gateways will route the call to a FXS port according to the DID number bound with the port. Note: Setting a number to be bound is required or this setting is invalid.
Polarity reversed signal detection	If a PSTN line supports reverse polarity, make a selection here. Or this setting is invalid. By default, this is not selected.
Caller ID detection	Select if the detection function of caller ID for this FXO port is enabled. By default, this is selected.

Name	Description
Outgoing call barring	Select if this FXO port bars outgoing call service to PSTN. By default, this is not selected.
Echo cancellation	Select if echo cancellation is enabled for this FXO line.By default, this is selected.
Connect signal delay	After making an outgoing call from a FXO port, the gateway will send a 200 OK message to the platform with delay if this parameter is selected. If unselected, the system sends a 200 OK message to the platform after off hook on the FXO port. Used with the configuration item of "Answer delay" on the "Advanced > Trunk" interface.

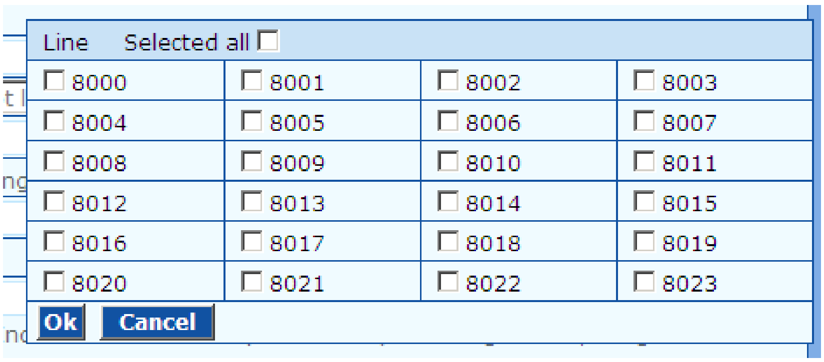
2.5.5 Feature Batch

After login, click "Line > Feature Batch" to open this interface.

Figure 2-13 feature batch configuration interface



Click , the following interface is shown. Choose batch configured features and click "ok"

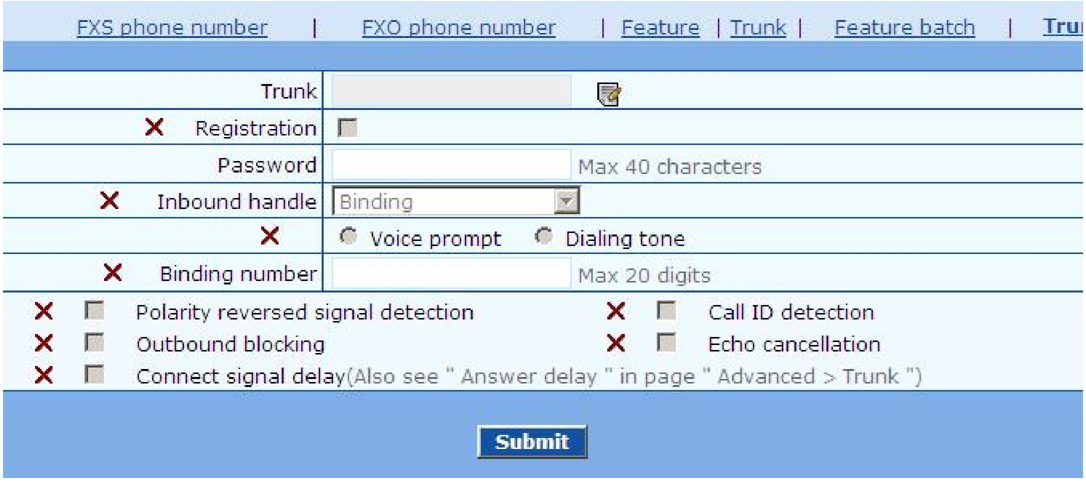


Click to choose whether to activate this function to configurate this parameter. Seen in "Subscriber Line Features".

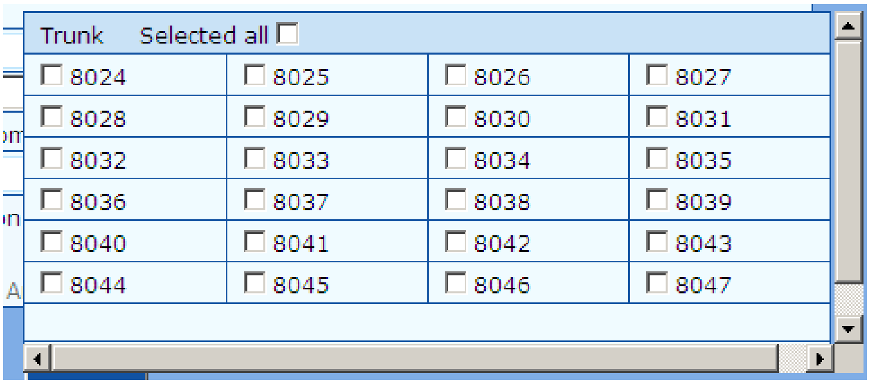
2.5.6 Trunk Batch

After login, click “Line > Trunk Batch” to open this interface.

Figure 2-14 Trunk Batch configuration interface



Click , the following interface is shown. Choose batch configured trunks and click “ok”



Click to choose whether to activate this function to configurate this parameter. Seen in “Trunk Line Features”.

2.6 **Advanced Configuration**

2.6.1 System

After login, click the label of “Advanced > System” to open this interface.

Figure 2-15 Inferface of system advanced configuraiton

Table 2-16 Parameters of system advanced configuration

Title	Explanation
NAT	
NAT traversal	Gateways support several mechanisms for NAT traversal. Usually, static NAT is used when fixed public IP address is available. It's necessary to perform port mapping or DMZ function on router when choosing dynamic or static NAT.
Refresh period	The refresh time must be filled in here when choosing dynamic NAT or STUN traversal. Besides, refresh time interval shall be determined by giving consideration into the NAT refresh time of the LAN router which the gateway is located. Gateway's NAT holding function and STUN function will carry out periodically operation accoding to this parameter. With second as its unit, default value of 60 seconds.
SDP Address	This parameter determines the IP address used in transmitted SDP. ı WAN IP Address: Apply NAT address into the transmitted SDP; ı Local IP Address: Apply the gateway's IP address into the transmitted SDP. Note: The parameter should come into effect only on condition that gateway successfully obtained NAT address.
NAT IP address	This parameter must be filled when using static NAT traversal, in which gateway works under LAN and the WAN address is fixed. The WAN address should be filled in this field, which will be used in SDP. This parameter can be set in IP address format or hostname format (note: DNS service should be activated when hostname format is used). There is no default value for this field.
STUN server	Set the IP or domain name of STUN server. No default value. If the set is empty, the gateway will adopt the STUN server address configured at factory. When choosing STUN for NAT traversal, the gateway will carry out STUN operation periodically according to the configured interval time of NAT refresh.
RTP Receiving Port	The gateways will send the RTP receiving port selected here to the remote side. ı NAT port: Use NAT mapped port， which is obtained through STUN, for example; ı Local port: Use local SIP and RTP port.

Title	Explanation
Remote management	
Remote management	The gateways support EMS which is a centralized gateway management server provided by Real Tone, and Auto-provision.
EMS	
Server URL	User needs to enter the IP address and port of EMS server for activating EMS service.
Auto provision	
Server URL	Gateways may download software upgrade packages and configuration files automatically through auto-provision server. Once the auto provision is selected, you have to enter the IP address of ACS here.

2.6.2 Media Stream

After login, click the label of “Advanced > Media Stream” to open this interface.

Figure 2-16 Media stream configuration interface

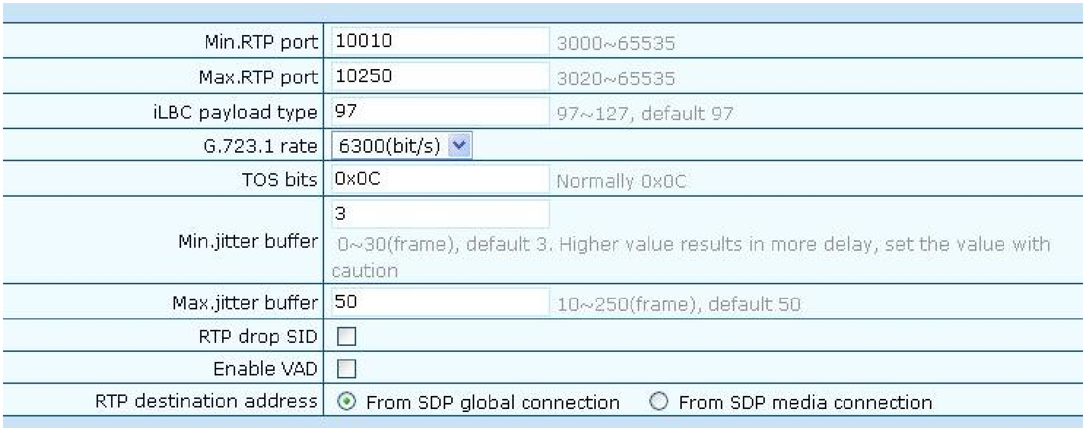


Table 2-17 Media stream configuration parameter

Title	Explanation
Min. RTP port	The minimum value of UDP ports for RTP transmission and receiving, and the parameter must be greater than or equal to 3000. This field must be filled in. Note: each phone call will occupy RTP and RTCP ports. If the gateway is equipped with 4 subscriber lines (or trunk line), then at least 8 UDP ports are needed.
Max. RTP port	The maximum values of UDP ports for RTP’s transmission and receiving. This field must be filled in. It’s advisable to be greater than or equal to “2× number of lines ＋min. RPT port”.
iLBC payload type	Set the RTP payload type of iLBC，and the default value is 97. Accepted value is 97 ~ 127. The parameter shall be configured in conformity to that of platform.

Title	Explanation
G.723.1 rate	Set G.723.1 coding rate，the default value is 6300. The optional parameters are followings: ı 5300: the Bit rate is 5.3k per second; ı 6300: the Bit rate is 6.3k per second
TOS bits	This parameter specifies the quality assurance of services with different priorities. The default value is 0x00. Eg: TOS=0xB8 indicates level 5 that has no reliability requirement.
Min. Jitter buffer	RTP Jitter Buffer is constructed to reduce the influence brought by network jitter. This default value is 3.
Max. Jitter buffer	RTP Jitter Buffer helps to reduce the influence brought by network jitter. The default value is 50.
RTP drop SID	Determine whether to discard received RTP SID voice packets. By default, SID voice packets will not be dropped. Note: RTP SID packets should be dropped only when they are in unconformity to the specifications. Nonstandard RTP SID data could generate noise for calls.
Enable VAD	Only applicable to G.723, GSM, iLBC. In case of selecting this parameter, it will not send any voice packet during mute period. By default, this is selected.
RTP destination address	This parameter determines where to obtain the IP address of the receiving side for RTP packets. By default, the IP address is obtained "From SDP global connection". ı From SDP global connection: Obtain the IP address from SDP global connection; ı From SDP media connection: Obtain the IP address from SDP Media Description.

