IP-PBX Series

OM10

OM20

OM200

User Manual

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Amendent Records

Document Rev. 2.0 (Jan. 07, 2010)

Content updates of the document are as follows:

1) Several features are added, including dumping the TDM signal and IP packets from Web-based utility
2) Changes are made for the configuration of analog extension, IP extension, IP trunks, and attendant, and dialing plan and feature codes

Document Rev. 1.1 (Oct. 27, 2009)

Content updates of the document are as follows:

3) Adjustment of content was made based on the new Web user interface.
4) OM20 and OM200 were added.
5) Instructions for use of recording, tie trunk and other new functions were added.

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1 Overview

1.1 Introduction

Officium (hereinafter called “OM”) is a “All-in-One” business telephone system designed for small and mid-size enterprises. It is integrated with Internet telephony, legacy PBX features, instant messaging, user status display, and local and remote management mechanism to provide an efficient and easy-to-use unified communication platform for office staff. Featuring built-in interfaces to analog phones, PSTN lines, SIP-based trunks and SIP registration service, OM connects directly to analog phones, facsimile machines, lines from local Central Office, as well as to components in IP telephony network such as soft switches, IP-PBXs and SIP telephone and soft phone. OM offers functions such as auto-attendant, monitoring of extension status, mobile extension, callback, corporate CRBT, click-to-dial, call transfer, call waiting, voice mail, call recording and etc.

OM is a cost-effect solution for carriers to deploy business telephone service to enterprises, and it is also used for small and mid-size enterprises to build private telephone system serving multi-location branch offices and home working.

OM Series include OM10,OM20,OM100 and OM200 subseries. Their features are similar with the main differences as follows:

Table 1-1 Differences Between OM Series

	Capacity	Chassis	Line Card	Installation	Power
OM10	Able to configure 4-8 user ports and register 30 IP extensions	Plastic casing	Built-in	Desktop	5-9 VDC
OM20	Able to configure 16-24 user ports and register 60 IP extensions	19–inch wide and 1U high chassis	Built-in	Rack	100-240 VAC
OM100	Able to configure 24-48 user ports and register 120 IP extensions	19–inch wide and 1U high chassis	Pluggable	Rack	100-240 VAC, -48 VDC (Optional)
OM200	Able to configure 32-96 user ports and register 120 IP extensions	19–inch wide and 1U high main chassis and extension chassise as option	Pluggable	Rack	100-240 VAC, -48 VDC (Optional)

1.2 Functions and Features

- Built-in SIP register and proxy server
- Local analog extensions and remote SIP extensions
- Mobile extensions
- Auto-attendant

- Operator
- Direct inward dialing (DID)
- Automatic call distribution
- Extension features such as call forwarding, call waiting, call hold, call pickup, call transfer, five-level call restriction, CRBT, and etc.
- Voice mail and call recording
- Mornitoring of extension and trunk status
- Encryptions
- Call logs
- Call detailed record
- Web-based utility for local and remote management
- SIP-based Tie trunks
- Routing table up to 100 rules
- Gain adjustment of extensions and analog trunks
- STUN and NAT traversal
- Support terminals including telephone, facsimile machine and PBX
- T.30/T.38 fax
- Second stage dialing or voice prompt over FXO ports
- PSTN failover through FXO ports
- XML/HTTP-based API for 3rd party application software

1.3 **Equipment Structure**

1.3.1 OM10

OM10 is the product with smallest capacity in OM IPPBX Series. Designed with small plastic structure for desktop placement, OM10 can provide up to 8 analog line interfaces. OM10 supports the following types of configuration:

Table 1-2 OM10 Configurations

Models	Local extension	Analog trunks	SIP extensions	SIP trunks
OM10-4S	4	0	30	20
OM10-8S	8	0	30	20
OM10-4FXO	0	4	30	20
OM10-8FXO	0	8	30	20
OM10-4S/4	4	4	30	20

Figure 1-1 OM10 Front Panel

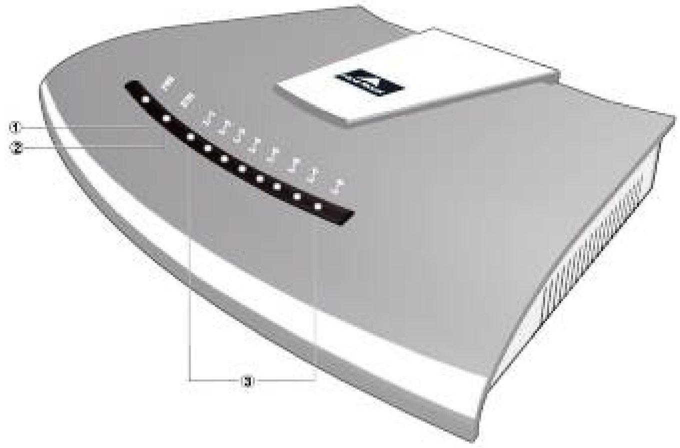


Table 1-3 Description of OM10 Front Panel

#	Description
①	Power indicator (PWR), the light on indicates that it has been powered.
②	Ethernet interface indicator (ETH), the light on indicates successful connection, the light flashing indicates that data packets are being received or sent.
③	Analog line (FXS) or analog trunk (FXO) interface indicator, the light on indicates that it is in use. After power-on and normal startup, FXS or FXO indicators #1, 3, 5, 7 and 2, 4, 6, 8 cross flashing indicates that OM10 has detected IP address conflict.

Figure 1-2 OM10 Back Panel

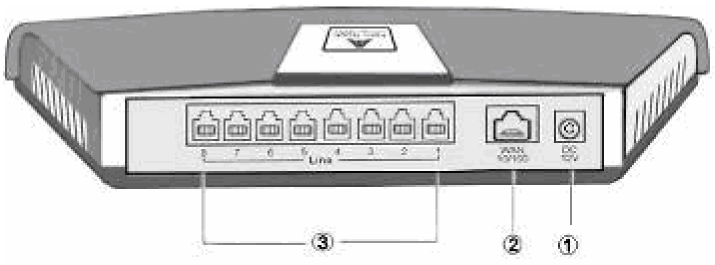


Table 1-4 Description of OM10 back panel

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

Table 1-5 Configuration Description of Analog Extension Interfaces for OM10

OM10 Models	RJ11 Interface Configuration							
	1	2	3	4	5	6	7	8
OM10-4S	Line 1	Line 2	Line 3	Line 4	NA	NA	NA	NA
OM10-8S	Line 1	Line 2	Line 3	Line 4	Line 5	Line 6	Line 7	Line 8

OM10 Models	RJ11 Interface Configuration							
	1	2	3	4	5	6	7	8
OM10-4FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	NA	NA	NA	NA
OM10-8FXO	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4	Trunk Line 5	Trunk Line 6	Trunk Line 7	Trunk Line 8
OM10-4S/4	Line 1	Line 2	Line 3	Line 4	Trunk Line 1	Trunk Line 2	Trunk Line 3	Trunk Line 4

1.3.2 OM20

Designed with a 1U high and 19–inch wide compact chassis, OM20 is suitable for installation in a standard cabinet. It has a built-in 110-220V power module. OM20 uses RJ45 for the interface socket of analog lines and trunks. OM20 supports the following types of configuration:

Table 1-6 OM20 Configuration

Models	Local extension	Analog trunk	SIP extensions	SIP trunks
OM20-16S	16	0	60	20
OM20-24S	24	0	60	20
OM20-16FXO	0	16	60	20
OM20-8S/8	8	8	60	20
OM20-12S/4	12	4	60	20
OM20-16S/8	16	8	60	20
OM20-20S/4	20	4	60	20

Figure 1-3 OM20 Front Panel

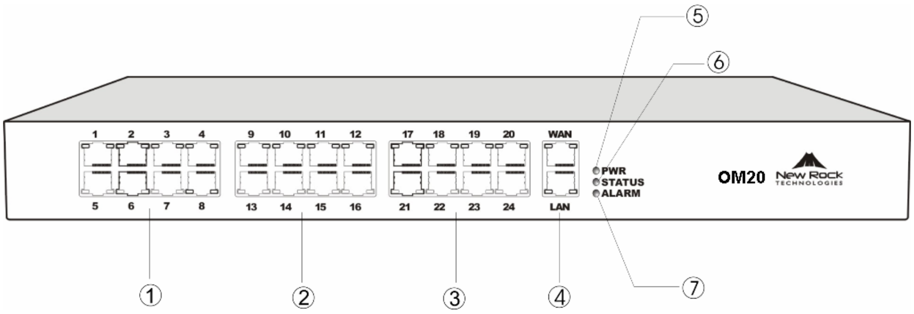


Table 1-7 Description of OM20 Front Panel

#	Description
①	Offer the first 8 analog line interfaces (FXS or FXO)，interface type is RJ45.
②	Offer 9th-16th analog line interfaces (FXS or FXO), interface type is RJ45.
③	Offer 17th-24th analog line interfaces (FXS or FXO), interface type is RJ45. When the total system capacity of OM20 is 16 lines, these interfaces will not be provided.
④	One 10/100 Ethernet uplink interface (WAN) and one 10/100 Ethernet user interface (LAN), interface type is RJ45. The uplink interface is used for calls and management. The user interface is reserved for future use.

#	Description
⑤、⑥、⑦	Three indicators of PWR, STATUS and ALARM represent the power, status and alarm respectively. Specific meanings of these indicators are described in Table 1-10.

Table 1-8 Pins and Indicators for FXS and FXO of OM20

Pin								LED	
1	2	3	4	5	6	7	8	Yellow	Green
NC	NC	NC	RING	TIP	NC	NC	NC	Interface Type	Interface Status

Note: The yellow LED on RJ45 socket works only in the mixed FXS/FXO type configuration of OM. The yellow LED light on indicates the interface is FXO type, and the yellow light off indicates the interface is FXS type. In an configuration with only FXS ports or FXO ports, the yellow LED remains off.

Table 1-9 for Pin and Indicator Table of Ethernet Ports

Pin				LED	
1	2	3	6	Yellow	Green
TX+	TX-	RX+	RX-	Connection Status	Activity Status

Table 1-10 Indicators of OM20

Mark	Function	Status	Description
PWR	Power Indication	Green	Power on
		Off	Power off
STATUS	Status Indication	Off	System locked and inactive
		Green Flash	In normal operation
		Constant Red	System in the process of power up and not in the normal operation mode
		Red Flash	System in a diagnostic mode and able to execute only limited operation
ALARM	Alarm Indication	Green	No alarms
		Red Flash	New alarms occurred but not confirmed
		Red	Alarms existed and all alarm information confirmed
After normal startup of OM20, three indicators of PWR, STU and ALM on the front panel have been found to turn green, then STATUS (STU) and ALARM (ALM) indicators turning from green into red indicating the equipment has detected IP address conflicts.			

Figure 1-4 OM20 back panel

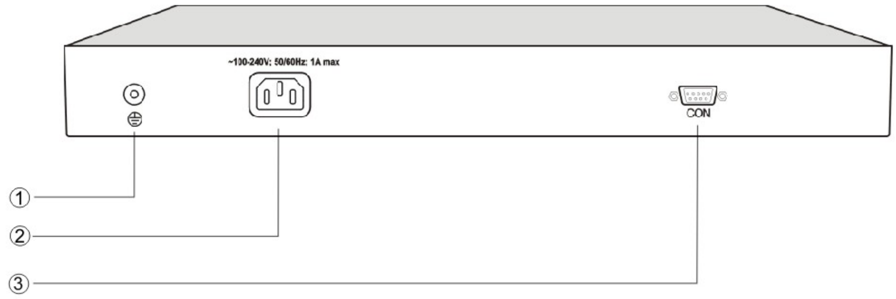


Table 1-11 Description of OM20 Back Panel

#	Description
①	Ground pole
②	AC power input socket, 100-240 VAC voltage input.
③	The configuration interface (CON) is used for local management and debugging. It connects with RS232 port on a computer, and local PCs can establish a connection with OM20 through an emulator on the configuration terminal. Table 1-12 describes the interface properties.

Table 1-12 Properties of OM20 CON Port

Properties	Description
Connector	DB9
Number of interface	1
Interface standard	RS232
Baud rate	38400
Data bit	8
Parity check	No
Stop bit	1
Traffic control	No

1.3.3 OM100

Designed with a 1U high and 19-inch wide compact chassis and a swappable modular structure, OM100 offers flexible on-site configuration and replacement. The interface card of OM100 uses a Champ50-type socket and is connected to the distribution panel in equipment room using a 25-pair cable supplied with the unit. OM100 supports the following types of configuration:

Table 1-13 OM100 Configuration

Models	Local extension	Analog trunk	SIP extensions	SIP trunks
OM100-48S	48	0	60	60
OM100-32FXO	0	32	60	60
OM100-16S/16	16	16	60	60
OM100-32S/4	32	4	60	60
OM100-32S/8	32	8	60	60

Figure 1-5 OM100 Front Panel

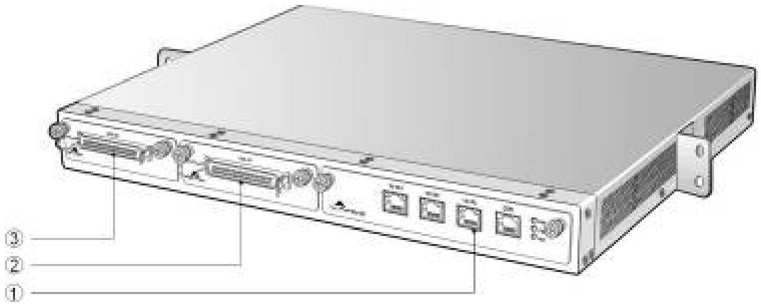


Table 1-14 Description of OM100 Front Panel

#	Description
①	Main control module. It offers one 10/100 Ethernet port and one configuration interface (CON).
② and ③	Two interface slots, and each can contain one desired type interface card.

Table 1-15 Indicators of OM100

Mark	Function	Status	Description
PWR	Power indication	Green	Power on
		Off	Power off
STU	Status indication	Off	System locked and inactive
		Green flash	In normal operation
		Constant red	System in the process of power up and not in the normal operation mode
		Red flash	System in a diagnostic mode and able to execute limited operation
ALM	Alarm indication	Green	No alarms
		Red flash	New alarms occurred but not confirmed
		Red	Alarms existed and all alarm information confirmed
After normal startup of OM100, three indicators of PWR, STATUS (STU) and ALARM (ALM) on the front panel have been found to turn green, and then ALARM (ALM) indicator turning from green into red indicating the equipment has detected IP address conflicts.			

Figure 1-6 OM100 Back Panel

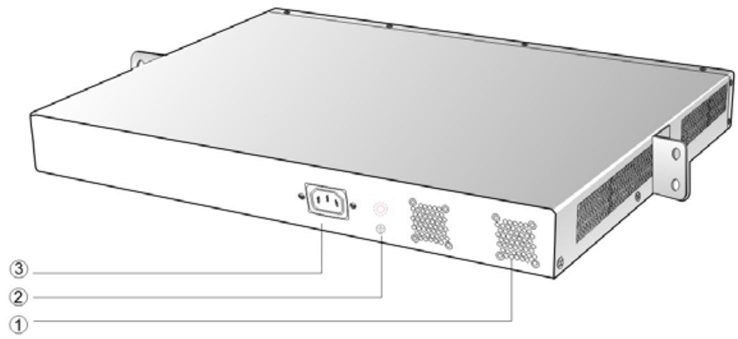


Table 1-16 Description of OM100 Back Panel

#	Description
①	Two cooling fans
②	Ground pole
③	AC power socket, 100-240 VAC voltage input.

1.3.4 OM200

Designed with a 1U high and 19-inch wide compact chassis with swappable modular structure of interfaces, OM200 can be scalable to have an expansion chassis which holds two interface cards. The interface card of OM200 use RJ45 sockets and they are connected to the distribution panel of equipment room using a CAT-5 Ethernet cable to offer flexible user interface configuration.