2.6.3 SIP related configuration

The SIP messages consist of request message and response message. Both include SIP message header field and SIP message body field. SIP message header maily describes the message sender and receiver; SIP message body mainly describes the specific implementation method of the dialog.

Message of request: the SIP message sent by a client to the server, for the purpose of activating the given operation, including INVITE, ACK, BYE, CANCEL, OPTION and UPDATE etc.

Message of response: the SIP message sent by a server to the client as response to the request, including 1xx, 2xx, 3xx, 4xx, 5xx, and 6xx responses.

Message header: Call-id.

Parameter line: Via, From, To, Contact, Csq, Content-length, Max-forward, Content-type, White Space, and SDP etc.

WSS gateways provide good flexibility in content setting in order to improve the compatibility with the platform.

After login, click the label of "Advanced > SIP" to open this interface.

Figure 2-17 SIP related configuration interface

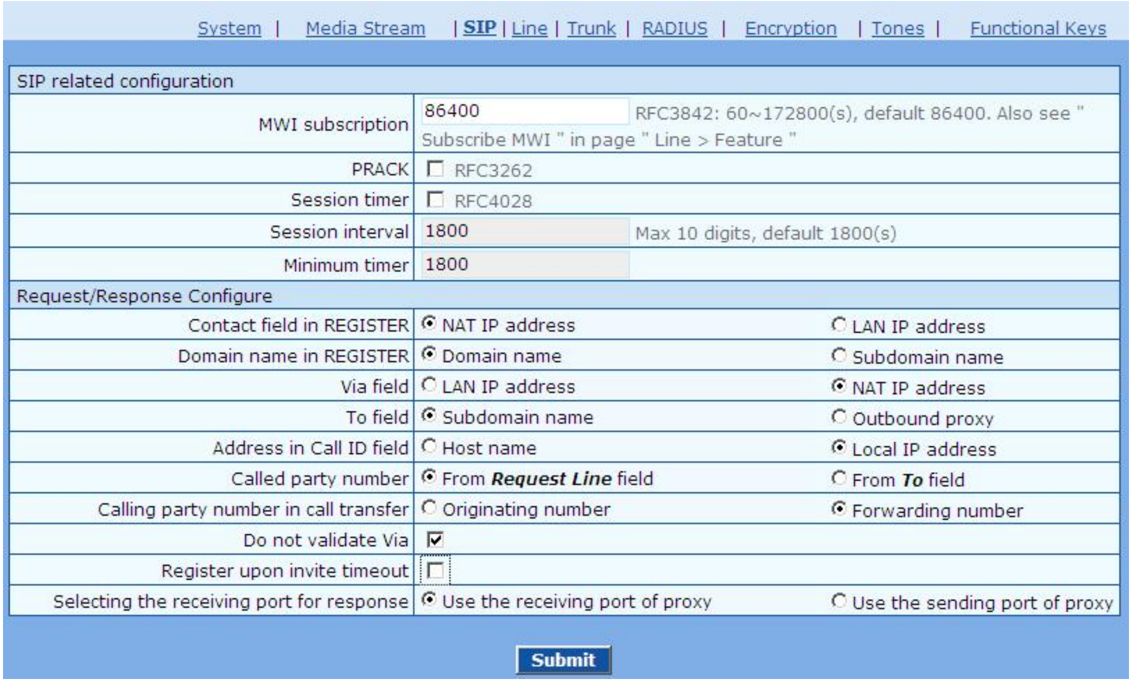


Table 2-18 SIP related configuration parameter

Title	Explanation
SIP related configuration	
MWI subscription	The default is 86400 seconds. The gateway will send platform a message to confirm that has subscribed MWI service at intervals of the time period set here. This parameter should be used in conjuection with voice mail subscription on the page of subscriber line.
PRACK	Determine whether to activate Reliable Provisional Responses. (RFC 3262)
Session timer	Choose to activate session refresh (Session Timer, RFC 4028). By default, session timer is not activated.
Session interval	Set the session refresh interval, the gateway will enclose the value of Session-Expires into INVITE or UPDATE messages. Default value is 1800 in second.
Minimum timer	Set the minimum value of session refresh interval.
Request/Response Configure	
Contact field in REGISTER	Choose the registration mode of gateway under LAN traversal circumstance, the default is “NAT IP Address”. ı LAN IP address: Keep original content of “Contact” when register; ı NAT IP address: Use the NAT information returned by registration server.

Title	Explanation
Domain name in REGISTER	The default is “Domain name”. ı Domain name: Complete domain name used for registration (for example: 8801@registrar.RealTone.com); ı Subdomain name: Only use the common part of the name of domain (for example: 8801@RealTone.com).
Via field	Choose whether to use NAT IP address or LAN IP address for “Via” header field value, the default is “NAT IP address”.
To field	Choose whether to apply Domain name or Outbound proxy to “To” header field, the default is “Domain name”.
Call ID field	Choose whether to fill Call ID field with host name or local IP, the default is “local IP address”.
Called party number	Choose whether the gateway acquires the called number from Request Line header field or To header field. The default is “from Request Line”.
Calling party number in call transfer	Under call forwarding, the calling party number sent can be choosed from Originating number or Forwarding number being set for sending, the default is “Forwarding number”. For example: the subscriber line 2551111 on the gateway activates call forwarding feature and set the destination to 3224422. When caller with 13055553333 calls 2551111, the call will be forwarded to 3224422: ı if choose“Originating number”，the number 13055553333 will be sent to 3224422 as calling party number； ı if choose“Forwarding number”，the number 2551111 will be sent to 3224422 as calling party number；
Do not validate Via	Set whether to ignore Via field, By default, Via is ignored.
Register upon INVITE timeout	Set whether to activate registration when SIP message of INVITE is failed or time expired, and by default, re-registration is not selected.
Selecting the receiving port for response	Use the receiving port of proxy or use the sending port of proxy

2.6.4 Characteristics of subscriber line

After login, click the label of “Advanced > line” to open this interface.

Figure 2-18 Subscriber-line characteristics configuration interface

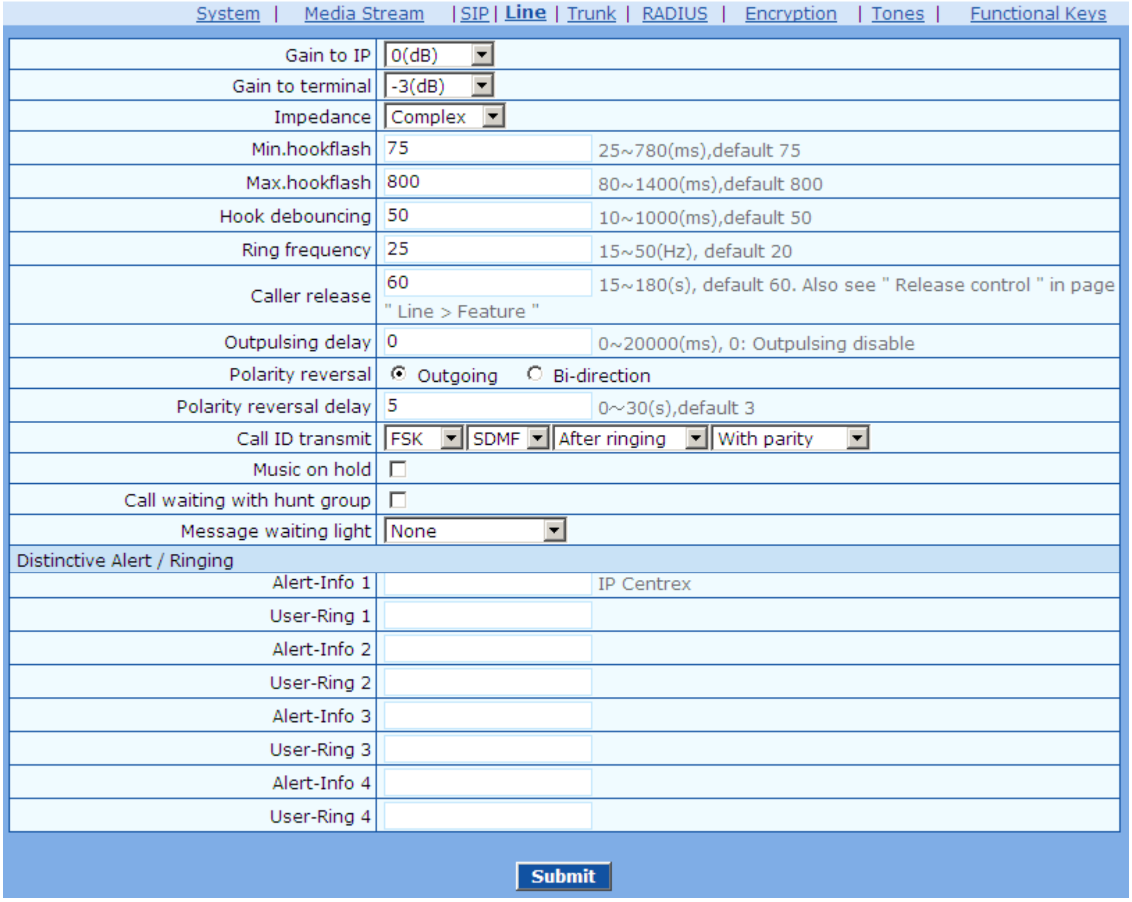


Table 2-19 Subscriber-line characteristics configuration parameter

Title	Explanation
Gain to IP	Set the voice volume gain towarding IP side, the default is 0. Taking decibel as the unit, setting range is -3 ~ +3 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Gain to terminal	Set the voice volume gain towarding FXS port side, the default is -3. Taking decibel as the unit, setting range is -6 ~ +3 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Impedance	Select the parameter of FXS port line impedance, and the default value is 600 ohm. The optional values as below: ı Complex ı 600（ohm） ı 900（ohm）
Min.hookflash	Used by gateway to detect Hook Flash event, the default is 75 milliseconds. The gateway will ignore any flash that fall short of the shortest flash time. Generally, this value should not be less than 75 milliseconds.