The main chassis of OM200 can hold two interface cards which enable to flexibly configure the number of FXS and FXO ports，and each card equips up to 24 ports. The expansion chassis can also hold two 24-port interface cards which enable to flexibly configure the number of FXS and FXO ports. OM200 dual-chassis system can provide up to 96 ports. It supports the following configurations:

Table 1-17 OM200 Configuration

Models	Local extension	Analog trunk	SIP extensions	SIP trunks
OM200-48S	48	0	120	80
OM200-72S	72	0	120	80
OM200-96S	96	0	120	80
OM200-24FXO	0	24	120	80
OM200-48FXO	0	48	120	80
OM200-72FXO	0	72	120	80
OM200-96FXO	0	96	120	80
OM200-40S/8	40	8	120	80
OM200-64S/8	64	8	120	80
OM200-88S/8	88	8	120	80
OM200-36S/12	36	12	120	80
OM200-60S/12	60	12	120	80
OM200-84S/12	84	12	120	80
OM200-32S/16	32	16	120	80
OM200-56S/16	56	16	120	80
OM200-80S/16	80	16	120	80
OM200-28S/20	28	20	120	80
OM200-52S/20	52	20	120	80
OM200-76S/20	76	20	120	80
OM200-24S/24	24	24	120	80
OM200-48S/24	48	24	120	80
OM200-72S/24	72	24	120	80
OM200-44S/28	44	28	120	80
OM200-68S/28	68	28	120	80
OM200-40S/32	40	32	120	80

Models	Local extension	Analog trunk	SIP extensions	SIP trunks
OM200-64S/32	64	32	120	80
OM200-36S/36	36	36	120	80
OM200-60S/36	60	36	120	80

Figure 1-7 OM200 Front Panel

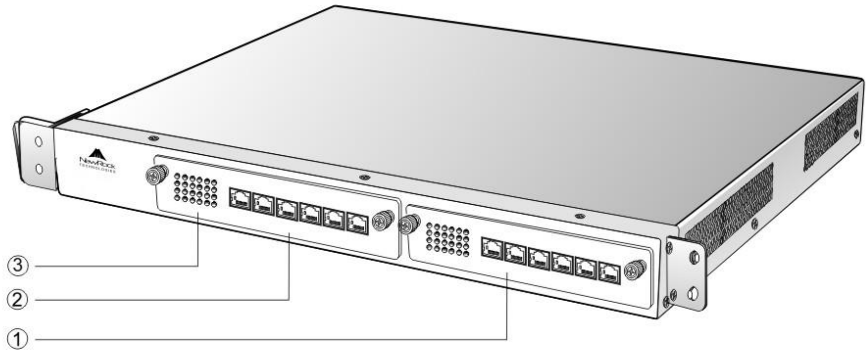


Table 1-18 Description of OM200 Front Panel

#	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 OM200 when equipment is powered on.

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 OM200 RJ45 Socket Port

Numerical sequence of line	1st Pair		2nd Pair		3rd Pair		4th Pair	
Pin No. of RJ45 contact	1	2	3	**6**	**4**	**5**	7	8
Corresponding RJ11	TIP1	RING1	TIP2	RING2	TIP3	RING3	TIP4	RING4

Table 1-20 Corresponding Relation Between OM200 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 installation of equipment or wiring check. This mode can be entered by disconnecting the network interface (two Ethernet interfaces on the host are disconnected) when connecting lines at installation stage, or through interface control when the check is made during normal operation. After entering the mode, LED constant on indicates that the corresponding line is equipped and is an analog telephone line, LED flashing indicates that the corresponding line is equipped and is an analog trunk lines; LED off indicates that the corresponding line is not equipped or is faulty.
- 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 OM200 System.

Figure 1-8 OM200 Back Panel

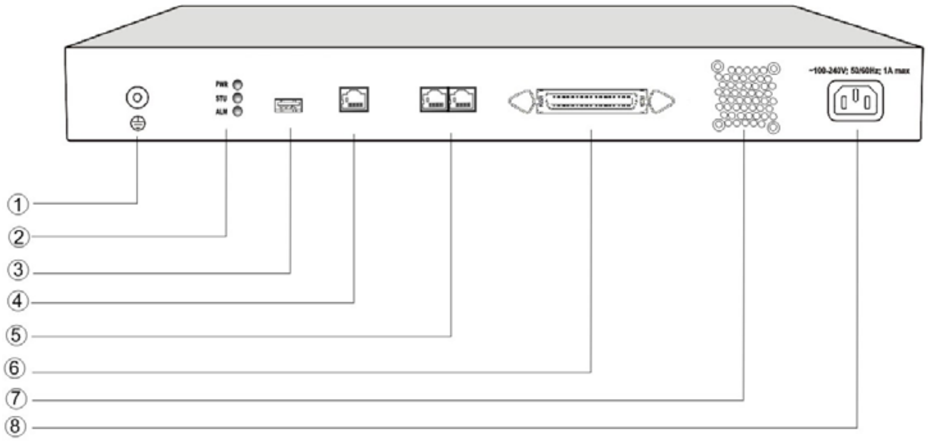


Table 1-21 OM200 Back Panel

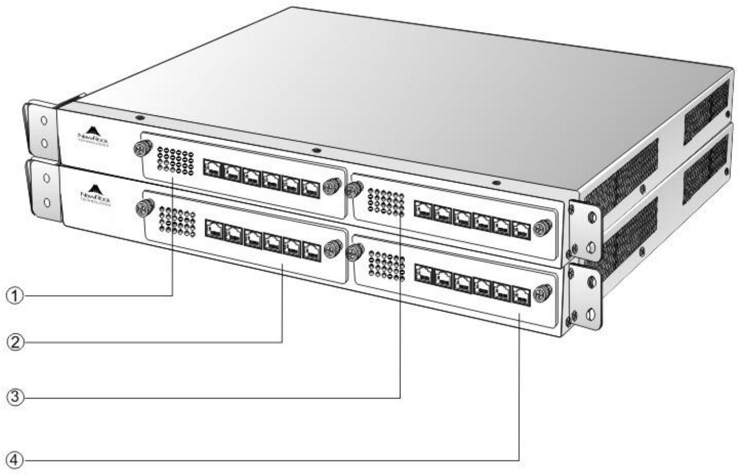
#	Description
①	Ground pole
②	Indicator, see Table 1-22 for description.
③	USB interface, reserved for future use.
④	Configuration interface (CON), used for local management and debugging.
⑤	Two Ethernet interfaces: ETH1 and ETH2, only ETH1 has been set when the equipment is delivered from factory, default IP address: 192.168.2.240
⑥	Connection interface of expansion chassis
⑦	Cooling fan
⑧	AC power socket, 100V-240 VAC voltage input.

Table 1-22 Meanings of OM200 Indicators

Mark	Function	Status	Description
PWR	Power indication	Green	Power on
		Off	Power off
STU	Status indication	Off	System locked and inactive
		Green flash	In normal operation
		Constant red	System in the process of power up and not in the normal operation mode
		Red flash	System in a diagnostic mode and able to execute limited operation
ALM	Alarm Indication	Green	No alarms
		Red Flash	New alarms occurred but not confirmed
		Red	Alarms existed and all alarm information confirmed
After normal startup of OM200, the indicator displaying letter C flash indicates that OM200 has detected IP address conflicts.			

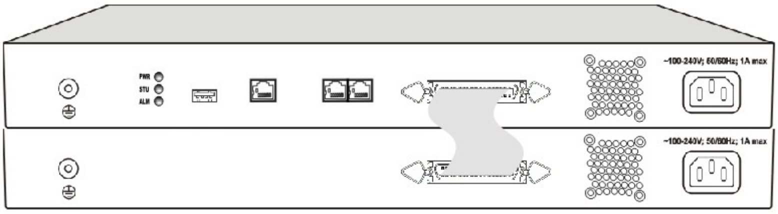
OM200 single-chassis system can provide up to 48-line with different analog line type configuration combinations, and OM200 dual-chassis system (main chassis and expansion chassis) can provide maximum of 96-line with different analog line type configuration combination. OM200 dual-chassis system is a logical integrated system formed by adding an expansion chassis and interface cards on top of the single chassis system. An expansion cable is required to connect the two chassises, and the operation of the main chassis will not be affected when connecting the expansion cable, suitable for on-site capacity expansion. The system resources (e.g. number of concurrent calls) are determined by the main chassis, so users who are planning to expand capacity should take into account the needs for future system resources when initially ordering the single-chassis system, to support the smooth capacity expansion and upgrade.

Figure 1-9 Schematic Diagram for Front Panel of OM200 Dual-Chassis System



Numbering definition of system interface slots: On the left side of main chassis is #1 slot (marked with ① in Figure 1-9), on the right side of main chassis is #2 slot (marked with ③ in the Figure 1-9), on the left side of expansion chassis is #3 slot (marked with ② in the Figure 1-9), and on the right side of expansion chassis is #4 slot, marked with ④ in Figure 1-9).

Figure 1-10 Schematic Diagram for Back Panel of OM200 Dual-Chassis System



OM200 dual-chassis system consists of a main chassis and an expansion chassis. Seen from the front, it is different from the screen-printed mark: main chassis printed with "OM200-MAIN" mark and expansion chassis printed with "OM200-EXT" mark. Seen from the back, the difference is obvious: expansion chassis does not have status indicator, USB interface, CON interface and two Ethernet interfaces. The main and expansion chassises have their own independent power supply and cooling system, which are connected by a 36-core flat cable to form a logical integrated system.

To ensure the reliable communications between two chassises and reduce EMI interference, the communication cable for connecting two chassises should be short. OM200 dual-chassis system must be placed adjacently in installation (users who plan to expand capacity should reserve a space for expansion chassis during the initial installation of single chassis), the up or lower position is not strictly required for the chassises.

2 Parameter Setting

2.1 Login to Web-based Utility

2.1.1 Obtain OM IP Address

OM10 and OM20 start DHCP service by default, and automatically obtain an IP address on the LAN; users can use the factory default IP address if it is unable to be obtained (e.g. when connected directly with a computer).

OM100 and OM200 use a static IP address by default.

Table 2-1 Default IP Address of OM

Type	Default DHCP Service	Default IP Address
OM10	Enabled	192.168.2.218
OM20	Enabled	192.168.2.228
OM100	Disabled	192.168.2.240
OM200	Disabled	192.168.2.240

- DHCP Used in Network

  Users can dial "# #”to obtain the current IP address of the device and version information of firmware using the telephone connected to a FXS port after the equipment is powered on.

  If the OM is only configured with FXO ports for analog trunks without FXS ports (e.g. OM10-4FXO or OM20-24FXO), users can dial into the OM by connecting a PBX extension line or CO line to a FXO port, and press "# #" to obtaining the current IP address of the device and version information of firmware.

- Fixed IP Address Used
  - If the DHCP service on the network is not available or the OM is directly connected with a computer, the OM 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 OM 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. For example, if the OM 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 OM 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 OM to receive the IP address and version information of firmware.

- IP Address Port Configuration of Equipment

  It is typically required to fill in the port when using DDNS since the router possibly maps to a different port in port mapping.

  Normally, it is unnecessary to fill in the port value.

2.1.2 Logon

Double-click the icon to open IE browser, and enter the OM IP address in the browser address bar (e.g. 192.168.2.218). You can logon to the web-based utility by entering a password on the login interface.

Figure 2-1 Login Interface of OM Web-based Utility

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 OM

Type	Default Operator Password	Default Administrator Passwords (lowercase letters required)
OM10	operator	admin
OM20	operator	admin
OM100	operator	admin
OM200	operator	admin

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

The OM allows multiple users to log in:

- ➢ The aadministrator 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 OM Web-based Utility**

"Submit" and "Refresh" buttons are at the bottom of configuration interface.

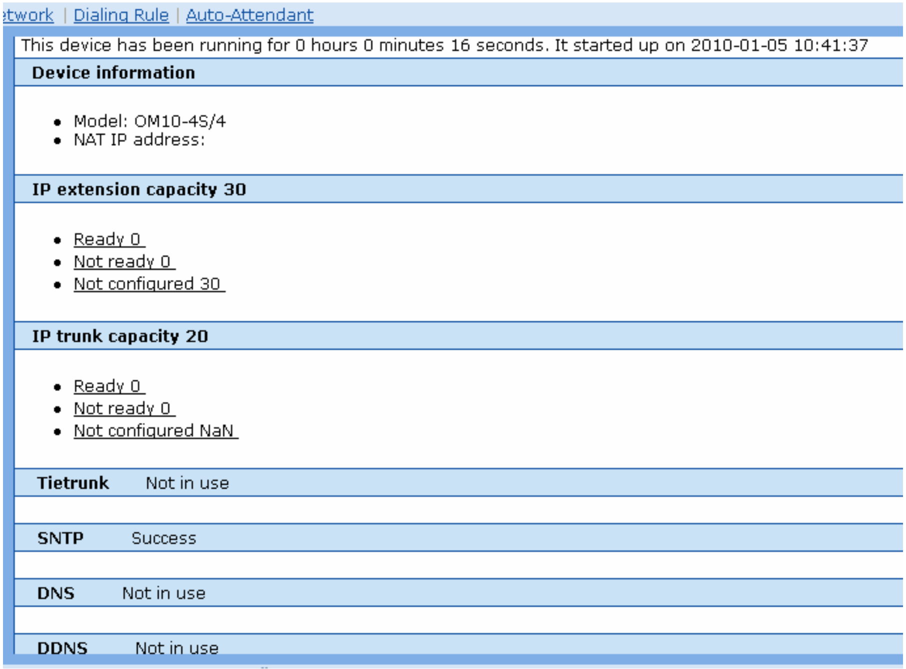
- Submit" Button: Submit configuration information. Users click "Submit" button after completion of parameter configuration on a page. A success prompt will appear if configuration information is accepted by the system; if a "The configuration takes effect after the system is restarted"dialog box appears, it means that the parameters are valid only after system restart; it is recommended that users press the "Rstart" button on the "Tool" page to validate the configuration after changing all parameters to be modified.
- "Refresh": Cancel changed operation that has not been submitted.

2.3 **Basic Configuration**

2.3.1 Status

After login, click "Basic > Status" tab to open the status interface.

Figure 2-2 Status Interface



2.3.2 Network Configuration

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

Figure 2-3 Network Configuration Interface

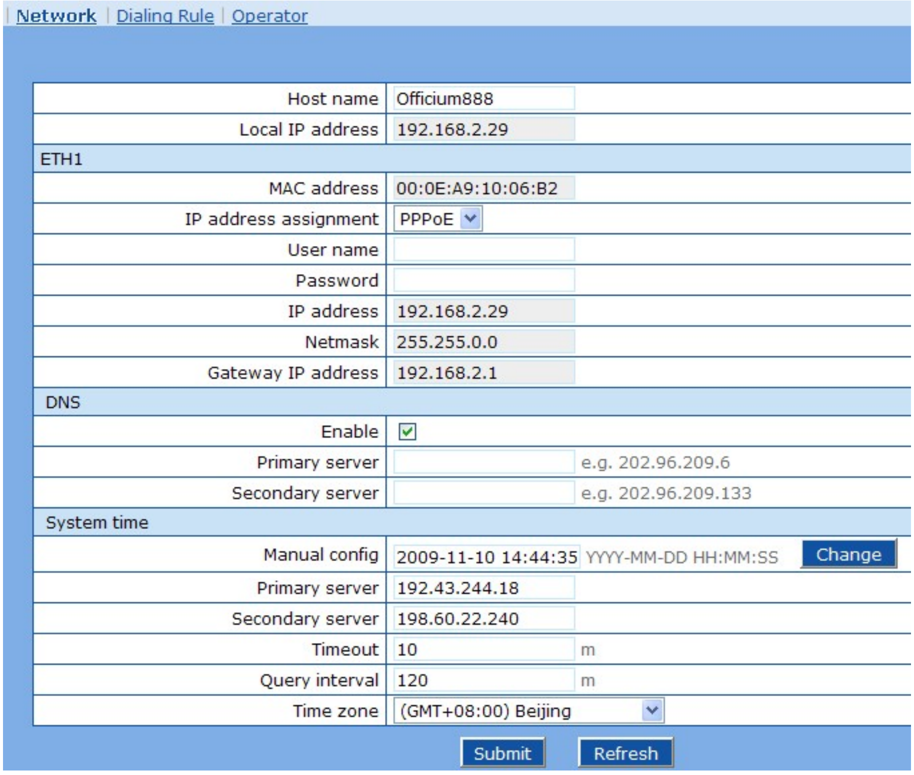


Table 2-3 Configuration parameter of Network

Name	Description
Host name	This is the equipment name of OM. The default values of OM10, OM20, OM100 and OM200 are Officium10, Officium20, Officium100 and Officium200 respectively. Users can set a different name for each OM 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 OM100 and OM200, used to display the actual OM IP address in use.
ETH	
MAC address	Display the MAC address of OM.
IP address assignment	Methods for obtaining an IP address ● Fixed: Static IP address is used; ● DHCP: Activate DHCP service and use the dynamic host configuration protocol (DHCP) to allocate IP addresses and other network parameters; ● 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.