Title	Explanation
Max.hookflash	Used by gateway to detect hook flash, the default is 800 milliseconds. The gateway will regard the flash duration between “Min.hookflash” and “Max.hookflash” as effective flash. Any flash lasting over the longest time will be considered by gateway as hang up. Generally, this value should not be less than 800 milliseconds.
Hook debouncing	Used by gateway to avoid the glitch of the phone status, with default of 50 milliseconds. When the duration from hang-up to off-hook falls short of this value, the gateway will ignore the status variation, and consider the phone remains hang-up status. In case of vice versa, the gateway will ignore the status variation, and consider the phone remains off hook status. Effective range of setting is 10~1000 milliseconds.
Ring frequency	Set the ringing frequency to be transmitted by gateway to the phone, ranging from 15 to 50 Hz, with default of 20 Hz.
Caller release	Set the delay release time of line as caller control method, with default of 60 seconds. Effective range of setting is 15~180 seconds.
Outpulsing delay	Used when gateways’ FXS port is connected with the trunk interface of PBXs. For calls from gateway to PBX, gateways will relay the extensions to PBX after the delay set here. Setting of “0” means no extension number relay.The default is 0 millisecond.
Polarity reversal	Set the trigger for polarity reversal the default is “Outgoing”. l Outgoing:Transmit reverse polarity signal only when the outbound is connected; l Bi-direction: Transmit reverse polarity signal for the connection of both inbound and out bound calls.
Polarity reversal delay	The delay time from call being answereed to the transmission of reverse polarity signa. The default value is 3 in seconds. Effective range of setting is 0 ~ 30 seconds.
Call ID transmit	Select transmission mode of Caller ID signal from the FXS port to the phone. l FSK or DTMF; l SDMF or MDMF; l Sending Caller ID data before or after ringing; l Sending Caller ID data with or without parity.
Music on hold	Choose whether to play the background music while call waiting, and the default is not to play.
Call waiting with hunt group	Choose whether to activate hunt group feature for call waiting, Default not selected.
Message waiting light	Choose the lighting method of message waiting indicator of voice mail here: None, Polarity reversed, FSK. Message waiting indicator refers to the special LED on a phone, working with voice mail function. When user gets the latest mail, the gateway will light this lamp upon receiving the notice from platform; the light goes off when there’s no unheard mail. It’s essential to understand whether the phone supports the indicators and lighting method when selecting the lighting method.
Distinctive ring	Apply to enterprise customers
Alert info 1	To match with “User ring 1”. Four patterns of user ring are offered. When the Alert-info value of INVITE message matches with this parameter, “User ring 1” is activated.

Title	Explanation
User ring 1	Configure user ring 1. Eg 1: if the user ring is set “2,500,500,1000,3000”, the ringing effection will display as 0.5s ringing, 0.5s pause; 1s ringing, 3s pause. Eg 2: if the user ring is set “2000,4000”, the ringing effection will display as 2s ringing, 4s pause.
Alert info 2	To match with “user ring 2”
User ring 2	Configure user ring 2
Alert info 3	To match with “user ring 3”
User ring 3	Configure user ring 3
Alert info 4	To match with “user ring 4”
User ring 4	Configure user ring 4

2.6.5 Characteristics of trunk line

After login, click the label of “Advanced > trunk” to open this interface.

Figure 2-19 Trunk line characteristics configuraiton interface

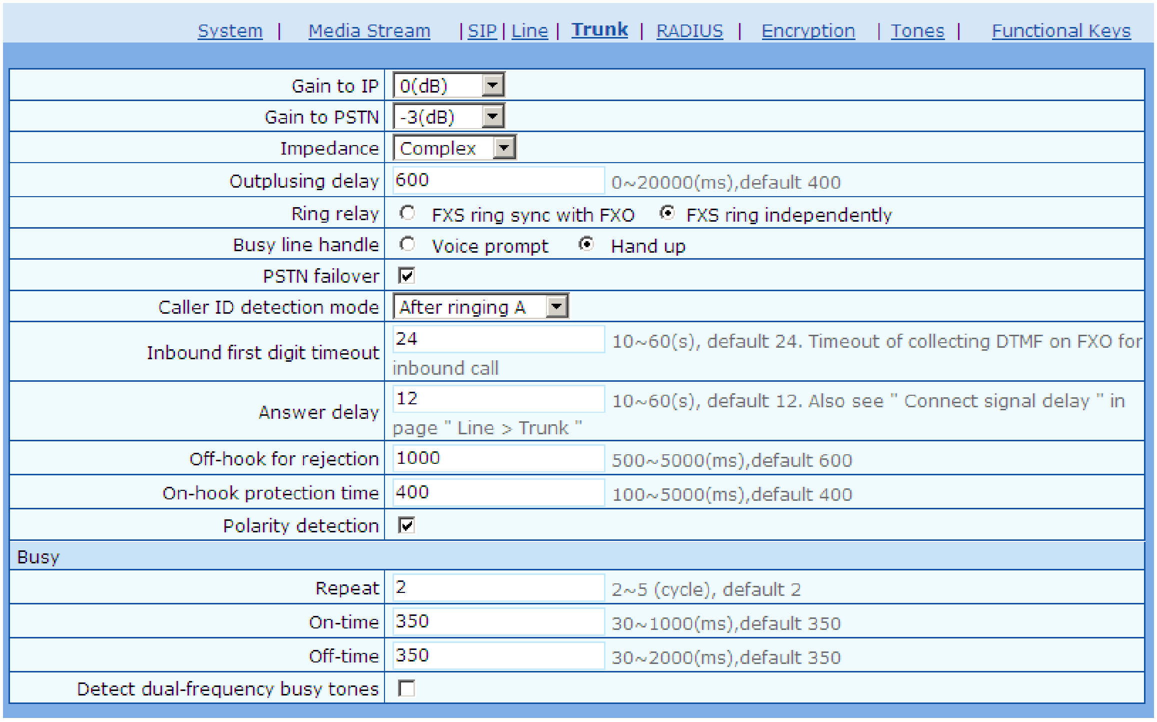


Table 2-20 Configuration parameter of trunk line characteristics

Title	Explanation
Gain to IP	Set the voice volumn gain towarding IP side, the default is 0. Taking decibel as the unit, setting range is -3 ~ +9 decibels. -3 means declining of 3 decibels; +3 denotes the amplification of 3 decibels.
Gain to PSTN	Set the voice volumn gain towarding PSTN side, the default is -3. Taking decibel as the unit, setting range is -6 ~ +9 decibels.

Title	Explanation
Impedance	Set the parameter of FXO line impedance, with the default of 600 ohm. The optional settings as below: ı Complex ı 600（ohm） ı 900（ohm）
Outplusing delay	Set the time interval between FXO going off-hook and starting outpulsing the first digit to PSTN. The default is 400 in milliseconds.
Ring relay	Whether to relay the ring of inbound call to the FXS port when applying to DID. The default is “FXS ring independently”.
Busy line handle	Either a voice prompt or hanging up can be applied to FXO port when an incoming call goes to the FXS port which is in busy. This only applies to DID feature.
PSTN failover	Whether to route a call to PSTN through FXO port when the IP network faults or no response to the call request. Default selected.
Caller ID detection mode.	ı After ring A； ı After ring B； ı Before ring A； ； ı Before ring B；
Inbound first digit timeout	Set the timeout of calling DTMF on FXO port for inbound calls, ranging from 10-60 seconds, with default of 24 seconds.
Answer delay	Set the delay time of outbound connection ranging from 10-60 seconds, with default of 12 seconds. Working with “Line >Trunk” interface and “Connect signal delay” configuration.
Off-hook for rejection	Used for binding a FXO port with a FXS port. For inbound calls to a FXO port, if the FXS port which binging with the FXO port is in the state of busy line, the gateway will hang up after hook off according to the time set by the parameter, so as to refuse the upcoming call. The duration of off hook is 500~5000 milliseconds, with default of 600 milliseconds.
On-hook protection time	Protection period following hang up of FXO port. During this period, gateway ignores any voltage variation of line. Value range is 100~5000 milliseconds, the default is 400 in milliseconds.
Polarity detection.	Choose whether to activate the detection of reverse polarity signal of FXO port inlet. Note the detection will work only when the trunk supports polarity reversal.
Busy Detection	
Repeat	Gateways will regard the busy tone signal with the repeat times specified here as hang-up signal. Default is 2, effective range is 2 ~ 50.
On-time	Set duration of busy tone signal, the default is 350 in milliseconds.
Off-time	Set the interval time of busy tone, the default is 350 in milliseconds.

2.6.6 Radius call logs

After login, click the label of “Advanced > RADIUS” to open this interface.

Figure 2-20 Configuration interface of Radius call logs

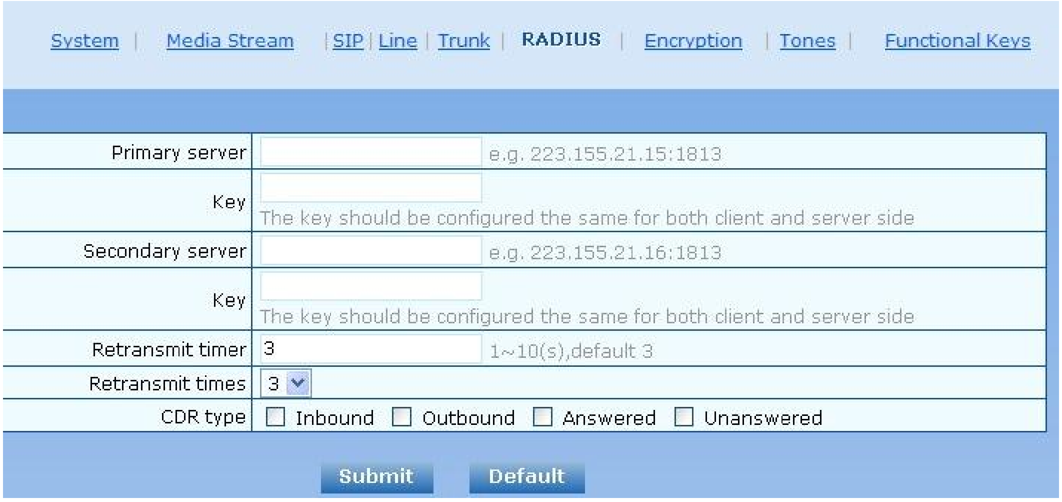


Table 2-21 Configuration parameter of Radius call logs

Title	Explanation
Primary server	Set IP address and port number of preferred Radius server. Note: if the port number is not configured yet, please use Radius default port number of 1813.
Key	Set the share key to be used for encrypted communications between Radius client and server. Note: the share key should be configured the same for both client and server side
Secondary server	Set the IP address and port number of standby Radius server. When the fault appears in communications between gateway and preferred Radius server, the gateway will automatically activate standby Radius server. Note: in case of no configuration of port number, use default port number of 1813.
Key	The share key for communications between Radius client and standby Radius server. Note: the key should be configured the same for both client and server side
Retransmit timer	Set the amount of overtime on response after transmission of Radius message, the default is 3 seconds. The retransmission will be performed If no response is given after the timeout.
Retransmit times	Set the times of retransmission of Radius message when no response is received default is 3 times.
CDR type	ı Outbound: Set whether to send RADIUS charge message for outbound calls; ı Inbound: Set whether to send RADIUS charge message for inbound calls; ı Answered: Set whether to send RADIUS charge message when calls are connected; ı Unanswered: Set whether to send RADIUS charge message for unanswered calls.

2.6.7 Encryption

After login, click the label of “Advanced > Encryption” to open this interface.

Figure 2-21 Encryption configuration interface

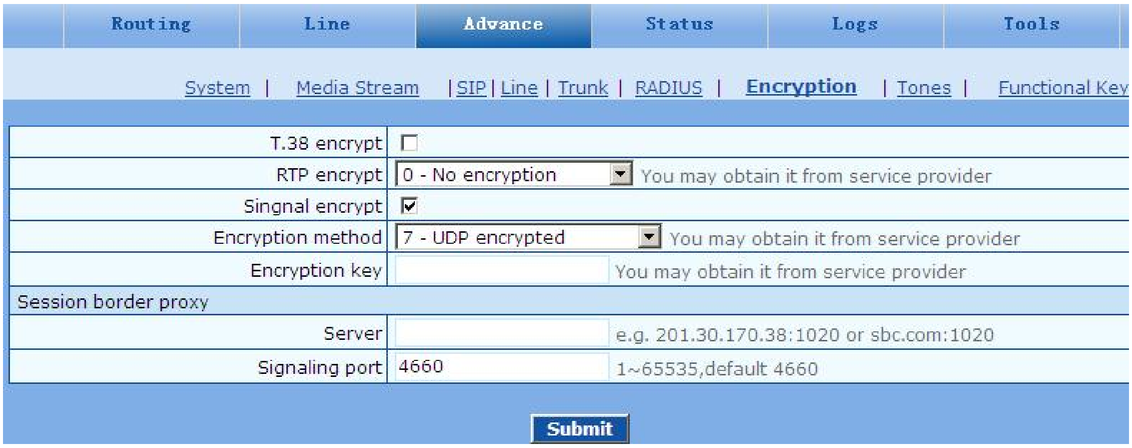


Table 2-22 Encryption configuration parameters

Title	Explanation
Singnaling encrypt	Choose whether to encrypt signaling. By default, this is not selected.
T.38 encrypt	Choose whether to encrypt T38 data. By default, this is not selected.
RTP encrypt	Choose whether to encrypt RTP voice pack, the default is “0” ı 0: No encryption; ı 1: Entire message encryption; ı 2: only encrypt RTP header; ı 3: only encrypt RTP body;
Encryption mode	Set the gateway encryption method, default is 7. The optional parameters as below: ı 2: TCP Not Encrypted; ı 3: TCP Encrypted; ı 6: UDP Not Encrypted; ı 7: UDP Encrypted（Real Tone）; ı 8: Using key words, coordinate with platform; ı 10: RC4; ı 13: Encrypt13, coordinate with platform; ı 14: Encrypt14（Real Tone）; ı 16: Word Reverse（263）; ı 17: Word Exchange（263）; ı 18: Byte Reverse（263）; ı 19: Byte Exchange（263）; ı 20: Word Exchange (VOS).
Encryption key	You may obtain it from service provider

Title	Explanation
Session Border Proxy	
Server	Set the IP address and port number of session border proxy server. The character of “:” must be used between IP address and port number. Server address could be set into IP address or domain name. When domain name is used, “DNS service” must be activated as shown in the page of “configure network parameter”, and “DNS server” must be configured. Example: “201.30.170.38:5060” and “softswitch.com:5060”
Signaling port	Signaling port value of the gateway, the default value is 4660. Signaling port number could be set at will, but can not conflict with other ports of equipment.

2.6.8 Call progress tone plan

After login, click the label of “Advanced > Tones” to open this interface.

Figure 2-22 Call progress tone configuration interface

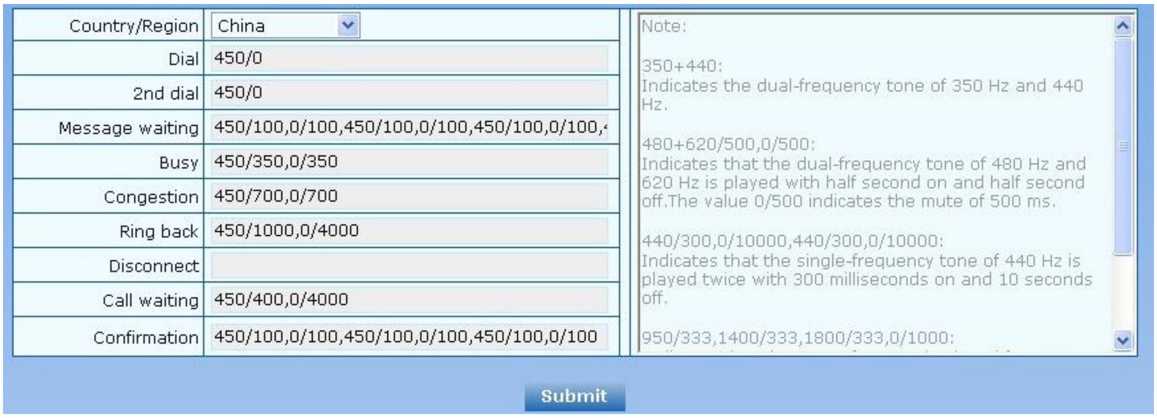


Table 2-23 Call progress tone configuration parameters

Title	Explanation
Country/region	There are progress tone plans for several countries and regions which are pre-programmed in gateways. Users may also specify the tone plan according to the national standard. Gateways provide tone plans for the following countries and regions: China; the United States; France; Italy; Germany; Mexico; Chile; Russia; Japan; South Korea; Hong Kong; Taiwan; India; Sudan; Iran; Algeria; Pakistan; Philippines; Kazakhstan;
Dial	Prompt tone of off-hook dialup
2nd dial	Used for the second stage dialup
Message waiting	Used for prompt of voice mail, or when the subscriber line is set with “Don’t Disturb Service and Call Transfer”.
Busy	Used for busy line prompt
Congestion	Used for notification of call set up failure due to resource limit
Ring back	The prompt tone sent to caller when ring
Disconnect	Used for reminding the subscriber of off-hook and no dialup status of the phone

Title	Explanation
Call waiting	Used for notification in call waiting
Confirmation	Used for confirming function keys being entered.

Here are examples which illustrate the rules of defining call progress tone.

- 350+440

  Indicates the dual–frequency tone consisting of 350 and 440 Hz
- 480+620/500,0/500

  Indicates the dual–frequency tone consisting of 480 and 620 Hz, repeated playing with 500 milliseconds on and 500 milliseconds off. Note: 0/500 indicates 500 milliseconds mute.
- 440/300,0/10000,440/300,0/10000

  Indicates 440 Hz single frequency tone, repeated twice in terms of 300 milliseconds on and 10 seconds off.
- 950/333,1400/333,1800/333,0/1000

  Indicates repeated playing 333 milliseconds of 950 Hz, 333 milliseconds of 1400 Hz, 333 milliseconds of 1800 Hz, and mute of 1 second

2.6.9 Functional keys

The function key consists of system function key and service function key. The system function key is used for acquiring gateway information, and the later is used for users to activate and inactivate supplementary services.

After login, click the label of “Advanced > Functional Keys” to open this interface.

The following are the examples of the dialing rule for the function key:

a) Using *xx (dial * and 2 digits number) to activate a service;

b) Using #xx (dial # and 2 digits number) to cancel a service.

Illustrate with following defaults of various parameters, which may be modified according to requirements.