Name	Description
IP address	If “Static” or “DHCP” is selected for the network type but an address fails to be obtained, the OM will use the IP address filled in here. If the OM obtains an IP address through DHCP, the system will display the current IP address automatically obtained from DHCP by the OM. This parameter must be set due to no default value.
Netmask	The subnet mask is used with an IP address. When the OM uses 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 OM is located. When the OM obtains 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.
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	
Manual config	The system will use the time set here if time server doesn’t work.
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.
Timeout	Timeout when periodically sending a request for clock synchronization fails
Query interval	Interval of sending a request for clock synchronization to the time server

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

2.3.3 Dialing Rules

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

Figure 2-4 Dialing rule configuration interface

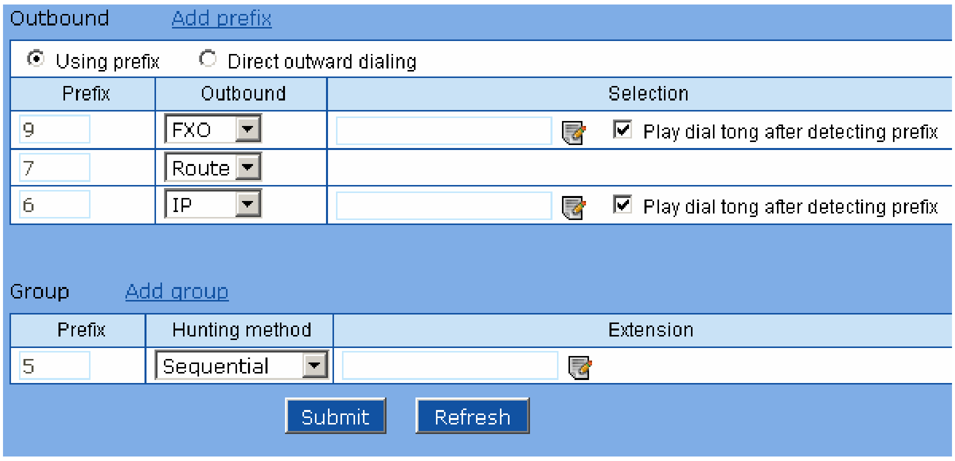


Table 2-4 Configuration parameters of dialing plan

Name	Description
Outbound	
Prefix	Used for selecting a trunk for an outgoing call. Value range: 0~9.
Outbound	The types of trunk for outgoing calls: • FXO: Using analog trunks for outgoing calls with the corresponding prefix. Example: if the prefix of analog trunk is set to 9, users dial 9 to select an analog trunk for outbound call. Click to select the specified analog trunks. • IP: Using SIP trunks for outgoing calls with the corresponding prefix. Example: if the prefix of IP trunk is set to 6, users dial 6 to select an SIP trunk for outbound call. Click to select the specified SIP trunks. • Route: Selecting the trunks according to the routing table. Note: If an outbound trunk is not specified, the equipment will select any available trunk.
Selection	Used to define outbound line.
Group	
Prefix	Used for selecting the group when making an incoming call. Optional value: 0~9.

Name	Description
Hunting method	• Simultaneous: Set a prefix for a group of extensions. When the prefix is pressed in an incoming call, the OM rings idle extensions in the group concurrently; when an extension receives the call, the OM will stop ringing of other extensions in the group. Click to select the extensions for the group. These selected extension lines will ring simultaneously when users dial this prefix. • Sequential: Set a prefix for a group of extensions. When the prefix is pressed in an incoming call, OM distributes the call to the idle extension in a sequential order. Click to select the extensions for the group. • Circular: Set a prefix for a group of extensions. When the prefix is pressed in an incoming call, OM distributes the call to the idle extension in a circular order. Click to select the extensions for the group.
Selection	Used to define line for distributing incoming calls.
Direct Outward Dialing	Used to set direct outward dialing. When choosing "Direct Outward Dialing", you can dial the external phone number without any prefix to reach it. On the other hand, you need to add "*" prefix to an extension number when making an internal call.

2.3.4 Auto Attendant

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

Figure 2-5 Configuration interface of Auto Attendant

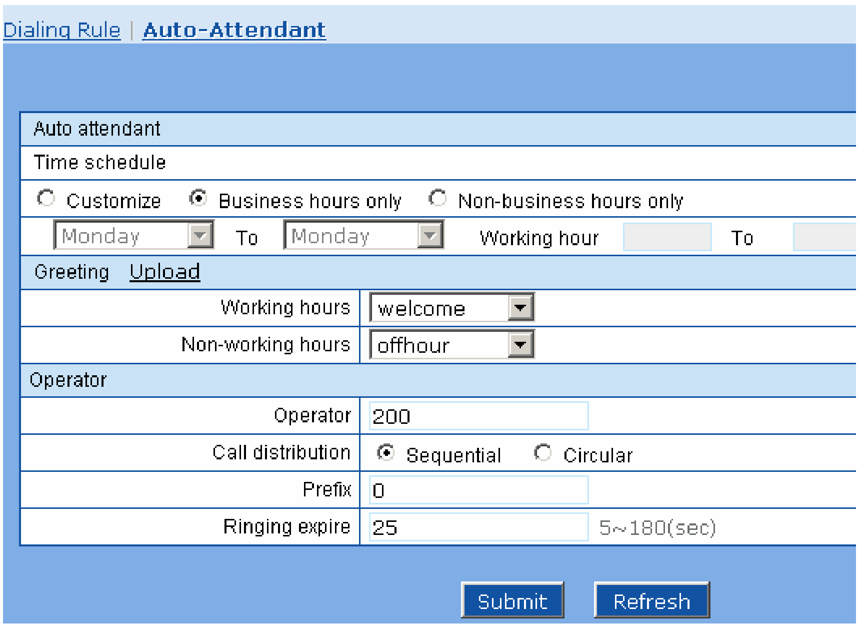


Table 2-5 Configuration Parameters of Auto Attendant

Name	Description
Auto Attendant	

Name	Description
Time Schedule	The IVR will use different voice prompts based on “working hours” and “non-working hours”. Setting of time schedule: ● Customize: Set the start day and end day of working for each week and the start time and end time of working for each day according to the working hours of company. ● Business hours only: Set all time schedules to be working hours. It is generally used for organizations providing 24-hour service. ● Non-business hours only: Set all time schedules to be non-working hours. It is generally used for holidays or out-of-business stage.
Greeting	Set the greetings of auto attendant for working hours and non-working hours.
Operator	Set up to 5 operator extensions.
Operator	Define extensions corresponding to operators with a maximum of 5.
Call Distribution	The caller may select an attendant to answer the call. The OM provides two call distributions methods: ● Sequential: Select idle extensions in sequential order. ● Circular: Select idle extensions in a circular order. These call distributions will perform the same if there is only one operator extension.
Prefix	Used for dialing the operator. Optional value: 0~9.
Ringing Expire	This parameter defines the no answer timeout for non-operator extensions. After the timeout, the call will be returned will be returned to the operator. The unit is second.

2.4 **Extension**

2.4.1 Analog Ext.

After login, click “Extension > Analog Ext.” tab to open the configuration interface.

Figure 2-6 Basic configuration interface of analog extension

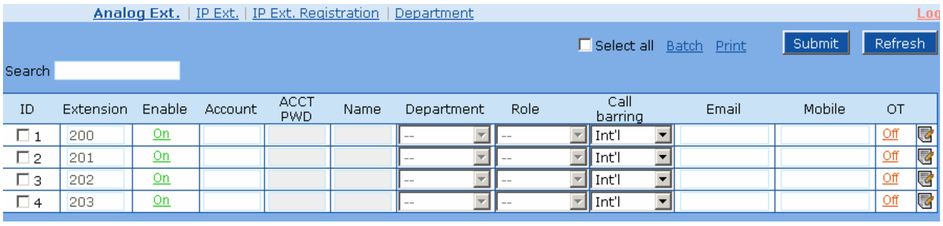


Table 2-6 Basic configuration parameter of analog extension

Name	Description
ID	Port ID of analog extension (read-only).
Extension	Assign the extension number for this analog extension. The default value is not null and its numbering starts with 200.

Name	Description
Enable	Select if the line is set to maintenance status, namely, stop to supply of power for the line port. In this status, the line fails to be used normally. The administrator can disable the extension by setting the line to off status.
Account	Set a user account for logging on to the assistant. It is needed to enable the account if a department is selected.
ACCT PWD	Set a logon password for this account.
Name	User name, which is the same as account name by default. It is not a mandatory option but only used for monitoring of assistant and operator.
Department	Set a department (group) on this line. It is not a mandatory option but only used for monitoring of assistant and operator.
Role	Set the role type on this line (including general user, privileged user and operator) ● User: A user who has permission to use all basic functions except extension and trunk monitoring. ● Manager: A manager can monitor extension and trunk status in addition to playing the "general" role. The system can set up to 3 privileged users. ● Attendant: It is an account type specially designed for attendants. It has no call forwarding and DND but provides setting of call transfer and corporate prompt tone, can monitor all extensions and trunks and view all call logs.
Call Barring	Set the outbound call barring on this line, including: ● None: Can only receive calls, but cannot make any calls; ● Intercom: Can only make internal calls; ● Local: Can make internal and local calls; ● Domestic: Can make internal, local and domestic long-distance calls; ● Int'l: No restriction.
Email	Set an Email bound with this line. This is used for voice mail.
Mobile	Set a mobile number bound with this line. This is used for mobile extension.
OT	Outbound Transfer. This parameter is used with Call Transfer. Select if users are barred for outbound call transfer. By default, this is disabled.

Click to open the whole configuration interface of analog extension.

Figure 2-7 Others configuration interface of analog extension

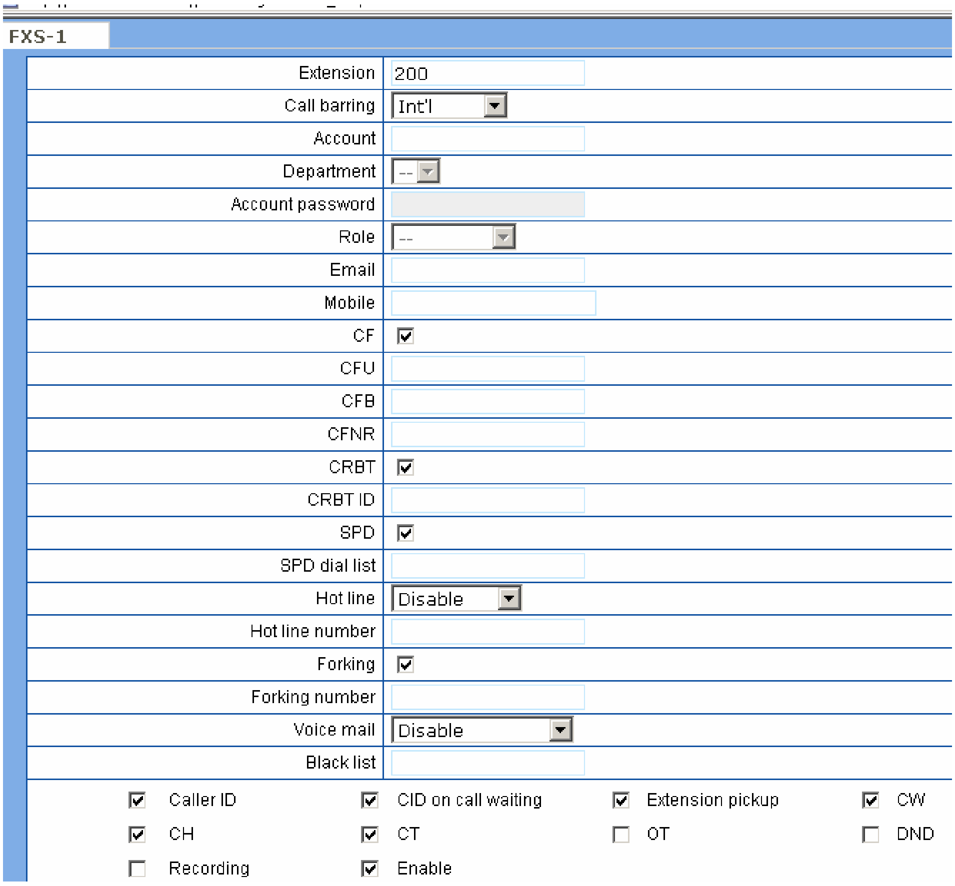


Table 2-7 Others configuration parameter of analog extension

Name	Description
CF	Select if Call Forwarding is activated on this line. By default, this is activated.
CFU	If it is required to forward all incoming calls unconditionally, 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.
CFNR	If it is required to forward an incoming call when there is no answer, enter the new destination number here.
CRBT	Select if CRBT is activated on this line.
CRBT ID	Set the CRBT number with a valid rang of 0~255, where 0 indicates disabling CRBT. The default value is 0.
SPD	Select if the Speed dials is activated on this line. By default, this is not selected.
SPD 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: 20-61202700/23-13052475522/30-96961. Users can set the list on a telephone set and display it here.

Name	Description
Hot Line	Select if the OM is required to automatically dial out the preset hotline number after offhook. By default, hot line is disabled. It supports two types: ● Immediate: Automatically dial out the preset hotline number after offhook. ● Delayed: Automatically dial out the preset hotline number after offhook with a time of 5 seconds.
Hot Line Number	After the hotline function is activated on this line, the hotline number must be entered 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	Forking allows the OM to initiate a call to another telephone terminal (eg. the user’s mobile phone) preset by users while ringing on this extension; and the OM will immediately end up the call on another terminal when users lift off the hook on either terminal. The administrator can set numbers corresponding to forking terminals in this parameter, and users can set it through telephone.
Voice Mail	Select if Voice Mail is activated.
Black list	Enter the list of phone numbers which OM should reject. separated with “,”. e.g.: 02161243400,1305678899
Caller ID	Set if 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.
CID on call waiting	Select if Caller ID Display is activated on this line during call waiting. By default, this is not selected.
Extension pickup	Select if Pickup is activated on this line.
CW	Select if Call Waiting is activated on this line.
CH	Select if Call Hold is activated on this line. By default, this is activated. Note: If this function is activated, the OM will automatically activate Call Transfer.
CT	Select if Call Transfer is activated on this line. By default, this is activated. Example: When A calls B, B picks up the call and transfers it to C. Note: The Call Hold must be activated before Caller Transfer.
OT	Outbound Transfer. This parameter is used with Call Transfer. Select if users are barred for outbound call transfer. By default, this is disabled.
DND	Select if “Do Not Disturb” is activated on this line. By default, this is not selected.
Recording	Select if telephone Recording is activated.
Enable	Select if the line is set to maintenance status, namely, stop to supply of power for the line port. In this status, the line fails to be used normally. The administrator can disable the extension by setting the line to off status.

2.4.2 IP Ext.

After login, click “Extension > IP Ext.” tab to open the configuration interface.

Figure 2-8 Basic configuration interface of IP extension

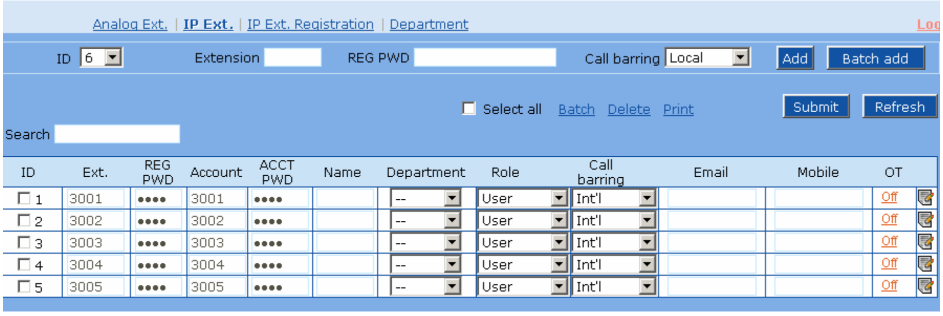


Table 2-8 Basic configuration parameter of IP extension

Name	Description
ID	IP extension line ID, which is automatically created by the system.
Ext.	Number assigned to this IP extension. By default, this is null.
REG PWD	Registration password registered to the OM by IP extension.
Account	Set a user account for logging on to the assistant. It is needed to enable the account if a department and a assistant are selected.
ACCT PWD	Set a logon password for this assistant account.
Name	Display the user name, which is the same as account name by default. It is not a mandatory option but only used for monitoring of assistant and operator.
Department	Set a department (group) on this line. It is not a mandatory option but only used for monitoring of assistant and operator.
Role	Same as analog extension.
Call Barring	Same as analog extension.
Email	Set an Email bound with this line. This is used for voice mail.
Mobile	Set a mobile number bound with this line. This is used for mobile extension.
OT	Select if Outbound Transfer is activated on this line. By default, this is disabled. When this function is mainly used for call forwarding and caller transfer, if this lined can be forwarded or transferred to externally.

Click  to open the whole configuration interface of IP extension.

Figure 2-9 Others configuration interface of IP extension

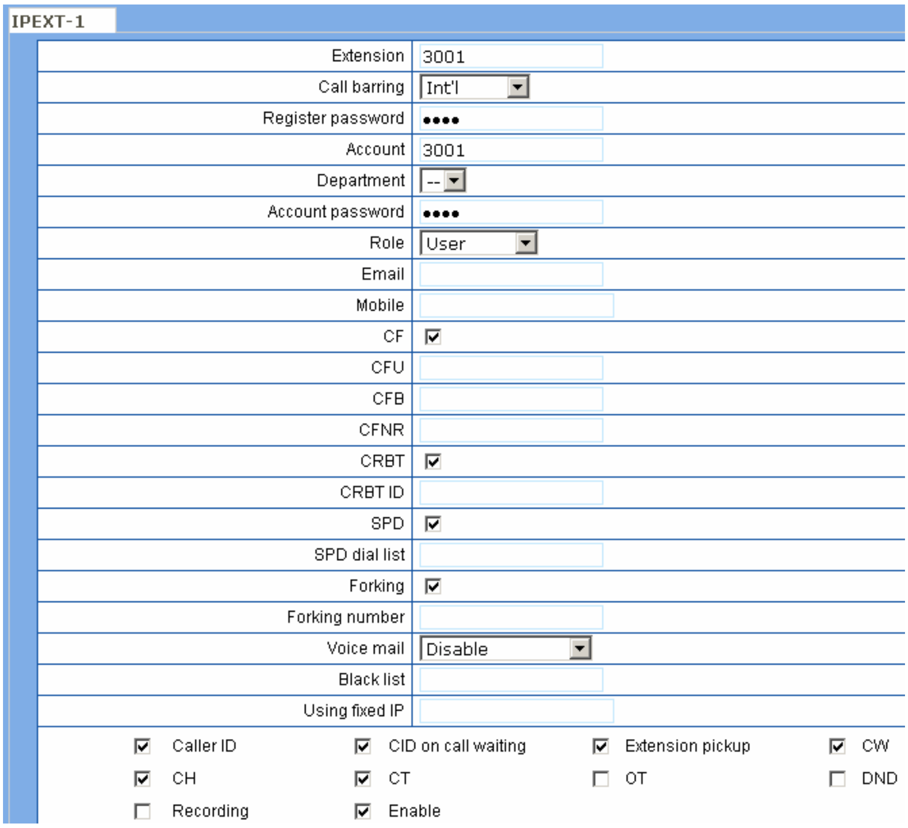


Table 2-9 Others parameter of IP extension

Name	Description
	Same as analog extension.
Using fixed IP	If the SIP terminal uses IP address as the means of security verification, enter the static IP address of the terminal.

2.4.3 Department

After login, click “Extension > IP Ext. Registration” tab to open the configuration interface.

Figure 2-10 Registration configuration interface of IP extension

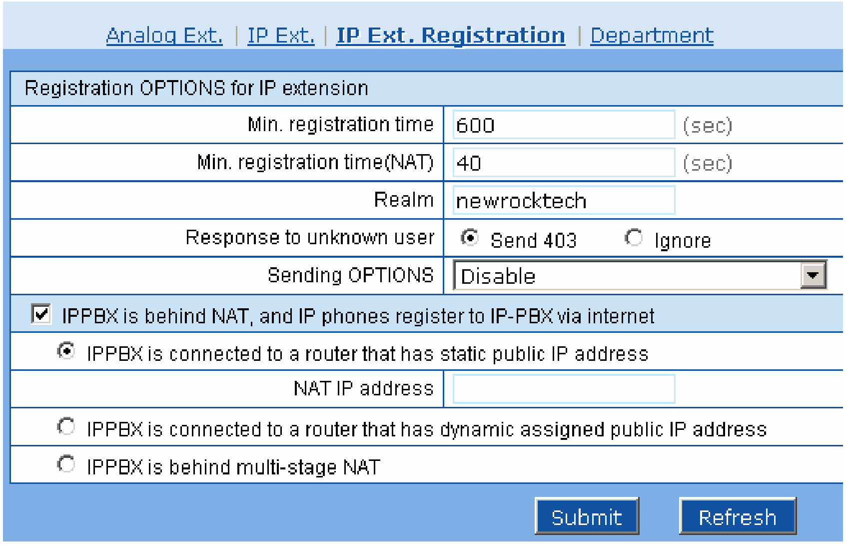


Table 2-10 Registration configuration parameter of IP extension

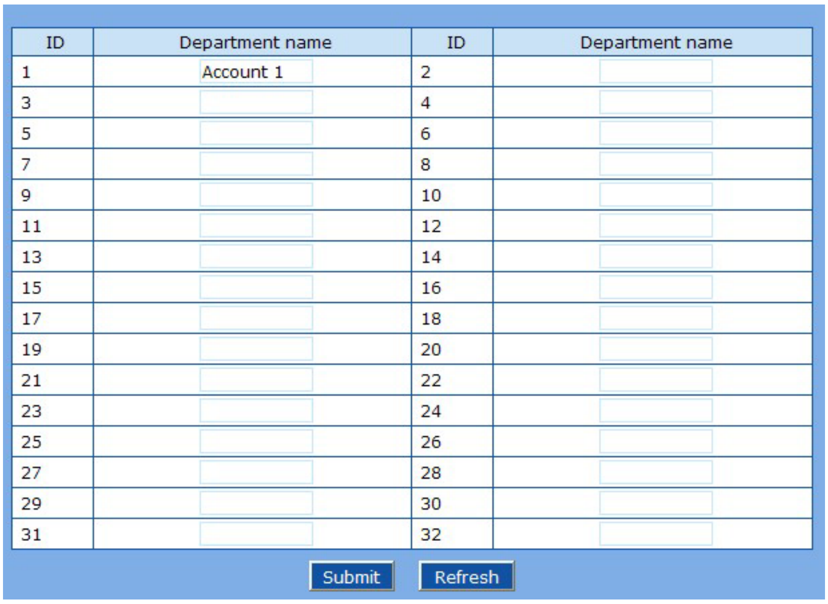
Name	Description
Min. Registration Time	When IP extension is not behind NAT, the minimum registration time cannot be smaller than this value. If the registration time of IP extension is smaller than this value, the OM will tell IP extension terminal about the restriction with a 423 message.
Min. Registration Time of IP Extension Behind NAT	When IP extension is behind NAT, the minimum registration time cannot be smaller than this value. If the registration time of IP extension is smaller than this value, the OM will tell IP extension terminal about the restriction with a 423 message. Normally, when IP extension is behind NAT, the OM should allow the extension for a new registration in a shorter time to ensure reliable connection.
Realm	Used for IP extension authentication.
Response to Unknown User	Processing modes for SIP terminal without an account attempting to register to the OM: Send 403: Send 403 to the IP extension in response; Ignore: Make no response to registration.
Sending OPTIONS	The OM maintains the session status by sending OPTIONS regularly. The following selections can be made according to application scenarios: ● Disable ● To registered IP Ext. behind NAT ● To any registered IP Ext.
Time interval for OPTIONS	30~65535 s.

<table>
<tr><th>Name</th><th>Description</th></tr>
<tr><td>IPPBX is behind NAT, and IP phones register to IP-PBX via internet</td><td>
IPPBX is connected to a router that has static public IP address. Set the public IP address of router.
In this case, the public IP address that the router maps to the OM is static, so external IP extensions can register to the OM through this public address. But since the OM is behind the router (NAT), it is required to enable port mapping for signaling and voice port of the OM on the router to ensure normal communication by IP extension.
IPPBX is connected to a router that has dynamic assigned public IP address. Set if DDNS service supplied with the OM is used: fill in the user name, password and domain name if Yes is selected; fill in the DDNS domain name if No is selected.
When enterprises are connected to the Internet through ADSL, the public IP addresses that operators provide are usually dynamic, namely, the public IP addresses that the router maps to the OM are always changing. So external IP extensions cannot register to the OM through a static address. To solve this problem, users are required to assign a domain name for the OM by applying for DDNS service, then external IP extensions can register to the OM through this domain name. Also, since the OM is behind the router (NAT), it is required to enable port mapping for signaling and voice port of the OM on the router.
IPPBX is behind multi-stage NAT (e.g. There exists a multi-stage router between the IPPBX and the Internet).
In this case, since there exists a multi-stage router between the OM and the Internet, and the router controlled by an enterprise is behind NAT, the communication problem between the OM and external IP extensions cannot be solved simply by port mapping. Therefore, it is needed to enable STUN function (STUN protocol provides a way by which it can get the address and port mapped by NAT, and so replace the private network address and port in application layer to traverse NAT) for working with the STUN server in public network to access the OM behind NAT, ensuring normal communication by external IP extensions. The OM has three strategies for using STUN service:
1、 Startup: The OM sends STUN request for signaling port to the STUN server only at application boot;
2、 Periodic: The OM sends STUN request for signaling port to the STUN server on a periodical basis;
3、 Periodic & RTP: The OM sends STUN request for signaling and voice ports to the STUN server on a periodical basis.
In this case, it is still needed to fill in DDNS or static NAT address for IP extension registering to the OM.
</td></tr>
</table>

2.4.4 Department

After login, click “Extension > Department” tab to open the configuration interface.

Figure 2-11 Department configuration interface



Department setting interface is mainly used for setting department (group) name, with which is filled in the blanks. Intra-department pickup and assistant functions will use the department information. After the department is set, it will appear on the dropdown list under “Department” on the “Extension > Analog Ext.” and “Extension > IP Ext.” configuration interface for extension management and assistant.

2.5 **Trunking**

2.5.1 Analog Trunk

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

Figure 2-12 Configuration interface of analog trunk

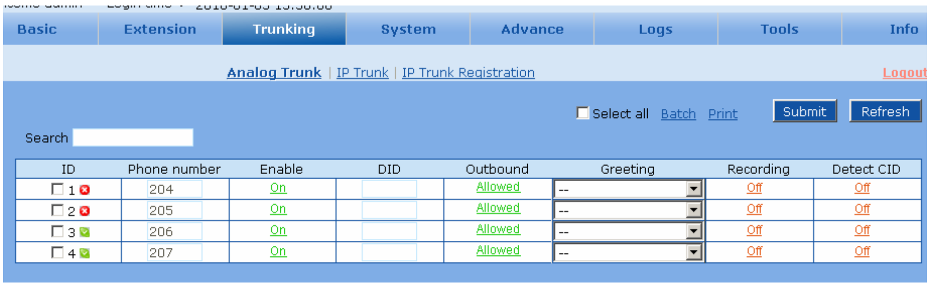


Table 2-11 Configuration parameter of analog trunk

Name	Description
ID	Analog trunk ID, which is automatically created by the system.
Phone Number	Assign a phone number for this trunk line. When there is no caller ID display, the system will use this number as the caller ID. This number is also used in the Call Detail Record (CDR).
Enable	Enable this analog line. If FXO port fails or other maintenance happens, it is suggested to disable this line to ensure it will not be selected when making outgoing calls.

Name	Description
DID	Set an extension number bound with this analog trunk. Enter the extension number to enable DID function. When there is an incoming call on this line, the system will ring the corresponding extension directly.
Outbound	Select if Outbound is activated. This line can only be used for incoming calls when this function is activated.
Greeting	Set the operator greeting for this analog trunk. The default greeting is used when the status is off.
Recording	Select if Recording is activated. The OM will record calls (including incoming and outgoing calls) on this analog trunk when this function is activated.
Detect CID	Select if Caller ID detection is activated.

2.5.2 IP Trunk

After login, click "Trunking > IP Trunk" tab to open the configuration interface.

Figure 2-13 General configuration interface of IP trunk

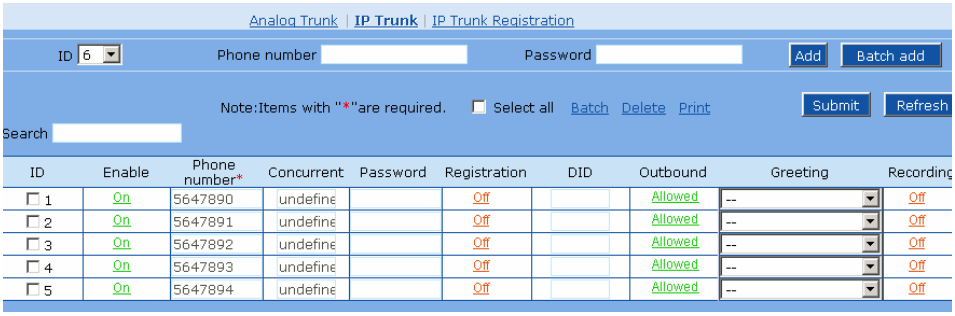


Table 2-12 General configuration parameter of IP trunk

Name	Description
ID	IP Trunk ID, which is automatically identified by the system.
Enable	Enable this IP line.
Phone number	Assign a phone number for this trunk line. When there is no caller ID display, the system will use this number as the caller ID. This number is also used in the Call Detail Record (CDR).
Concurrent	Set the maximum concurrent calls allowed on the trunk. The default is 1.The number must be no bigger than the total number SIP trunks.
Password	Configure an authentication password for the registration account. Obtain it for the operator or system administrator.
Registration	Select if this account is registered with the operator. By default, this is off.
DID	Set an extension number bound with this analog trunk. Enter the extension number to enable DID function. When there is an incoming call on this line, the system will ring the corresponding extension directly.
Outbound	Select if Outbound is activated. This line can only be used for incoming calls when this function is activated.
Greeting	Set the operator greeting for this analog trunk. The default greeting is used when the status is off.

Name	Description
Recording	Select if Recording is activated. The OM will record calls (including incoming and outgoing calls) on this analog trunk when this function is activated.

2.5.3 IP Trunk Registration

After login, click “Trunking > IP Trunk Registration” tab to open the configuration interface.