Figure 2-23 Functional keys configuration interface

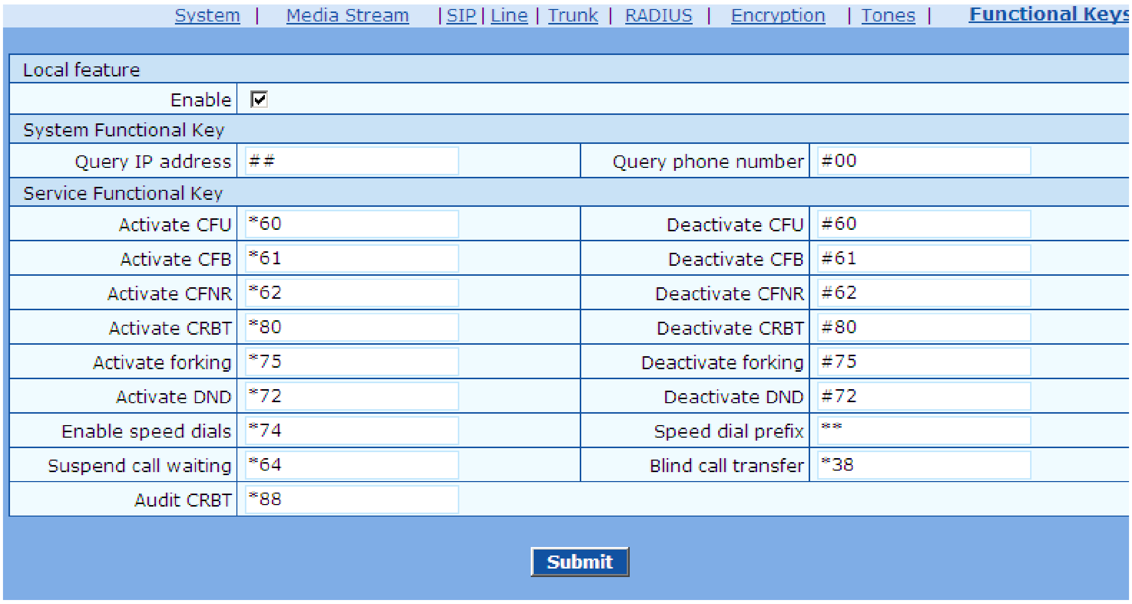


Table 2-24 Functional keys configuration parameter

Title	Explanation
Signaling functional keys	
Activate	Activate: subscriber line number matches with functional keys listed on this page, applied to local equipment Non: all dialed functional keys are sent to proxy server.
System Functional Key	
Query IP address	The function key for inquiring the IP address of gateway, with default of ##. Dialing this key, users can hear gateway broadcasting IP address and system software version number. Narrative: if the gateway is only equipped with FXO port, connect FXO port through PBX extension line or PSTN direct line, and dial the number of this line accordingly, press “##” immediately after hearing the second dial tone, users may thus hear IP address and system software version number of the gateway.
Query phone number	The function key for inquiring the phone number of this subscriber line, with default of #00. Dialing this key may hear the phone number of the subscriber line broadcasted by gateway.
Service Functional Key	
Activate CFU	The function key for activating unconditional call forwarding, with default of *60. Dialing this key may activate unconditional call forward of the line, and set the destination number for call forwarding. User operation: Off hook → press *60 →enter the destination number. Users can inquire the latest destination number set by dialing “ *60* ”. Note: it’s required to enable call forwarding service before using this function (please see the instructions on relevant configuration of “subscriber line”).
Deactivate CFU	The function key for deactivating unconditional call forwarding, with default of #60. User operation: Off hook → press #60 → hang up.
Activate CFB	The function key for activating call forwarding on busy， with default of *61. Dialing this key may activate CFB, and specify the destination number. Note: it’s required to enable call forwarding on busy service before using this function (please see the instructions on relevant configuration of “subscriber line”).
Deactivate CFB	The function key for deactivating call forwarding on busy, with default of #61. User operation: Off hook → press #61 → hang up.
Activate CFNR	The function key for activating call forwarding on no answer, with default of *62. Dialing the function key may activate call forwarding on no answer and specify destination number. Note: it’s required to enable call forwarding on no answer ervice before using this function (please see the instructions on relevant configuration of “subscriber line”).
Deactivate CFNR	The function key for deactivating call forwarding on no answer, with default of #62.

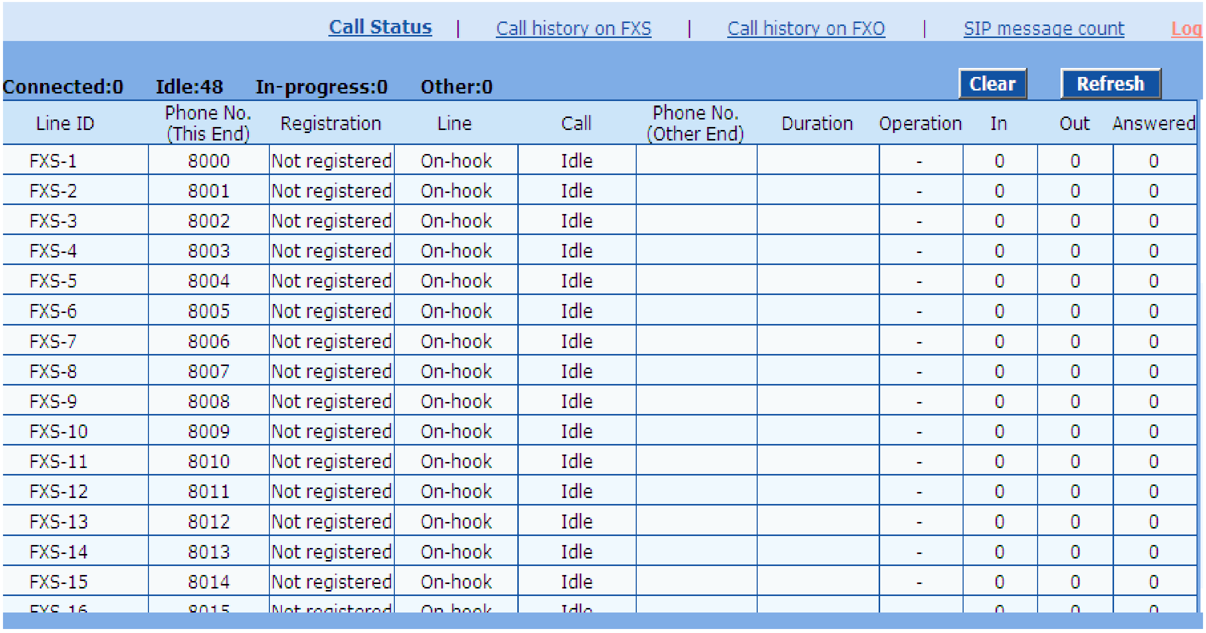
Title	Explanation
Activate CRBT	The function key for activating color ring, with default of *80. The subscribers may select their favorite color rings by using the key. Note: it's required to start color ring service before using this function (please see the instructions on relevant configuration of "subscriber line"). User operation: Upon off hook, the subscriber may press the function key (like *80), then, input two digit index numbers of color ring; "*80* " is used for hearing and inquiring the color ring that have been set already.
Deactivate CRBT	The function key for deactivating the color ring, with default of #80. The subscriber may use such key to recover the normal ring of phone. User operation: Off hook → press #80 → hang up.
Activate forking	The function key for activating double ringing feature, with default of *75.
Deactivate forking	The function key for deactivating the feature, with default of #75.
Activate DND	Activating "Don't Disturb Service", with default of *72. After dialing up, the gateway will reject all coming calls by sending busy tone to the caller. Note: it's required to start "Don't Disturb Service" before using this function (please see the instructions on relevant configuration of "subscriber line").
Deactivate DND	The function key to cancel "Don't Disturb Service", with default of #72. Dialing the function key may recover normal ringing upon the arrival of incoming calls.
Enable speed dials	Define the function key of dial, with default of *74. Dialing of this function key may build a table of 2 digits（20~49）of abbreviated numbers, which corresponding to the real numbers. Note: It's necessary to get the dial-up service under way before applying this function (please refers to the instructions about "subscriber line". User operation: Upon dialing the function key (such as " *74 ") set hereof, the subscriber may save the corresponding relationship into gateway following dialing 2 digits of abbreviated number and corresponding number with # as ending..
Speed dial prefix	The prefix number for applying abbreviated dialing, with default of "**". The said prefix should be added ahead of abbreviated dialing numbers when using abbreviated dialing. User operation: off hook → dial the prefix number of abbreviated dialing（**）and dial abbreviated dialing number（20）。
Audit CRBT	The function key for hearing the color ring， with default of *88. User operation: Off hook → press *88 → input color ring number.
Blind call transfer	Function key of blind call transfer， with default of *38. User operation: During the call, tap the phone hook switch or press R butto n→ dial *38→ dial the called number and then hang up.
Suspend call waiting	The function key for cancelling the call waiting for next call, with default of *64. Dialing this function key may temporarily shield the call waiting for next call, avoiding the possible intervention. Note: the function key works only for single cancel, if to cancel the call waiting completely, please refer to the instructions on relevant configuration of "subscriber line".

2.7 Call status and statistics

2.7.1 Call status

After login, click the label of “Status > Call Status” to open this interface.

Figure 2-24 Interface of call status



Call Status | Call history on FXS | Call history on FXO | SIP message count Log

Connected:0 Idle:48 In-progress:0 Other:0 Clear Refresh

Line ID	Phone No. (This End)	Registration	Line	Call	Phone No. (Other End)	Duration	Operation	In	Out	Answered
FXS-1	8000	Not registered	On-hook	Idle			-	0	0	0
FXS-2	8001	Not registered	On-hook	Idle			-	0	0	0
FXS-3	8002	Not registered	On-hook	Idle			-	0	0	0
FXS-4	8003	Not registered	On-hook	Idle			-	0	0	0
FXS-5	8004	Not registered	On-hook	Idle			-	0	0	0
FXS-6	8005	Not registered	On-hook	Idle			-	0	0	0
FXS-7	8006	Not registered	On-hook	Idle			-	0	0	0
FXS-8	8007	Not registered	On-hook	Idle			-	0	0	0
FXS-9	8008	Not registered	On-hook	Idle			-	0	0	0
FXS-10	8009	Not registered	On-hook	Idle			-	0	0	0
FXS-11	8010	Not registered	On-hook	Idle			-	0	0	0
FXS-12	8011	Not registered	On-hook	Idle			-	0	0	0
FXS-13	8012	Not registered	On-hook	Idle			-	0	0	0
FXS-14	8013	Not registered	On-hook	Idle			-	0	0	0
FXS-15	8014	Not registered	On-hook	Idle			-	0	0	0

Table 2-25 Parameters of call state

Title	Explanation
Line	There are six types of line statuses, including On-hook, Off-hook, Ringing, Maintenance, Disconnect, Parallel line in-use.
Call	The call state includes Idle, Ooutpusling, Ring, Entering number, In progress, Ring back, Talk, Near end hung up, Far end hung up, Timeout.

Click the label of “check detail” to open detail interface.