Figure 2-14 Registration configuration interface of IP trunk

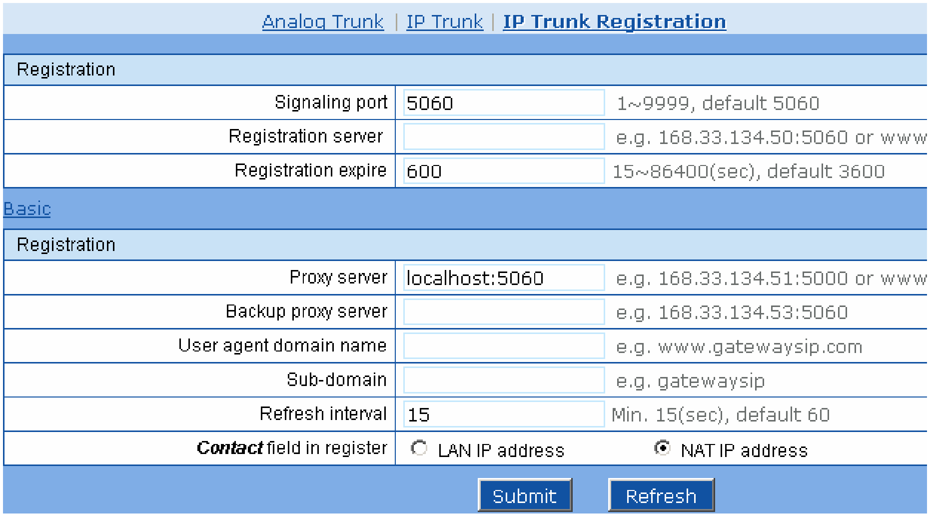


Table 2-13 Registration configuration parameter of IP trunk

Name	Description
Signaling Port	Configure the SIP port, 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 UDP port numbers used by the equipment.
Registration Server	Configure the address and port number of SIP registration server, and the address and port number are separated by “:”. It has no default value. The registration 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”.
Registration expire	Valid time of SIP re-registration in second.
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”.

Name	Description
Backup Server	Configure the IP address and port number of backup proxy server. It has no default value. Add the address of calling proxy server here, and the OM can support selection function of multiple softswitch addresses through IP address. The format must IP address format and complete and effective configuration, eg. “202.202.2.202:2727”. The proxy and registration servers must be identical. Conditions for falling over to the backup proxy server (any): 1）The OM registration is timeout; 2）No response to master server calls is timeout.
User Agent Domain Name	This domain name will used in INVITE messages. If it is not set here, the OM will use the IP address or domain name of IMS core network portal as user agent domain name. It has no default value. It is not recommended to use LAN IP address to set domain name parameter.
Sub-domain	Used for registered domain name, it is needed to configure only when the domain name used for registration and the user agent domain name are inconsistent. If it is null, the user agent domain name will be used for registration.
Refresh Interval	Interval of sending null UDP packet to SIP registration server, sent null UDP packet is used to ensure constant NAT port.
Contact field in register	Address behind NAT used in the Contact header field of SIP messages sent.

2.6 **System**

2.6.1 Characteristics of trunk line

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

Figure 2-15 Trunk line characteristics configuraiton interface

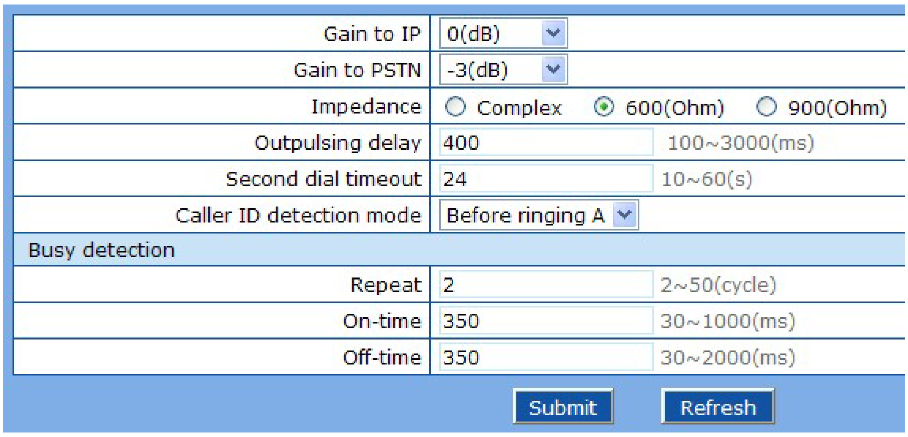


Table 2-14 Configuration parameter of trunk line characteristics

Name	Description
Gain to IP	Set the voice volume 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 volume gain towarding PSTN side, the default is -3. Taking decibel as the unit, setting range is -6 ~ +9 decibels.
Impedance	Set the parameter of FXO line impedance, with the default of 600 ohm. The optional settings as below: ● Complex ● 600（ohm） ● 900（ohm）
Outplusing delay	The time interval between FXO going off-hook and starting outpulsing the first digit to PSTN. The default is 400 in milliseconds.
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.
Caller ID detection mode.	Select according to the characteristics of field lines. Before ringing A, after ringing A, before ringing B, after ringing B. Note: A and B here refer to two detection means provided within the equipment.
Busy Detection	
Repeat	OMs will regard the busy tone signal with the repeat times specified here as a valid 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.

2.6.2 Characteristics of line

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

Figure 2-16 Line characteristics configuration interface

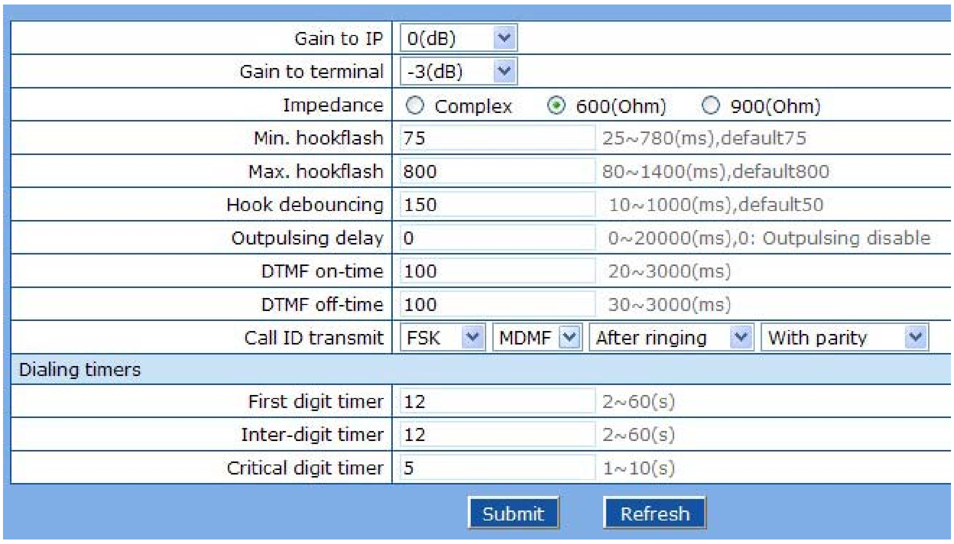


Table 2-15 Line characteristics configuration parameter

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.

Title	Explanation
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: ● Complex ● 600（ohm） ● 900（ohm）
Min.hookflash	Used to detect Hook Flash event, the default is 75 milliseconds. Any flash that fall short of the shortest flash time will be ignored. Generally, this value should not be less than 75 milliseconds.
Max. hookflash	Used to detect hook flash, the default is 800 milliseconds. The OM will regard the flash duration between “Min. hookflash” and “Max.hookflash” as an effective flash. Any flash lasting over the longest time will be considered as a valid hang up. Generally, this value should not be less than 800 milliseconds.
Hook debouncing	Used 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 OM will ignore the status variation, and consider the phone remains hang-up status. In case of vice versa, the OM will ignore the status variation, and consider the phone remains off hook status. Effective range of setting is 10~1000 milliseconds.
Outpulsing delay	Used when OMs’ FXS port is connected with the trunk interface of PBXs. For calls from OM to PBX, OMs will relay the extensions to PBX after the delay set here. Setting of “0” means no extension number relay. The default is 0 millisecond.
DTMF on-time	This parameter sets the on time (in ms) of DTMF signal sent from FXS 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 FXS port. The default value is 100 ms. Generally, the interval time should be set in the range of 80 ~ 150 ms.
Call ID transmit	Select transmission mode of Caller ID signal from the FXS port to the phone. ● FSK or DTMF; ● SDMF or MDMF; ● Sending Caller ID data before or after ringing; ● Sending Caller ID data with or without parity.
Dialing Timers	
First digit timer	If a user hasn’t dialed any number within a specified time by this parameter after offhook, the OMs will consider that the user has given up the call and prompt to hang up in busy tone. Unit: second, default value: 12 seconds.
Inter-digit timer	If a user hasn’t dialed the next number key from the time of dialing the last number key to the set time by this parameter, the OMs will consider that the user has ended dial-up and call out the dialed number. The default value is 12 seconds, unit: second.

Title	Explanation
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 user has dialed 021 and hasn’t dialed the next number within a set time by this parameter (eg. 5 seconds), the OMs will consider that the user has ended dial-up and call out the dialed number 021. The default value is 5 seconds, unit: second.

2.6.3 Greeting

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

Figure 2-17 Configuration interface of greeting

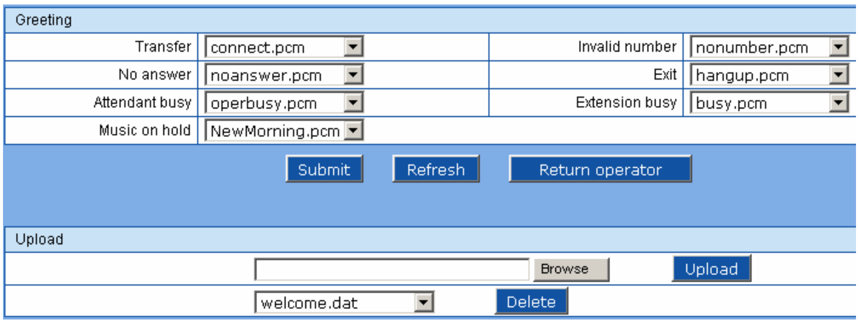


Table 2-16 Voice File Table Used on the OM

No (NN)	File Name	Prompt Usage	Prompt Content
00		Recording in buffer	
01	welcome	Greeting for working hours*	Thank you for calling. If you know your party's extension, please dial it now. Or, to transfer to an operator, press 0
02	offhour	Greeting for off hours	Thank you for calling. The office is closed. If you know the extension, please dial it now
03	operator	Transfer to operator	Please wait, while your call is transferred to an operator
04	operbusy	Operator in busy	Sorry, the call can not be answered at this time, please dial other numbers. Or, dial 0 for an operator
05	nonumber	Vacant or invalid number	Sorry, the number you dialed is not valid. Please check the number and dial it again
06	noanswer	No answer	Sorry, the call can not be answered at this time, please dial other numbers
07	busy	Busy line	Sorry, the line is busy at this time, please dial other numbers, or dial 0 for an operator

08	tryagain	Resource congestion	Sorry, your call can't be completed at this time, please hang up and try later
09	connect	Call transfer	Please wait, while your call is transferred
10	vm_all	Reminder for voice mail	Please leave a message after the tone
11	vm_busy	Reminder for voice mail	The extension is busy now, please leave a message after the tone
12	vm_noans	Reminder for voice mail	There is no one to answer the call, please leave a message after the tone
13	hangup	Disconnect a call	Sorry, your call will be disconnected. Thank you for calling
15	NewMorning	Background or CRBT	
	fring1	Background or CRBT	
	fring2	Background or CRBT	
	record	Reminder for recording	Start record after the tone

*: System default

2.6.4 Recording

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

Figure 2-18 Configuration interface of recording

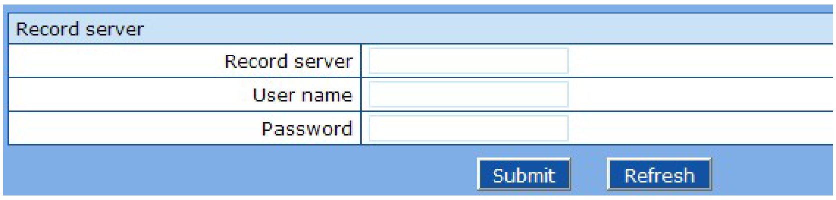
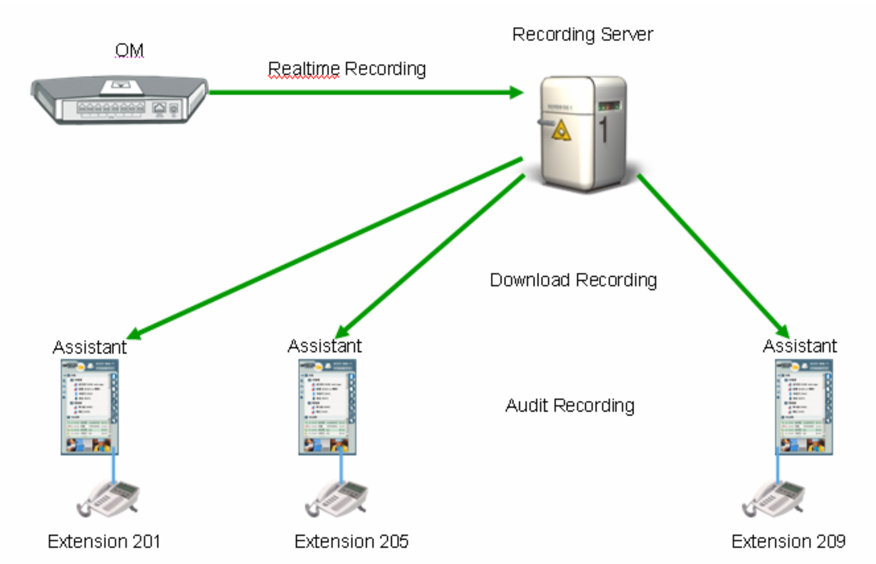


Table 2-17 Configuration parameter of recording

Name	Description
Address	Set the IP address and port number of recording storage server.
User Name	Set the ftp user name for logging on to the recording server. It is used for the assistant to download call record and voice mail, etc.
Password	Set the ftp password for logging on to the recording server. It is used for the assistant to download call record and voice mail, etc.

Figure 2-19   Framework of Recording



The in-built call recording function of OM can provide reliable and safe phone call recording for corporate users. The recording function consists of OM's in-built recording software module, external audio processing software and recording storage server, and the OM is connected to the recording storage server via LAN.

In-built recording software module: Send voices of the two parties to the recording storage server in the form of IP packet in batch;

Audio processing software: It (omrecord.exe) is an application software running on Windows XP in the background. Its main functions include receiving recording packets sent by the OM, enabling code conversion (convert G.711 or G.729 into MP3) and storing the recording in a time-dependent way.

Recording storage server: Windows XP system, used for recording packet processing and storage.

CAUTION

The recording software is only used to store call recording in MP3, by time and number. System administrator is responsible for the management of recorded files.

The recording can be played back with any MP3 player, such as Windows Media Player.

For more details about recording, see Appendix 3.2.

2.6.5 Call Record

After login, click "System > Call Record" tab to open the configuration interface.

Figure 2-20 Configuration interface of call record

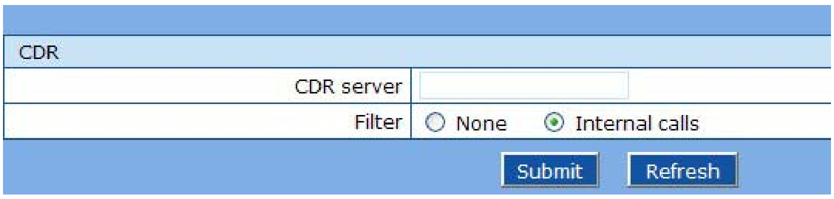


Table 2-18 Configuration parameter of call record

Name	Description
CDR Server	Set the IP address and port number of server for storing call records. Note: If port number is not configured, the default port number 1809 will be used.
Filter	Select what calls are not recorded: None or Internal calls.

2.6.6 Functional keys

The function key consists of system function key and service function key. The system function key is used for acquiring OM information, and the later is used for users to activate, inactivate extension features.

After login, click the label of “System> 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 2-21 Functional keys configuration interface

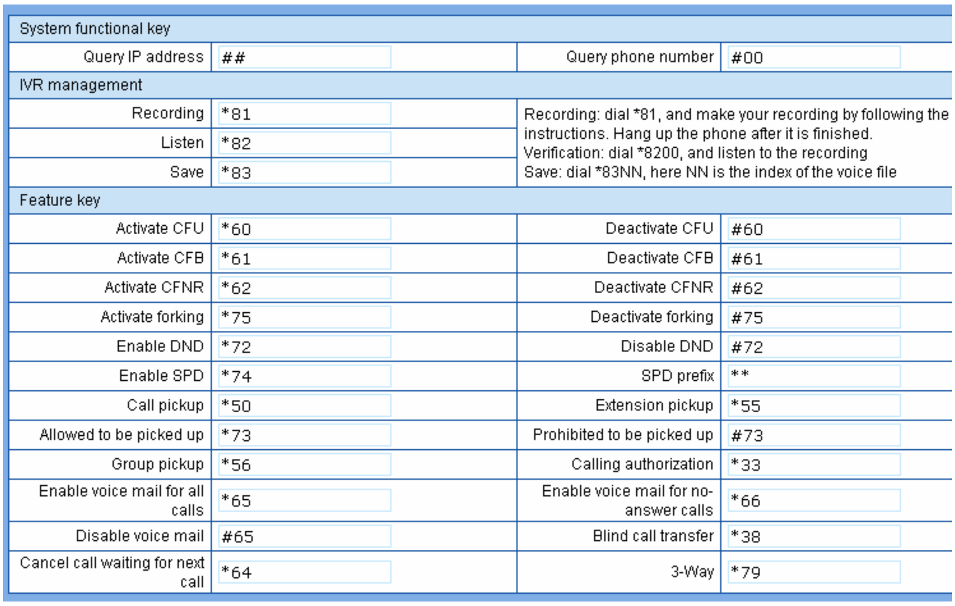


Table 2-19 The use of feature codes

System Codes	
Inquiry of IP address	Pick up the phone on any extension + ##
Inquiry of extension number	Pick up the phone of the extension + #00
Making Recording of Voice files	
Start recording	Pick up the phone of any extension + *81 + hang up after recording
Replay	Pick up the phone + *82

Save	Pick up the phone + *83NN Note: NN is the index of the voice files (see Appendix 4 for the list of all voice files)
Feature Codes	
Activate call forwarding variable (CFU)	Forwarding calls to an extension: Pick up the phone + *60 + the extension number of the third party Forwarding calls to an external number:: Pick up the phone + *60 + the prefix of selecting trunk + *the number of the new destination
Verify the destination number of CFU	Pick up the phone + *60*
Deactivate CFU	Pick up the phone + #60
Activate call forwarding on busy (CFB)	Forwarding calls to an extension: Pick up the phone + *61 + the extension number of the third party Forwarding calls to external number: Pick up the phone + *61 + the prefix of the selecting trunk + *the number of the new destination
Verify the destination number of CFB	Pick up the phone + *61*
Deactivate CFB	Pick up the phone + #61
Activate call forwarding on no answer (CFNR)	Forwarding calls to an extension: Pick up the phone + *62 + the extension of the third party Forwarding calls to an external number: Pick up the phone + *62 + prefix of selecting trunk + *the external number of the third party
Verify the destination number of CFNR	Pick up the phone + *62*
Deactivate CFNR	Pick up the phone + #62
Activate call splitting	Forking calls to an extension: Pick up the phone+ *75 + the external number Forking calls to an external number: Pick up the phone + *75 + prefix of selecting trunk + *the external number
Verify the phone number of the second terminal	Pick up the phone + *75*
Deactivate the call splitting	Pick up the phone + #75
Activate do not disturb (DND)	Pick up the phone + *72
Deactivate DND	Pick up the phone+ #72
Program the speed dial (SD) list	Pick up the phone+ *74 + MM + the original number Note: MM is in the range of 20~49
Listen to the SD list	Pick up the phone + *74 + MM + *

Delete SD list	Pick up the phone + *74 + MM
Dial SD	Pick up the phone + ** + MM
Pick up the call to an operator	Pick up the phone + *50
Pick up the call to an extension	Pick up the phone + *55 + the extension number
Allowed calls on the extension being picked up	Pick up the phone + *73
Inhibiting calls on the extension being picked up	Pick up the phone + #73
Group pickup	Pick up the phone + *56.
Calling authorization	Pick up the phone + *33 + the extension number + authorization code + the called party number Note: there should be no outbound calling restriction on this extension
Enable voice mail for all calls	Pick up the phone + *65.
Enable voice mail for no-answer calls	Pick up the phone + *66.
Disable voice mail	Pick up the phone + #65.
Listen to the color ring back tone (CRBT)	Pick up the phone + *88 + the two-digit index of the ring file
Enable CRBT	Pick up the phone + *80 + the two-digit index of the ring file
Deactivate CRBT	Pick up the phone + #80
Three-way calling	During the two-way conversation, hook-flash + *79 + the third party's number + hook-flash again after in talk with the third party
Disable call waiting for the next call	Pick up the phone+ *64
Blinded call transfer	During conversation with the first party, hook-flash + *38 + the extension number of the second party + hang up
Attended call transfer	D During conversation with the first party, hook-flash + the extension number + hang up

2.6.7 Tones

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

Figure 2-22 Configuration interface of tones

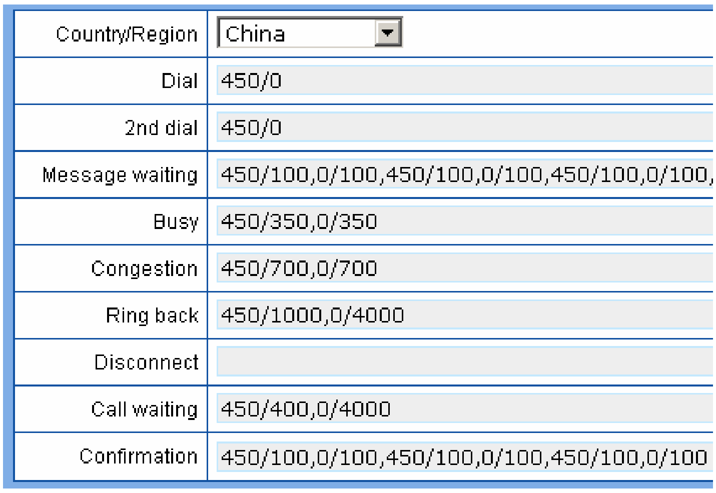


Table 2-20 Configuration parameter of tones

Name	Description
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 plan for the following countries and regions: China; the United States; France; Italy; Germany; Mexico; Chile; Russia; Japan; 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
Call waiting	Used for notification in call waiting
Confirmation	Used for confirming function keys being entered.

2.7 Advanced Configuration

2.7.1 System

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

Figure 2-23 Inferface of system advanced configuraiton

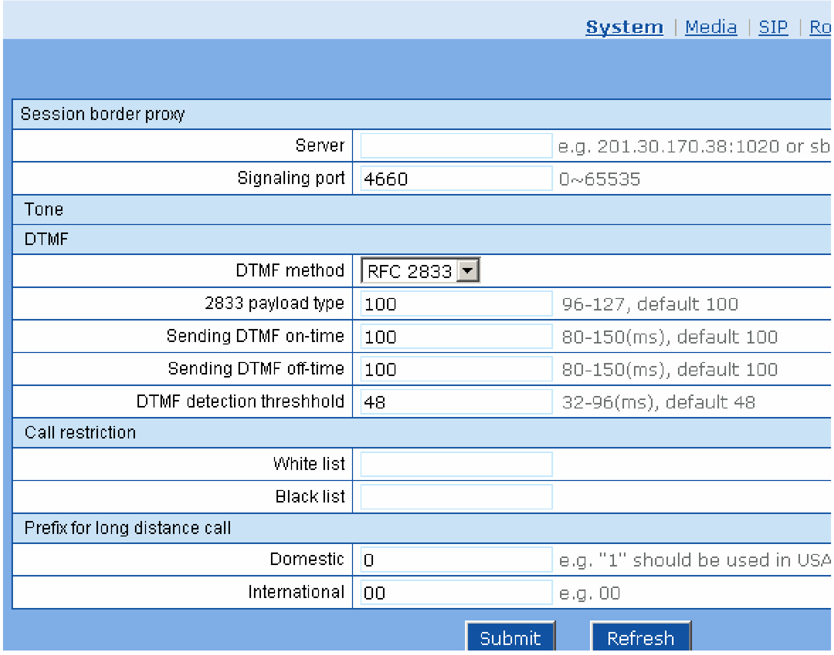


Table 2-21 Parameters of system advanced configuration

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. Examples: "201.30.170.38:5060" and "softswitch.com:5060".
Signaling port	Signaling port value of the OM，the default value is 4660. Signaling port number could be set at will, but can not conflict with other ports of equipment.
DTMF	
DTMF method	Transmission modes of DTMF signal supported by the OMs 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.
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 (e.g. platform).
Sending 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.

Title	Explanation
Sending 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 threshold	Minimum duration time of effective DTMF signal. Its effective range is 32-96 ms and the default value is 48 ms. The greater the value is set, the more stringent the detection is.
Call restriction	
White list	This parameter is used for setting a trunk number or prefix always allowed to call, such as emergency number (110), and even extensions without permission to call a trunk can call this number (number range). It is generally used to set emergency numbers such as 110|119|120.
Black list	This parameter is used for setting a trunk number or prefix prohibited to call, such as charge number range (900), and even extensions with permission to make international calls cannot call this number range. It is generally used to set numbers prohibited to call in any case. Examples: 900|17909.
Prefix for long distance call	
Domestic	This parameter is used for setting a trunk number or prefix prohibited to call domestically, and even extensions with permission to make domestic calls cannot call this number range. The default value is 00, indicating numbers starting with 00 are prohibited to call.
International	This parameter is used for setting a trunk number or prefix prohibited to call internationally, and even extensions with permission to make international calls cannot call this number range. The default value is null but can be set to codes of some countries, indicating numbers in these countries are prohibited to call.

2.7.2 Media

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

Figure 2-24 Configuration interface of media

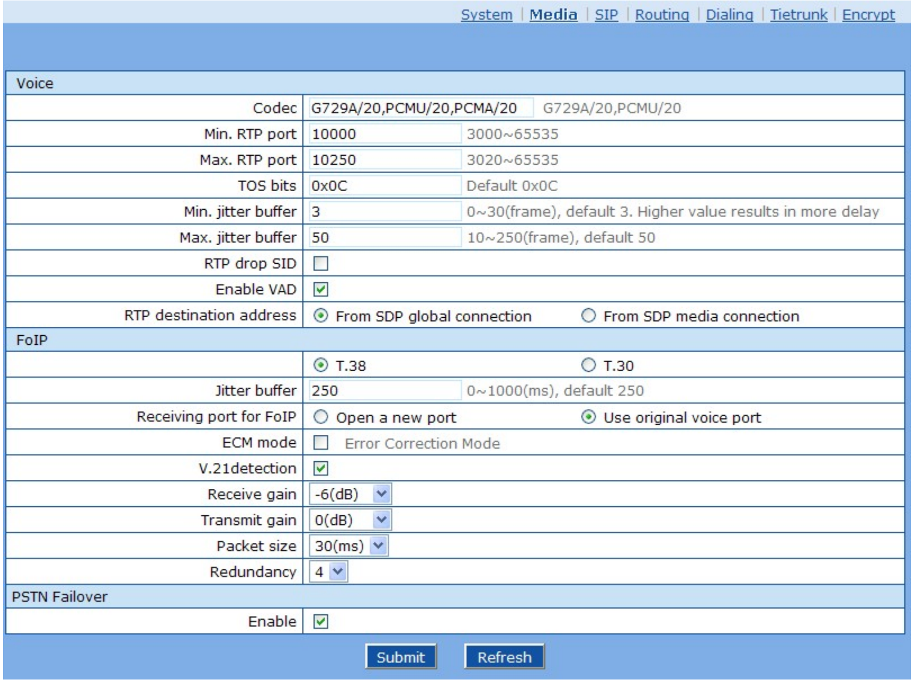


Table 2-22 Configuration parameter of media

Name	Description
Voice	
Codec	Codec supported by the OM include G729A/20, PCMU/20, PCMA/20 (as shown in Table 2-23). This parameter must be set due to no default value. Several encoding methods can configure in this item at the same time, separated with "," in the middle; the OM will negotiate with the platform in the order from front to back when configuring the codec methods
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.
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.
TOS bits	This parameter specifies the quality assurance of services with different priorities. The factory setting is 0x0C. For the mapping between the levels that has no reliability requirement and the TOS value, see Table 2-24.
Min. Jitter buffer	RTP Jitter Buffer is constructed to reduce the influence brought by network jitter. This default value is 3. Higher value of this parameter could result in longer delay; therefore, it should be set with caution.
Max. Jitter buffer	RTP Jitter Buffer helps to reduce the influence brought by network jitter. The default value is 50.

Name	Description
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.
RTP destination address	The default value is from SDP media connection. ● From SDP global connection: Obtain the peer IP address from SDP global connection; ● From SDP media connection: Obtain the peer IP address from Connection Information behind SDP Media Description.
FoIP	
	Select fax mode: T.38 or T.30
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 OM is switching to T.38 mode, and by default, original voice port will be used. ● 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.
V21 Signal detection	Select if V21 signal detection is used when T.38 faxes. The default value is “Not Select”.
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 milliseconds is the default value.
Redundancy	Set the number of the redundant frames in T.38 data package, default is 4.
PSTN failover	
Network fault routed to analog trunk	In the case of network fault, outgoing calls are automatically routed to FXO port. The default value is “Select”.

Table 2-23 Codec Methods Supported by OM

Codec Supported by OM	Bit Rate (Kbit/s)	Time Intervals of RTP Package Sending (ms)
G729A	8	10/20/30/40
PCMU/PCMA	64	10/20/30/40

Table 2-24 Mapping between the reliability requirement and the TOS value

Level	TOS Value (High Delay and High Throughput)	TOS Value (Low Delay and Low Throughput)	TOS Value (Low Delay and High Throughput)
0	0x08	0x10	0x18
1	0x28	0x30	0x38
2	0x48	0x50	0x58
3	0x68	0x70	0x78

Level	TOS Value (High Delay and High Throughput)	TOS Value (Low Delay and Low Throughput)	TOS Value (Low Delay and High Throughput)
4	0x88	0x90	0x98
5	0xA8	0xB0	0xB8
6	0xC8	0xD0	0xD8
7	0xE8	0xF0	0xF8

2.7.3 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 mainly 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.

OMs 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 2-25 SIP related configuration interface

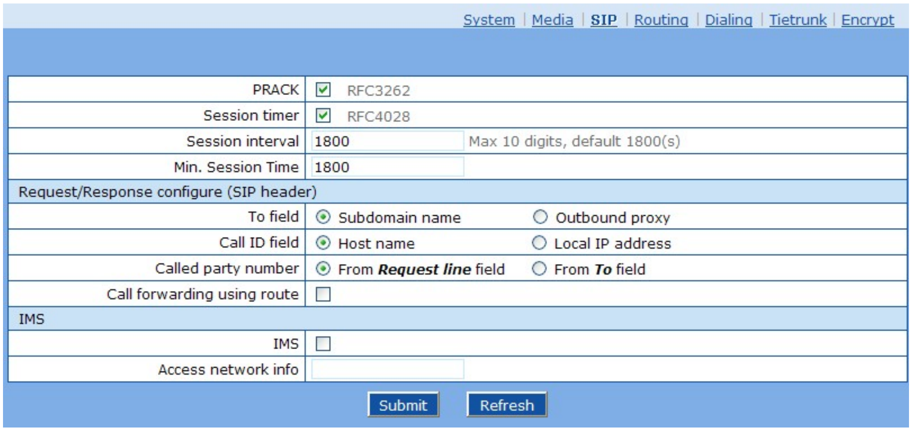


Table 2-25 Configuration parameter of SIP related

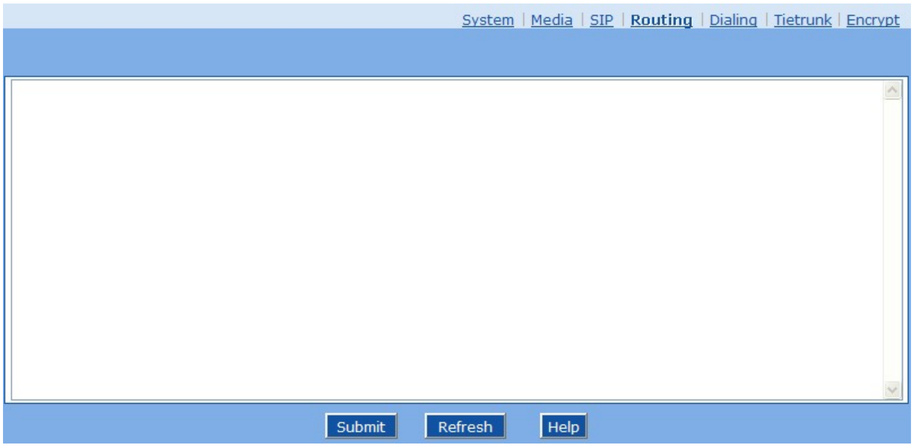
Name	Description
PRACK	Determine whether to activate Reliable Provisional Responses. (RFC 3262)
Session timer	Choose to activate session refresh (Session Timer, RFC 4028). By default, session timer is not activated.
Session interval	Set the session refresh interval. The value of Session-Expires is enclosed into INVITE or UPDATE messages. Default value is 1800 in second.
Minimum timer	Set the minimum value of session refresh interval.