Figure 2-25 Details for the call

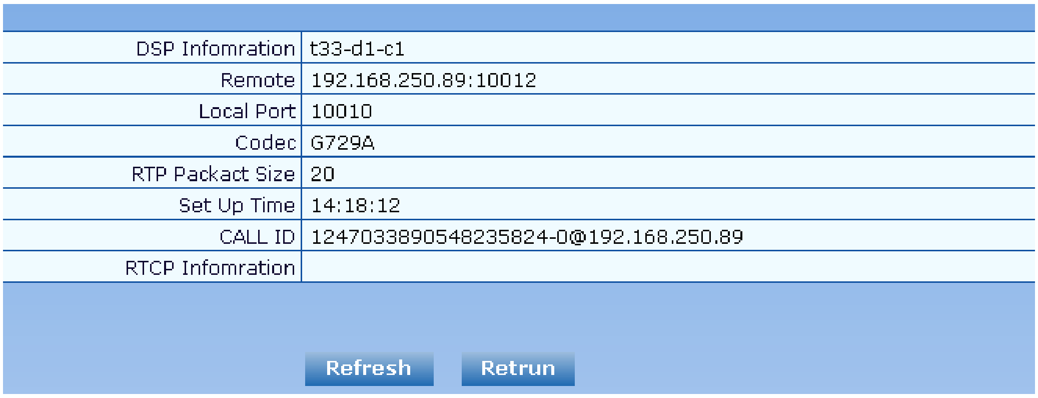


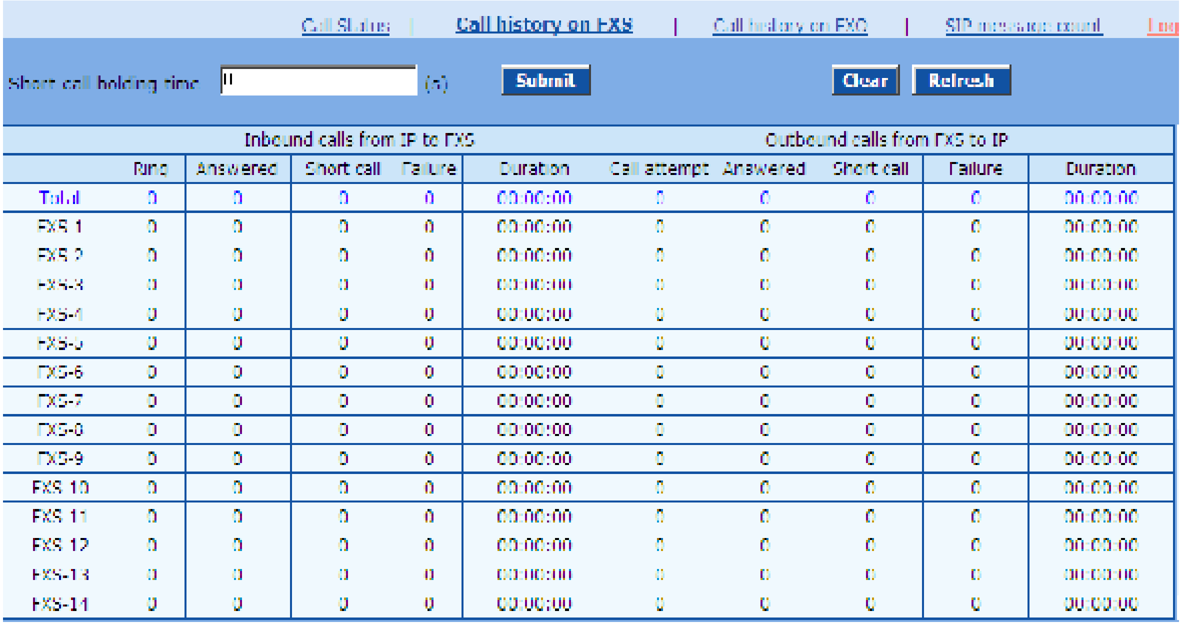
Table 2-26 Details for the call

Title	Explanation
DSP Information	This indicates the DSP chip information used for the call, in which “t” indicates time slot, “d” indicates the DSP chip, “c” refers to the channel on the chip.
Remote	The IP address of the equipment at the far end, followed with RTP port number.
Local port	Local RTP port number of the call.
Codec	The codec for this call.
RTP Packet Size （millisecond）	Packet length of the RTP of the call.
Set Up time	The time at which the call is answered.
CALL ID	Call ID in SIP message.
RTCP Information	The latest RTCP statistics report received by this call.

2.7.2 Call history on FXS

After login, click the label of “status > Call history on FXS” to open this interface.

Figure 2-26 Interface of call on FXS

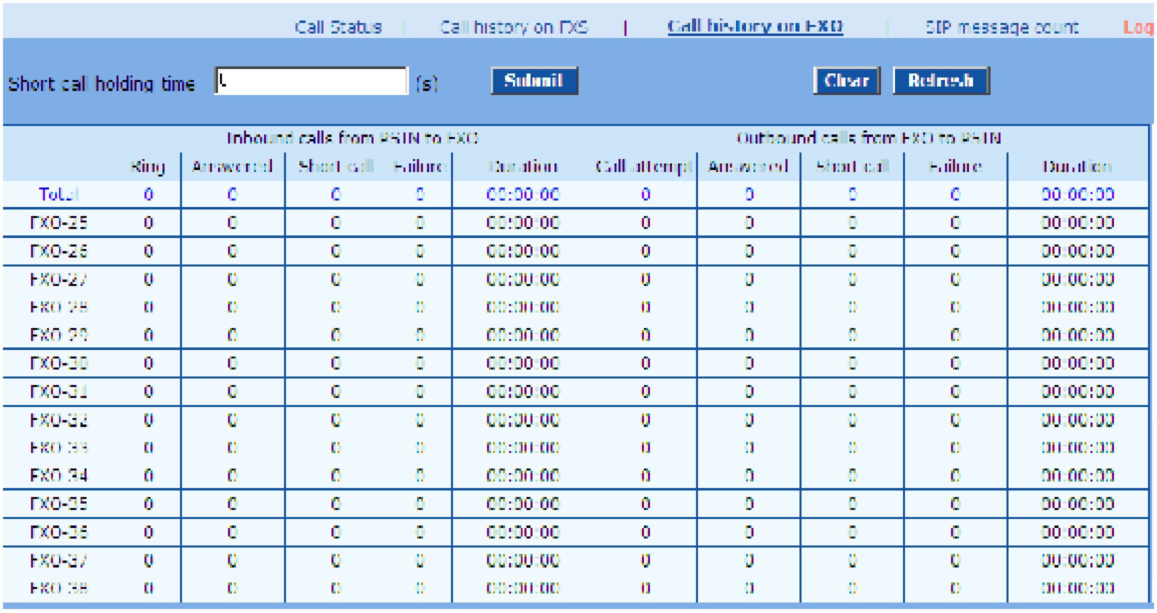


	Inbound calls from IP to FXS					Outbound calls from FXS to IP				
	Ring	Answered	Short call	Failure	Duration	Call attempt	Answered	Short call	Failure	Duration
Total	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-1	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-2	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-3	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-4	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-5	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-6	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-7	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-8	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-9	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-10	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-11	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-12	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-13	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXS-14	0	0	0	0	00:00:00	0	0	0	0	00:00:00

2.7.3 Call history on FXO

After login, click the label of "status > Call history on FXO" to open this interface.

Figure 2-27 Interface of call on FXO

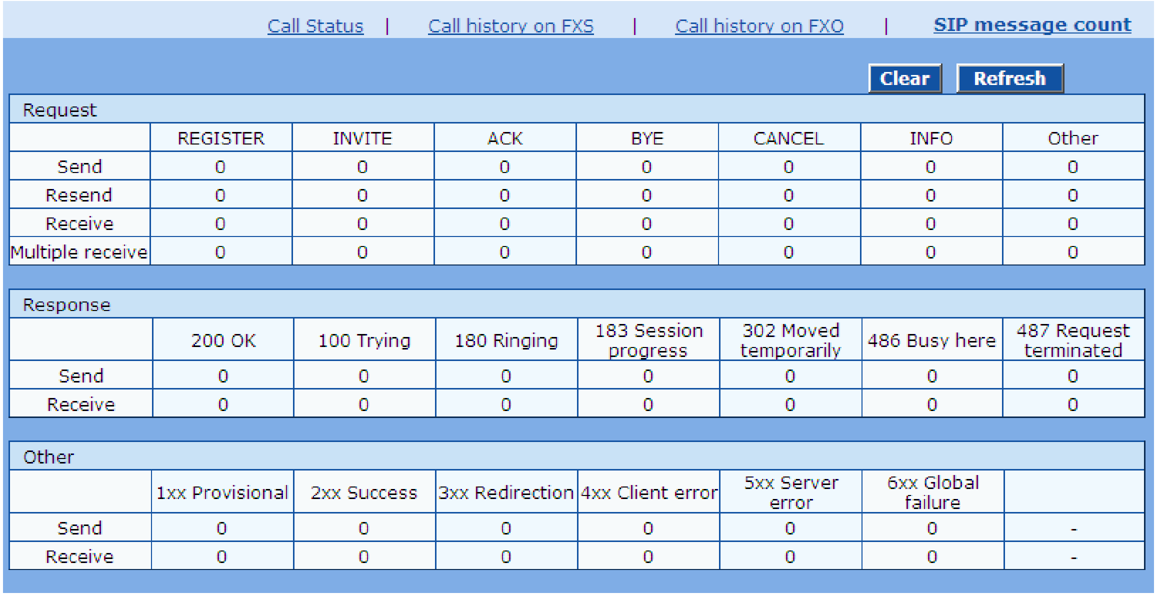


	Inbound calls from PSTN to FXO					Outbound calls from FXO to PSTN				
	Ring	Answered	Short call	Failure	Duration	Call attempt	Answered	Short call	Failure	Duration
Total	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-25	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-26	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-27	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-28	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-29	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-30	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-31	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-32	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-33	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-34	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-35	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-36	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-37	0	0	0	0	00:00:00	0	0	0	0	00:00:00
FXO-38	0	0	0	0	00:00:00	0	0	0	0	00:00:00

2.7.4 SIP message count

After login, click "status > SIP message count" to open this interface.

Figure 2-28 Figure 2-29 interface of SIP message count



Call Status | Call history on FXS | Call history on FXO | **SIP message count**

Clear Refresh

Request							
	REGISTER	INVITE	ACK	BYE	CANCEL	INFO	Other
Send	0	0	0	0	0	0	0
Resend	0	0	0	0	0	0	0
Receive	0	0	0	0	0	0	0
Multiple receive	0	0	0	0	0	0	0

Response							
	200 OK	100 Trying	180 Ringing	183 Session progress	302 Moved temporarily	486 Busy here	487 Request terminated
Send	0	0	0	0	0	0	0
Receive	0	0	0	0	0	0	0

Other							
	1xx Provisional	2xx Success	3xx Redirection	4xx Client error	5xx Server error	6xx Global failure	
Send	0	0	0	0	0	0	-
Receive	0	0	0	0	0	0	-

2.8 Log management

2.8.1 System status

Critical runtime information of gateways can be obtained in this interface, including:

1) The information about login interface (including IP address and jurisdiction of the user);
2) SIP registration status;
3) Call related signaling and media (RTP) information;

After login, click the label of "Logs > System Status" to open this interface.