Name	Description
Request/Response Configure	
To field	Choose whether to apply Domain name or Outbound proxy to “To” header field, the default is “Domain name”.
Address in 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 OM acquires the called number from Request Line header field or To header field. The default is “from Request Line”.
302 Contact with switched prefix	If activated: When sending 302 call forwarding, transferred number is required to switch over the routed number, or it will be used directly.
IMS	
IMS	Activate SIP function compatible with 3GPP IMS standard.
Access network information	Set the header field value of P-Access-network-info. The OM will include the header field of P-Access-network-info in SUBSCRIBE and INVITE message.

2.7.4 Routing Table

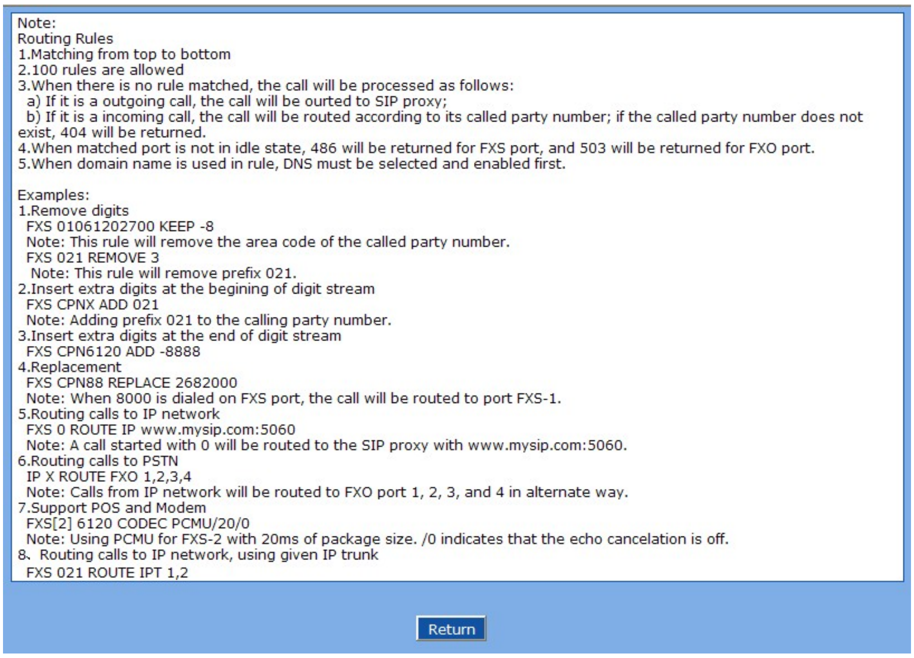
After login, click “Advanced > Routing Table” tab to open the configuration interface.

Figure 2-26 Configuration Interface for Routing Table



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

Figure 2-27 Illustrative interface for routing configuration



The routing table with 50 rules in capacity provides two functions including digit transformation and call routing assignment. Here are the general rules applied by OMs when executing the routing table:

1. The routing rules in the table are executed from top to down, and the number matching follows the principle of minimum & priority matching;
2. When there is no rule matched, the call will be processed as follows:
   a) For outbound calls, calls will be routed to SIP proxy;
   b) For inbound calls, calls will be routed according to its called party number, and if the called party number does not exist, 404 will be returned.
3. When matched port is not in idle state, 486 will be returned for FXS port, and 503 will be returned for FXO port.
4. When domain name is used in rule, DNS must be selected and enabled first.

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 submit.

The routing table is empty by default. The OM 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 021 REMOVE 3” means removing the prefix 021 of the called number for calls from FXS port, where “FXS” is source, “021” is number, and “REMOVE 3” 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 methods and routing destination are shown below.

Table 2-26 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, e.g. FXS1, FXO2 or FXS [1-2], etc.
Number	It could 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.114，61202700; ● Designate a number matching a prefix: such as 61xxxxxx. Note: the matching effect of 61xxxxxx is different from that of 61x or 61. ● 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 at least two-digit; 12x matches any number with at least 3-digit starting with 12.

Table 2-27 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 02161202700 KEEP -8 Keep the last 8 digits of the called number 02161202700 for calls from FXS. The transformed called number is 61202700.
REMOVE	Remove number. The positive number behind REMOVE means to remove several digits in the front of the number; the negative number means to remove several digits at the end of the number. For example: FXS 021 REMOVE 3 Remove 021 of the called number beginning with 021 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 021 FXS2 CPNX ADD 010 Add 021 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, replacing 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 call s 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 call will be routed according to the default routing when END is met, or the call will be routed according to the designed routing after ROUTE is met. 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 is being sending out.

Table 2-28 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 1: IP 800[0-3] ROUTE FXS 1,2,3,4 Select FXS port in a sequential order. 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 simultaneously.
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.

2.7.5 Application Examples of Routing Table

1) Remove digits

   FXS 01061202700 KEEP -8

   Note: This rule will remove the area code of the called party number.

   FXS 021 REMOVE 3

   Note: This rule will remove prefix 021.

2) Insert extra digits at the beginning of digit stream

   FXS CPNX ADD 021

   Note: Adding prefix 021 to the calling party number.

3) Insert extra digits at the end of digit stream

   FXS CPN6120 ADD -8888

4) Replacement

FXS CPN88 REPLACE 2682000

Note: When 8000 is dialed on FXS port, the call will be routed to port FXS-1.

5) Routing calls to IP network

FXS 0 ROUTE IP www.mysip.com:5060

Note: A call started with 0 will be routed to the SIP proxy with www.mysip.com:5060.

6) Routing calls to PSTN

IP X ROUTE FXO 1,2,3,4

Note: Calls from IP network will be routed to FXO port 1, 2, 3, and 4 in alternate way.

7) Support POS and Modem

FXS[2] 6120 CODEC PCMU/20/0

Note: Using PCMU for FXS-2 with 20ms of package size. /0 indicates that the echo cancellation is off.

8) Routing calls to IP network, using given IP trunk

FXS 021 ROUTE IPT 1,2

9) Add prefix

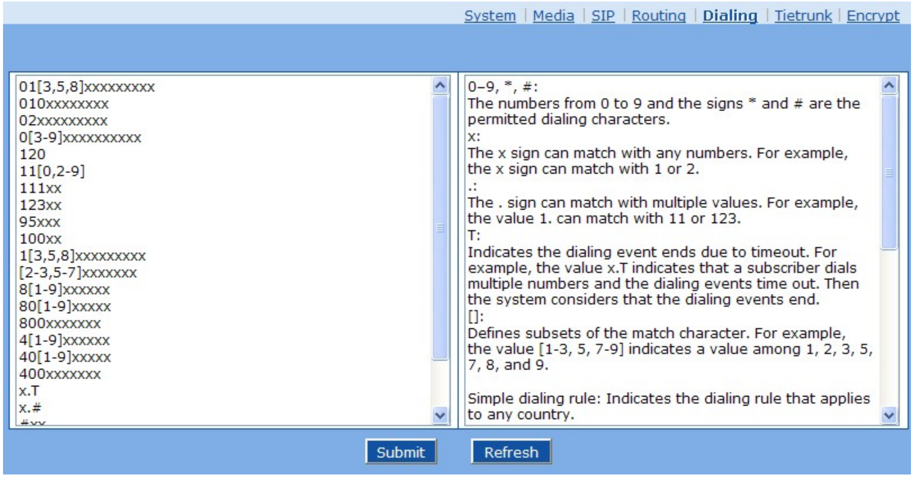
FXS CPNX ADD 021

Note: 021 Add 021 before the calling number when an extension is used to call it out.

2.7.6 Digit Map

After login, click “Advanced > Dialing” tab to open the dialing rules interface.

Figure 2-28 Configuration Interface for Digit Map



Digit map is 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 digit map can help to reduce the connection time of telephone calls.

The maximum number of digit map that can be stored in OM is 60. Each rule can hold not more than 32 numbers and 38 characters. The total length of digit map table (the total length of all digit map) cannot be more than 2280 bytes.

The default typical rules include constituent rules of domestic phone numbers and function keys used by the OM. Users can add new rules as required. Basic symbols in typical rules are as follows. The following table provides a description of typical rules:

Basic symbols:

- 0–9, *, #: the numbers from 0 to 9 and the signs * and # are the permitted dialing characters.
- x: the x sign can match with any numbers. For example, the x sign can match with 1 or 2.
- .: the "." sign can match with multiple values. For example, the value 1. can match with 11 or 123.
- []: defines subsets of the match character. For example, the value [1-3, 5, 7-9] indicates a value among 1, 2, 3, 5, 7, 8, and 9.
- T: indicates the dialing event ends due to timeout. For example, the value x.T indicates that a user dials multiple numbers and the dialing events time out. Then the system considers that the dialing events end.

Table 2-29 Description of Digit map

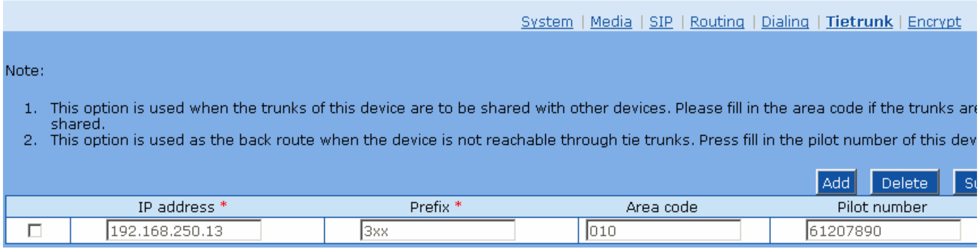
Digit map	Description
"##"	End after receiving two-digit dialing "##". "##" is an special dialing for users to query IP address and version number of firmware by default.
"x.T"	The OMs will detect any length of telephone number starting with any number between 0-9. The OMs will send the detected number 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 users press # key after dial-up, the OMs 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 extension features, 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 extension features, 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	An 11-digit number starting with 02, used to end the dialing starting with "02".
13xxxxxxxxx	An 11-digit number starting with 13, used to end the dialing.
11x	A 3-digit number starting with 11, used to end the special service calls.
9xxxx	A 5-digit number starting with 9, used to end special calls.

2.7.7 Tie trunk

The tie trunk of OM is an IP trunk based on SIP protocol, and it is used to connect more than two OMs via IP network. Abbreviated dialing, call transfer and other functions among different extensions of the OM can be achieved in such an OM group. Tie trunk is usually used for: 1) corporate non-local tie trunk; 2) system expansion. A specific example is used to explain the application of tie trunk following the introduction of parameter configuration of tie trunk below.

After login, click "Advanced > Tie trunk" tab to open the configuration interface.

Figure 2-29 Configuration interface of tie trunk

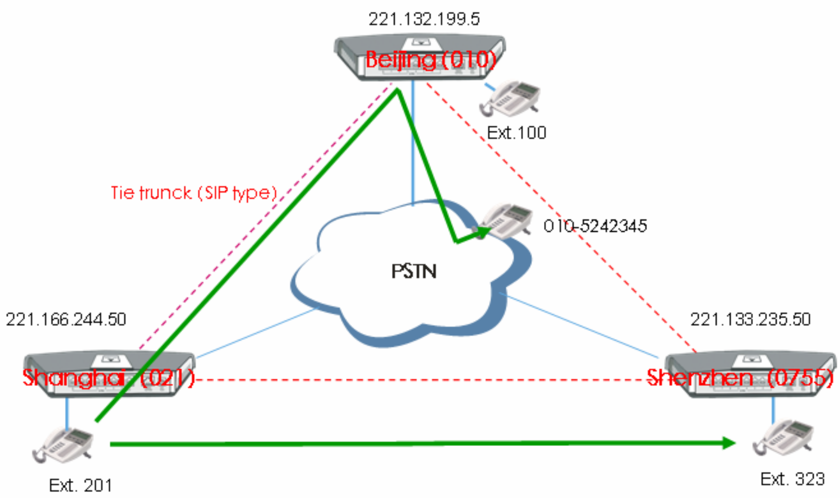


Click “Add” to add the devices.

Table 2-30 Configuration parameter of tie trunk

Name	Description
IP address	Enter the IP address of this OM tie trunk.
Prefix	Extension numbers are required to plan overall when the tie trunk is used. An extension prefix is the prefix part of an extension number, and each OM has a unique extension prefix; a number created like this is unique in the whole network; an extension prefix can not be “0” or used as an added dial of outbound call or front desk.
Area code	An area code is the regional code of fixed-line telephone network in a place where OM is located; for example, “021”is the zip code in Shanghai.
Pilot number	Trunk numbers are external numbers of the OM (such as operator number)

Figure 2-30 Reference Tie Trunk Diagram



Explanation:

1) Dial “323” directly if the extension 201 in Shanghai is used to dial the extension 323 in Shenzhen.

2) Dial “010-52452345”directly if the extension 201 in Shanghai is used to dial the urban phone number 010-52452345 in Beijing.

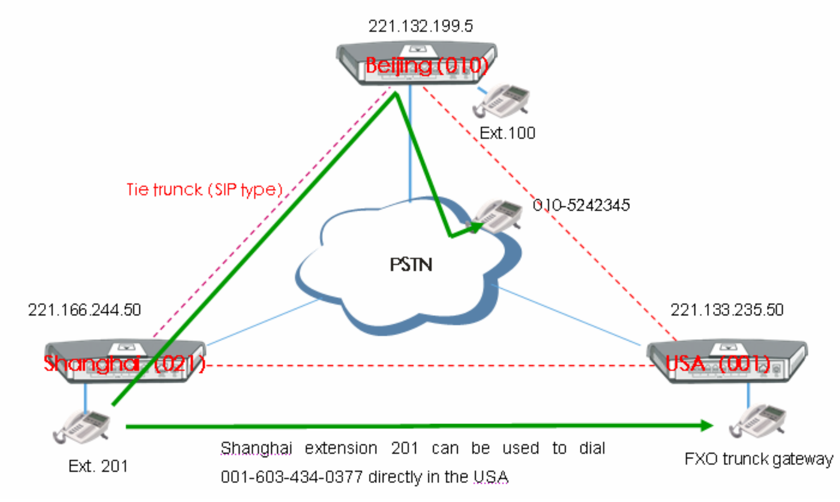
Table 2-31 Intelligent Routing Table

Examples	Dialing and Routing Selection
Dial a Beijing extension in Shanghai	Dial the extension number directly, and it will be connected to an extension of the OM in Beijing through tie trunk;
Dial a fixed Beijing phone number (010) in Shanghai	Dial the Beijing number directly, and it will be outbound from Beijing OM through tie trunk;
Dial a fixed Beijing phone number in Shanghai, but it is outbound from Shanghai.	Dial “9” and then the Beijing number;
Dial a fixed Jiangsu phone number in Shanghai	Dial this number directly and it is outbound from the local OM; or dial “9” and then this number.
Dial a fixed Jiangsu phone number in Shanghai, but it is outbound from Shanghai.	Dial “9” and then the Jiangsu number;
Dial a mobile phone in Shanghai	Dial “9” and then the mobile phone number;
Dial a Beijing extension in Shanghai, but the tie trunk is not accessed.	Dial the extension number directly and it is connected to Beijing OM after being outbound locally, then re-dial it after receiving the voice prompt;
Dialing Rules: ● Dial all extensions directly (including local and non-local extension numbers); ● Add “9” when dialing outbound trunks (Note: The prefix can be set); ● Long-distance fixed phone numbers (starting with “0”) can be dialed directly, or add ”9” to dial out; ● “9” should always be dialed for mobile phone, or it will be treated as extension if ‘9” is not dialed.	