Table 2-27 Parameters of system status

Title	Explanation

Login User Info	Show the IP address and jurisdiction of login user. The numbers following the IP address show the online jurisdiction of the user: 1- administrator; 2 - operator; 3 – viewer. The viewer can only read the configuration, but is not allowed to modify it.
	When more than one administrator log in at the same time, the first login's jurisdiction is 1, others are 3; also, when more than one operators log in at the same time, the first one's jurisdiction is 2, others are 3.
	For example:
	`Login User Info >>>>>`
	`1） 192.168.2.247  1`

SIP Registration Info	Show registration status:
	ı Not enabled: The registeration server's address is not entered yet;
	ı Latest response: The latest response message for the registration. 200 means registered successfully；
	ı No response: No response from registeration server. The cause may contribute to 1） incorrect address for the registration server; 2) IP network fault; or, 3) the registration server is not reachable.
	For example:
	`SIP Registration Info >>>>>`
	`  ---- Not enabled ----`
	`SIP Registration Info >>>>>`
	`Contact: <sip:2681403@220.218.77.70:1003；user=phone>`
	`       latest response: 200 （timeout-555）`
	`Contact: <sip:2681402@220.218.77.70:1003；user=phone>`
	`       latest response: 200 （timeout-555）`

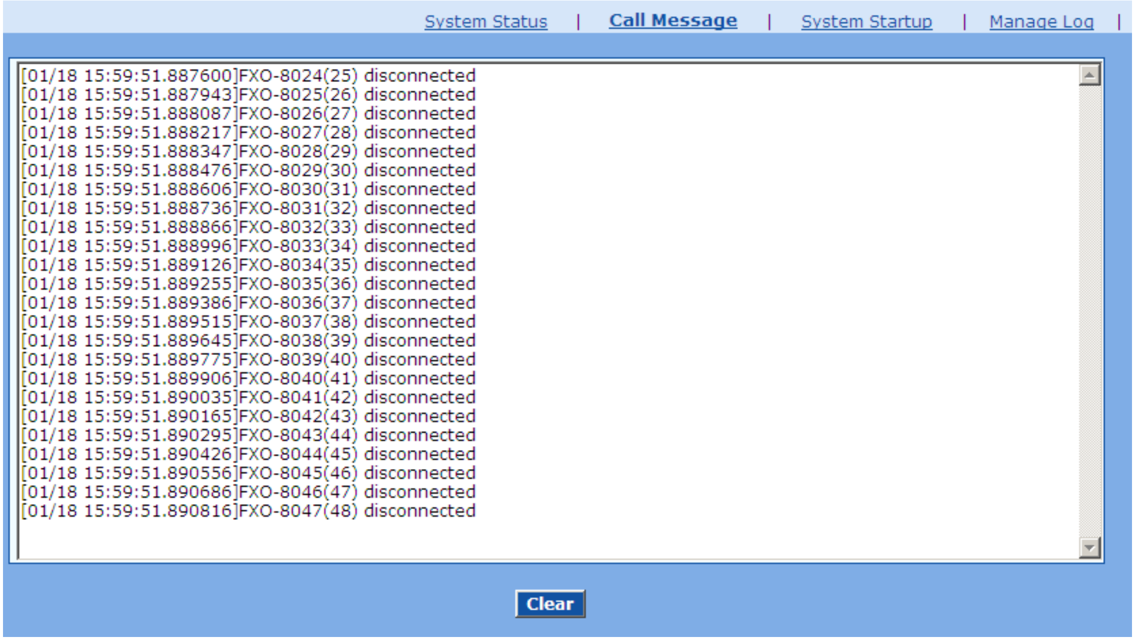
Call Context Info	Show the call status.

Rtp Context Info	Show the voice channel related to the calls.
	For example:
	`Rtp  Context Info >>>>>`
	`3） created， call =e011`

2.8.2 Call message

After login, click the label of "Logs > Call Message" to open this interface.

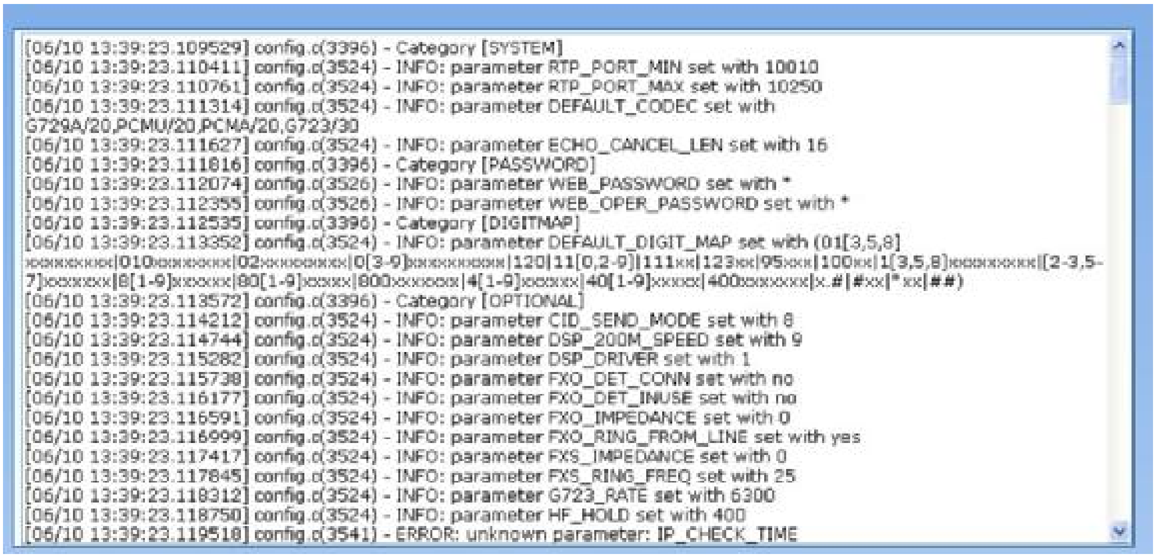
Figure 2-29 Call message interface



2.8.3 System Startup

After login, click the label of “Logs > System Startup” to open this interface. The gateway boot up information is available in this page, including the hardware configuration.

Figure 2-30 Interface of system startup



2.8.4 Manage log

After login, click the label of “Logs > Manage Log” to open this interface. Log files can be downloaded through this interface.

Figure 2-31 Interface of debugging log management

Table 2-28 Configuration parameters of debugging log management

Title	Explanation
Log download	See the description below.
System log server	Set the IP address of system log server.
Log server	IP address of debugging log server.
Log level	Select the log file level of gateway, default is 3. The setting range is 1 ~ 5, the higher the level goes, the more details the log file will be. Note: log level should be set to be 3 or lower when gateway is used in normal operation, avoiding influencing the system performance.

Procedure of downloading the debugging log:

Step 1: Click “download”, the gateway starts pack the logs.

Step 2: After few seconds, the interface of log saving will appear.

Step 3: click “Save”, and select path to save.

Step 4: The user may review the log from the server concerned.

WARNING

The procedure of downloading log files described hereof is only applicable to release 1.9.x.238 of WSS series or updated version of software.

2.9 System tool

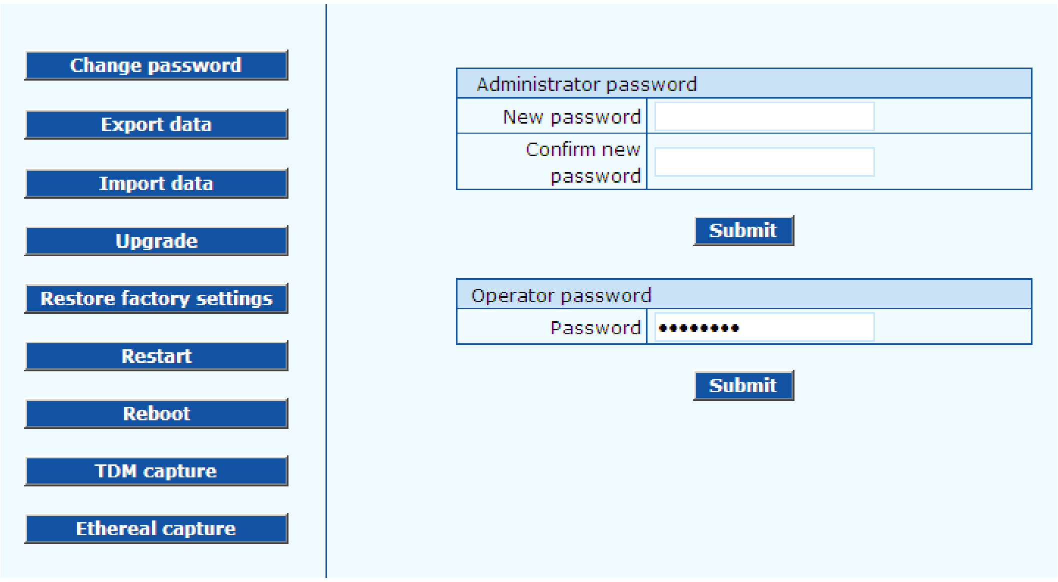
2.9.1 Change password

After login, click the label of “Tools” to open this interface. Only administrator is entitled to change the password of login.

For changing administrator password, it’s required to enter new password into “New password” field and “Confirm new password” field, then click “Submit”.

The password being used by operator will be displayed as hidden codes, which could be changed by administrator at any time. The administrator is allowed to change the operator’s password by entering new password into “Operator password>password”.

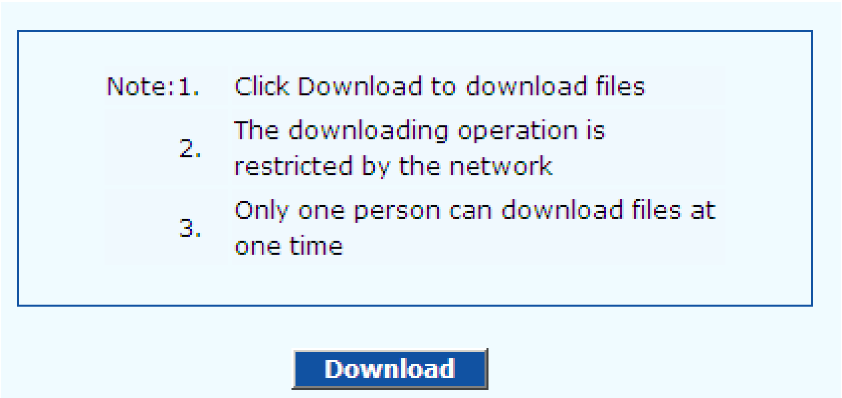
Figure 2-32 Interface of password changing



2.9.2 Configuration export

After login, click "Tools >Export of configuration" to open this interface. The downloading procedure is similar to the downloading procedure of log files..

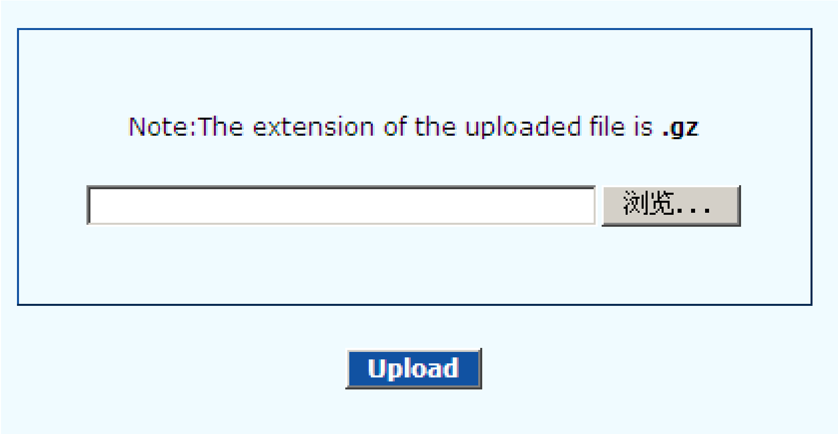
Figure 2-33 Interface of export data



2.9.3 Configuration import

After login, click "Tools>Import data" to open this interface. Operating procedure is the same as that of "software upgrade".

Figure 2-34 Interface of import data



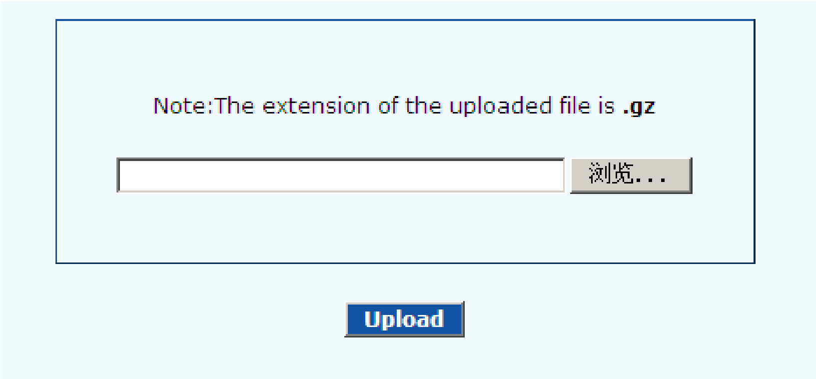
2.9.4 Software upgrade

After login, click “Tools > Upgrade” to open this interface. The software upgrading procedure is presented as below:

Step 1: Obtain the upgrade files (tar.gz file), and save the file onto a local computer.

Step 2: Click “System tool > software upgrade” to access to the page of software upgrade.

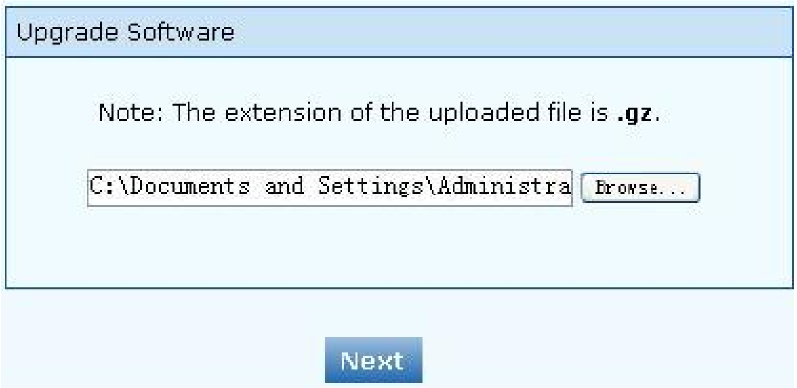
Figure 2-35 Interface of software upgrade



Step 3: Click “Browse” to select the upgrade files and click “Open”.

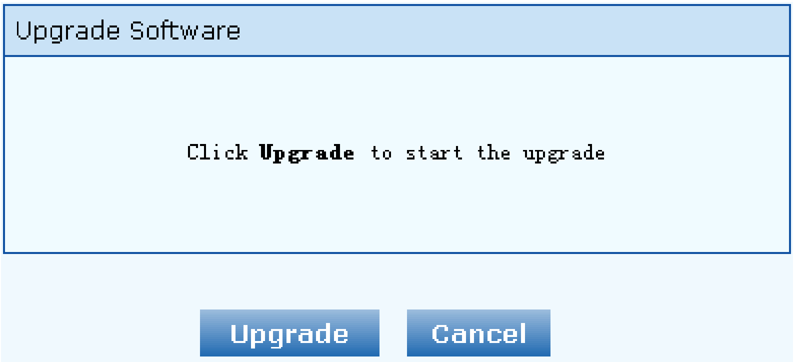
Step 4: Click “Next” when the following interface appears, and start uploading the upgrade files to the gateway.

Figure 2-36 Interface of file upload



Step 5: Uploading will be completed in about 30 seconds, and click “Upgrade” on following dialog.

Figure 2-37 Upgrade interface



Step 6: The following prompt appears during the upgrade.

Figure 2-38 Prompt of upgrade process

WARNING

A few minutes are needed to upgrade the gateway. Don’t operate the gateway during this period.

Step 7: After success in upgrade, the following dialog will appear, click “Confirm”.

Figure 2-39 Interface of successful upgrade

Step 8: The gateway is on the progress of reboot when the interface cannot be displayed.

Step 9: Wait for about 2 minutes, and access to the interface of gateway management system, click "Info" and check the software version.

WARNING

For WSS100 and WSS120 gateways, the software upgrade operation must be conducted on an 100M Ethernet port.

2.9.5 Restore factory settings

After login, click "Tools > Restore factory settings" to restore the parameters of gateway into the factory settings.

The factory settings are designed based on common applications, and therefore, no need to modify them in many deployment situations.

2.9.6 Software restart

After login, click "Tools > Restart" to restart the gateway, making modified configuration come into effect.

CAUTION

In most cases, 'there is no need to reset the gateway, and the modified parameters will come into effect upon confirming the "submit".

2.9.7 System reboot

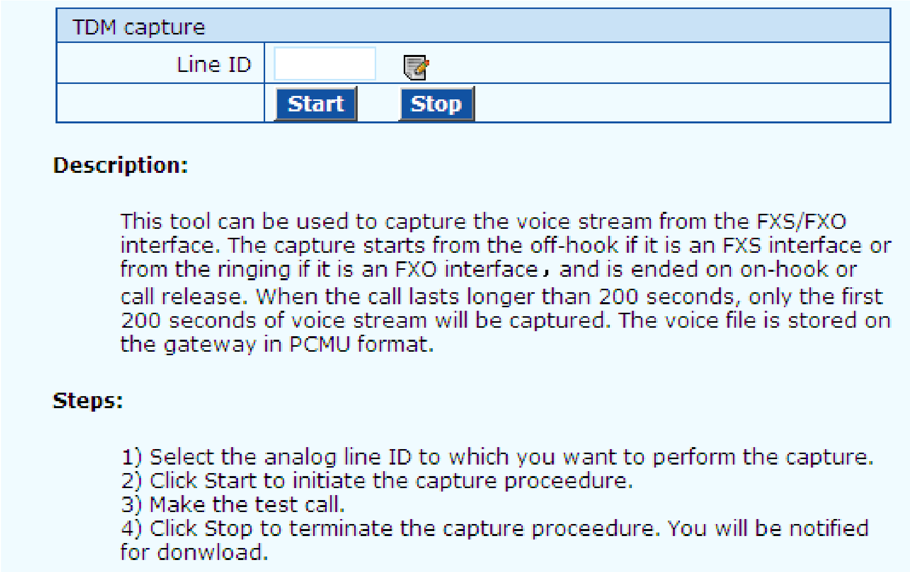
After login, click "Tools >Reboot" to restart the gateway. As this is a system wide reset, it takes longer time.

CAUTION

Generally, it's sufficient to restart software when the gateway confirms to reset; the system reboot will be required only when network settings of the gateway are changed.

2.9.8 TDM Capture

After login, click "Tools > TDM Capture" to open this interface. This tool can be used to capture the voice stream from the FXS/FXO interface. The capture starts from the off-hook if it is an FXS interface or from the ringing if it is an FXO interface， and is ended on on-hook or call release. When the call lasts longer than 200 seconds, only the first 200 seconds of voice stream will be captured. The voice file is stored on the gateway in PCMU format.



Steps:

1) Select the analog line ID to which you want to perform the capture.

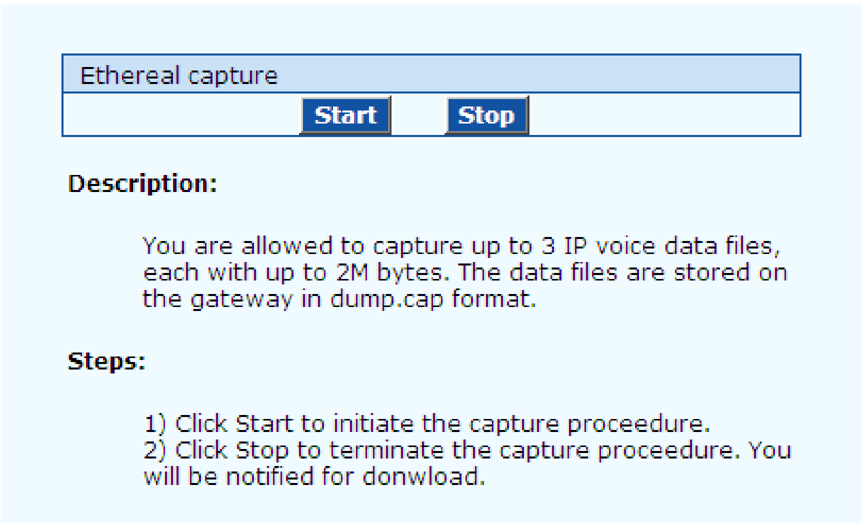
2) Click Start to initiate the capture proceedure.

3) Make the test call.

4) Click Stop to terminate the capture proceedure. You will be notified for donwload.

2.9.9 Ethereal Capture

After login, click "Tools > Ethereal Capture" to open this interface. You are allowed to capture up to 3 IP voice data files, each with up to 2M bytes. The data files are stored on the gateway in dump.cap format under catalog "/var/log".



Steps:

1) Click Start to initiate the capture proceedure.

2) Click Stop to terminate the capture proceedure. You will be notified for donwload.

2.10 Version information

After login, click “Info” to view the gateway hardware and software version information.

Software version	Rev 1.9.82.303
Hardware version	Rev 1.0.1 MI120-24S/24-C
Kernel version	Kernel 1.1.8 (F)
DSP version	Rev 1.8.195

2.11 Logout

After login, click the “Logout” at top right to exit the gateway management system and return to the login interface.

3 Appendix

3.1 WSS120 system operation state

Table 3-1 WSS120 system operation state

Glittery letter	Status meaning
"C"	The IP address of gateway conflicts with that of other equipment in LAN. Please settle this problem before the gateway can be operated normally.
"D"	Internal failures have been entountered during gateway start up procedure. Please contact your local distributor for further diagnosis.
"P"	The gateway is in progress of system software upgrade. Please guarantee stable power supply and do not conduct other operations during this period.
"T"	The application software of gateway has been exited. If it can not be restored by rebooting the system, please contact your local distributor for further diagnosis.

If you have any other problems please send mail to support@realtonetech.com

Thanks!