Table 2-32 Trunk Routing Table of OM Reference Configuration

Prefix	Area Code	IP Address	Pilot Number
2	021	221.166.244.50	61202700
1	010	221.132.199.50	54025012
3	0755	221.133.235.50	78404122

Figure 2-31 Tie Trunk Application Diagram in OM



2.7.8 Encrypt

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

Figure 2-32 Encrypt configuration interface

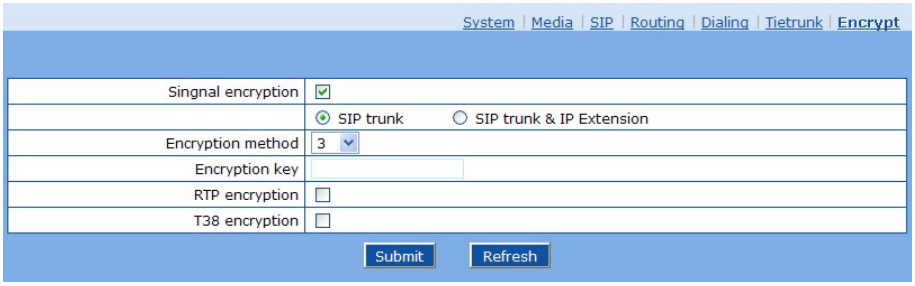


Table 2-33 Encrypt configuration parameters

Name	Description
Signaling encryption	Choose whether to encrypt signaling. By default, this is not selected.

Name	Description
Encryption method	Select the encryption method, and default is 7. The optional parameters as below: ● 2: TCP Not Encrypted； ● 3: TCP Encrypted； ● 6: UDP Not Encrypted； ● 7: UDP Encrypted（New Rock）； ● 8: Using Keyword； ● 9: Using Keyword2； ● 10: RC4； ● 11: Using Keyword 3； ● 12: Encrypt12； ● 13: Encrypt13； ● 14: Encrypt14（New Rock）； ● 16: Word Reverse； ● 17: Word Exchange（263）； ● 18: Byte Reverse； ● 19: Byte Exchange.
Encryption key	You may obtain it from service provider
RTP encryption	Choose whether to encrypt RTP voice pack, the default is “ not to activate”
T.38 encryption	Choose whether to encrypt T38 data. By default, this is not selected.

2.8 Log management

2.8.1 Resource status

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

1) The information about login of 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.

Figure 2-33 System Status Interface

Table 2-34 Parameters of system status

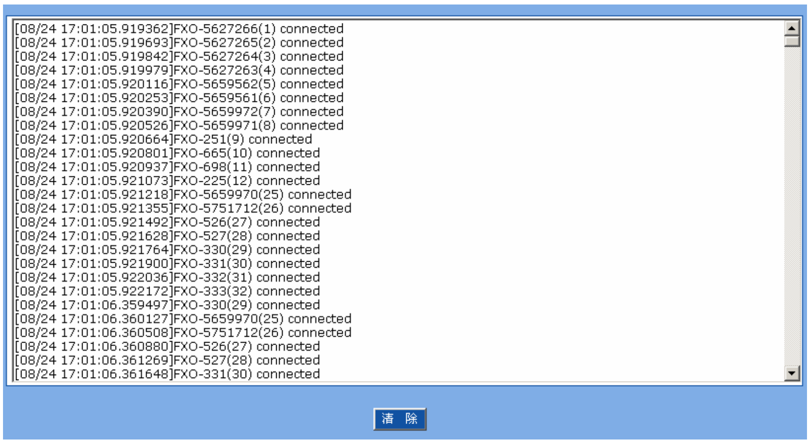
Title	Explanation
Login User Info	Show the IP address and jurisdiction of login user. The numbers following the IP address show the online jurisdiction of the user: 1- administrator; 2 - operator; 3 - viewer. The viewer can only read the configuration, but is not allowed to modify it. When more than 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: ● Not enabled: The registration server's address is not entered yet; ● Latest response: The latest response message for the registration. 200 means registered successfully； ● No response: Not received response from registration 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@220.248.27.70:1003；user=phone> latest response: 200 （timeout-555） Contact: <sip:2681402@220.248.27.70:1003；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.

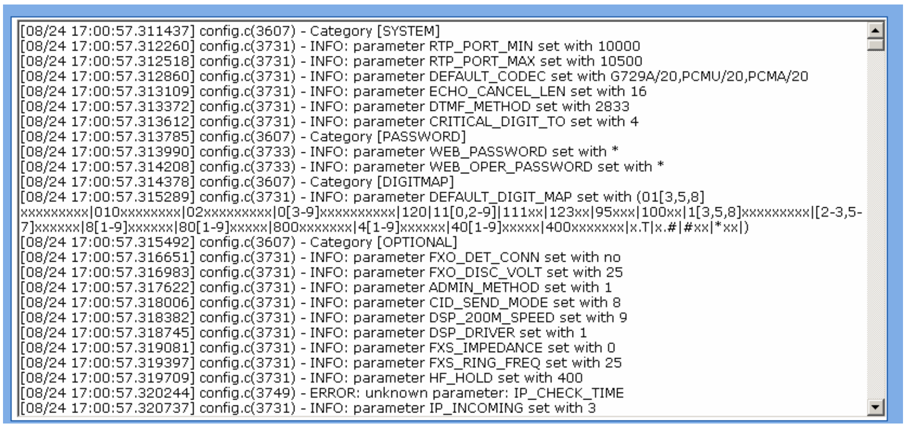
Figure 2-34 Call message interface



2.8.3 System Startup

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

Figure 2-35 Interface of system startup



2.8.4 Configure log

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

Figure 2-36 Interface of debugging log management

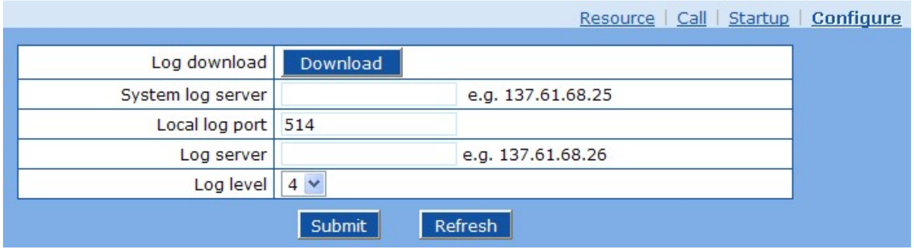


Table 2-35 Configuration parameters of debugging log management

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 OM, 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 OM is used in normal operation, avoiding influencing the system performance.

Procedure of downloading the debugging log:

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

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

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

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

2.9 System tool

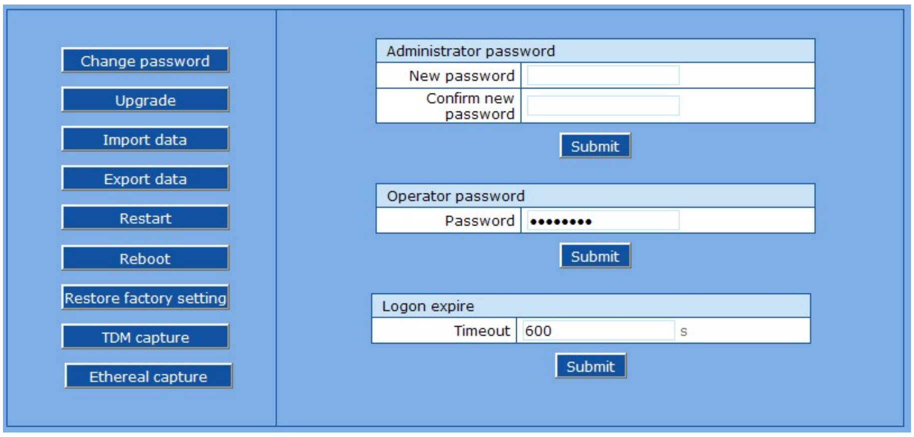
2.9.1 Change password

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 2-37 Interface of password changing



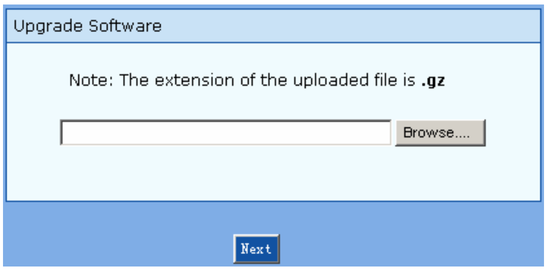
2.9.2 Software upgrade

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.

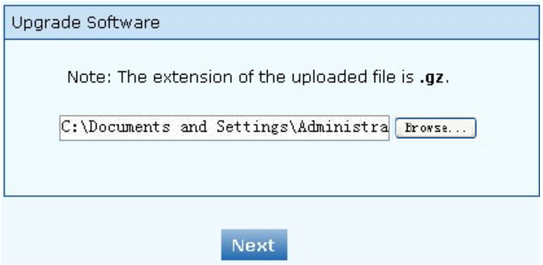
Figure 2-38 Interface of software upgrade



Step 3 Click "Browse" to select the upgrade files and click "Open".

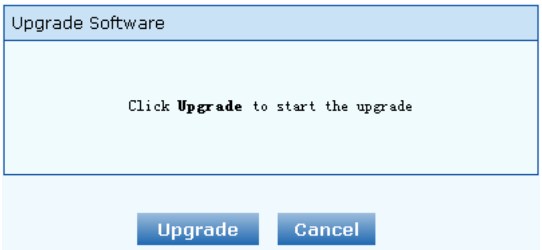
Step 4 Click "Next" when the following interface appears, and start uploading the upgrade files to the OM.

Figure 2-39 Interface of file upload



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

Figure 2-40 Upgrade interface



Step 6 The following prompt appears during the upgrade.

Figure 2-41 Prompt of upgrade process

WARNING

A few minutes are needed to upgrade the OM. Don't operate the OM during this period.

Step 7 After success in upgrade, the following dialog will appear, click "Confirm".

Figure 2-42 Interface of successful upgrade

Step 8 The OM will reboot, and the interface will be disappeared.

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

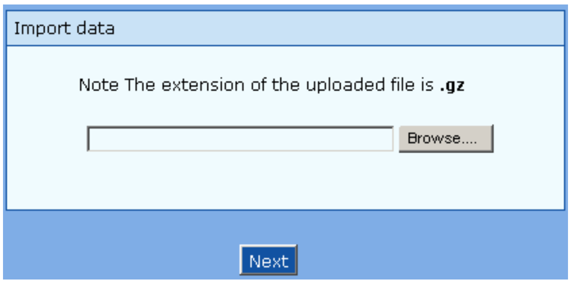
WARNING

ForOM100 and OM200, the software upgrade operation must be conducted on an 100M Ethernet port.

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".

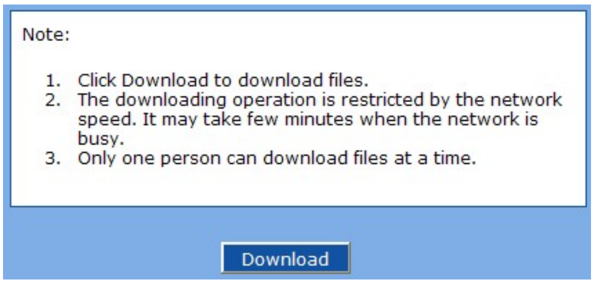
Figure 2-43 Interface of import data



2.9.4 Configuration export

After login, click “Tools >Export of configuration” to open this interface. It’s allowed to download the configuration files from the OM through this interface. The downloading procedure is similar to the downloading procedure of log files.

Figure 2-44 Interface of export data



2.9.5 Software restart

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

⚠ CAUTION

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

2.9.6 System reboot

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

⚠ CAUTION

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

2.9.7 Restore factory settings

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

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

2.9.8 TDM Capture

After login, click “Tools > TDM Capture” to open the capture interface.….

This tool is used to capture the received TDM signal on a FXS or FXO port. The maximum length of the capture is 200 seconds. The capture is stored in the flash memory of the device in TDM format, which is readable with reader such as CoolEditor.

Capture process:

Step 1 Selecting by clicking , or manually enter the port from which you want to capture the TDM signal.

Step 2 Click “Start” to begin the capture process.

Step 3 Click “Stop” to end the capture. You will be prompt to download. Download the capture file to the designated PC.

2.9.9 Ethereal capture

After login, click “Tools > Ethereal capture” to open the interface of the tool.….

This tool is used to capture the tcpdump of the device. Up to 3 files each with Max. 3 M bytes in size can be generated. The captured files are stored on the flash memory of the device in dump.cap format, which is readable by Ethereal.

Steps for capture:

Step 1 Click “Start” to begin the capture process. You will be notified if there is a on-going capture already.

Step 2 Making the call which you want to capture.

Step 3 Click “Stop” to end the capture process. You will be notified for download. Download the file to the designated PC.

2.10 **Product information**

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

Figure 2-45 Configuration interface of product information

Product name	OM100L
Number of analog extensions	24
Number of analog trunks	8
Software version	Rev 2.0.5.59
Kernel version	Kernel 8.5.5.2a (F)
DSP version	Rev 1.8.195 (0x2551)
Max. IP extension	120
Max. IP trunk	60
MAC	00:0E:A9:10:06:B2
Current system time	2009-11-10 15:13:32
Latest system reboot time	2009-11-09 13:30:44
More info	http://www.newrocktech.com/en/products/officium10.asp

2.11 Logout

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

3 Appendix

3.1 OM200 system operation state

Table 3-1 OM200 system operation state

Glittery letter	Status meaning
“C”	The IP address of OM conflicts with that of other equipment in LAN. Please settle this problem before the OM can be operated normally.
“D”	Internal failures have been encountered during OM start up procedure. Please contact your local distributor for further diagnosis.
“P”	The OM 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 OM has been exited. If it can not be restored by rebooting the system, please contact your local distributor for further diagnosis.

3.2 Setting of Recording Storage Server

Table 3-2 Hardware Requirements for Recording Storage Sever

Item	Minimum Configuration	Recommended Configuration
CPU	P4 2.4G	Xeon 3.0G (Dual core)
Memory	500M	1G
HD	According to the need (recording 170KB/min.)	According to the need (recording 170KB/min.)
Port (communicating with OM)	1311	1311
OS	Windows 2003/XP	Windows 2003/XP

Table 3-3 Specification of OM Recording

Item	Description
Voice Coding	Recording can operate normally when the OM selects G.711 or G.729A/B.
Recorded file format	MP3
Player	Users can play the calling and called voices through L&R sound channels
Performance	A maximum of 24-way recording can be done at one time.
Voice Monitoring	Not support

The installation steps of recording software are as follows:

Step 1 Copy OMRecord.exe, Install.cmd and Uninstall.cmd to the installation directory on Windows XP (storage server);

Step 2 Click to install Install.cmd. After installation, the recording software will run in the background by the name of “OMRecord”. It will run automatically after startup each time;

Step 3 Ensure that port 1311 on the storage server running OMRecord.exe has not been screened by the firewall.

Notes:

1) Recorded files (MP3 in format) are stored under the directory of omrecord.exe;
2) Naming format of recorded files: calling number_called number_day/month/year/-hour/minute/second.mp3.