PX Series IP PBX

User Manual



Table of contents

1 Product Overview....1
1.1 Product Introduction....1
1.2 Application Scene....1
1.2.1 Small and SOHO company....1
1.2.2 Small and medium enterprises....1
1.2.3 Hotel....2
1.2.4 Multi-building....2
1.2.5 Remote networking....3
1.3 Key Features....3
1.4 Voice Features....3
1.5 VoIP Protocol....3
1.6 Software Feature....4
1.7 PBX Feature....4
1.8 Management and Maintenance....4
1.9 Product Specification....5
2 Installation Guide....6
2.1 Device Interface....6
2.2 Device Indicator....6
2.3 Network Connection Dlagram....7
2.4 Login Web....7
2.5 Modify IP Address....8
3 Basic Operation....9
3.1 Make a call....9
3.2 Dial operator....9
3.3 Call Port Group....9
3.4 Incoming call....9
3.5 Make Outgoing Call....9
3.6 Call Pickup....9
3.7 Call on hold....9
3.8 Call Waiting....10
3.9 Call FORWARD....10
3.10 Flashing Hook Operation....10
3.11 Function Code....10
4 WEB Configuration....13
4.1 System Information....13
4.1.1 Equipment Information....13
4.1.2 Current Alarm....14
4.1.3 History Alarm....14
4.1.4 Operation Log....14
4.2 SIP User....14
4.3 Operator Setting....15
4.4 Port Group Setting....16

4.5 Enable/Disable Trunk Line 18
4.5.1 Analog Trunk 18
4.5.2 SIP Trunk 18
4.6 Day/Night Mode 19
4.7 CO line Incoming Call Processing 19
4.7.1 Ring Extension directly 19
4.7.2 Auto Attendant Mode(IVR) 20
4.7.3 CRBT Mode 20
4.8 Outgoing Call 21
4.9 Extension Access CO line authority 22
4.10 Call Duration Control 22
4.11 Outgoing call level setting 23
4.12 Call Forward Setting 24
4.13 Call Recording 25
4.14 Voice Mail 25
4.15 Alarm Clock 26
4.16 Networking Setting 26
4.17 Fxs Fxo Parameters 27
4.18 Media parameter 28
4.19 SIP Parameter 28
4.20 Fax Parameter 30
4.21 Dialing Rule 31
4.22 System Setting 31
4.23 Time Setting 32
4.24 Account Management 32
4.25 Device Management 33
4.26 Email Push 34
4.27 System Upgrade 35
4.28 License Upgrade 35
4.29 System Data Backup 35
4.30 Troubleshooting 36
4.31 System Voice 36
4.31.1 Music 37
4.31.2 CRBT 37
4.31.3 IVR Setting 38
5 Terminology 39

1 PRODUCT OVERVIEW

1.1 PRODUCT INTRODUCTION

CDX8000-PX series IPPBX is a PBX based on IP networks, providing a low-cost, easy-to-operate VoIP solution for government agencies, banking and finance, transportation and logistics, remote offices and multi-branch enterprises. On the one hand, it can connect with traditional telephone, fax and traditional analog PBX through the standard voice interface, and provide high-quality voice services. On the other hand, it uses the standard SIP protocol and can be compatible with the mainstream IPPBX, IMS soft switch platform and SIP-based network platform. It comes with an optical port, which can be stretched out by optical fiber, easily solving the difficult problem of construction deployment between multiple floors (multiple buildings). It supports intelligent recording recognition function, can automatically analyze the call recording (call content to text, emotion analysis, concern analysis, service attitude analysis, response speed analysis). It supports the secondary interface, can be used for secondary development.

1.2 APPLICATION SCENE

The following application scene is only for reference, and specific schemes should be implemented according to the actual needs of the project.

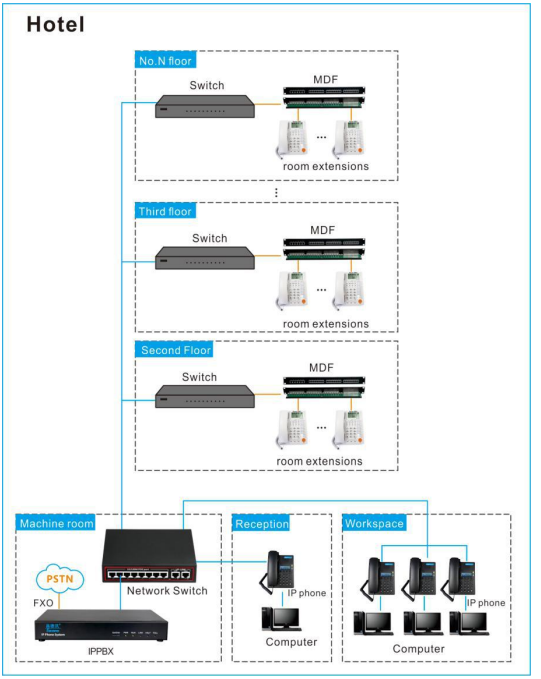
1.2.1 Small and SOHO company



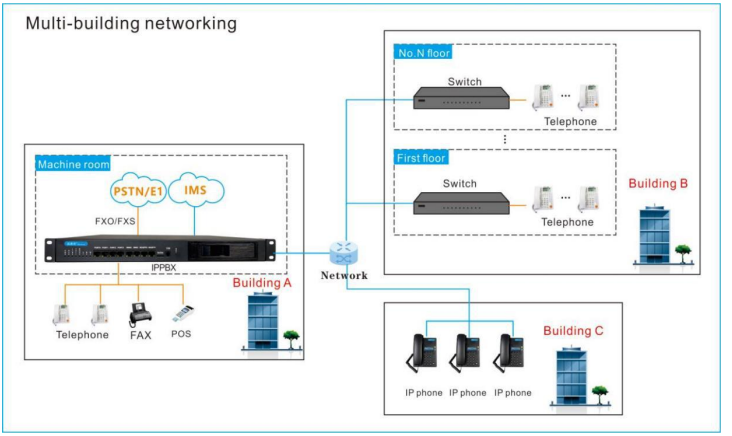
1.2.2 Small and medium enterprises



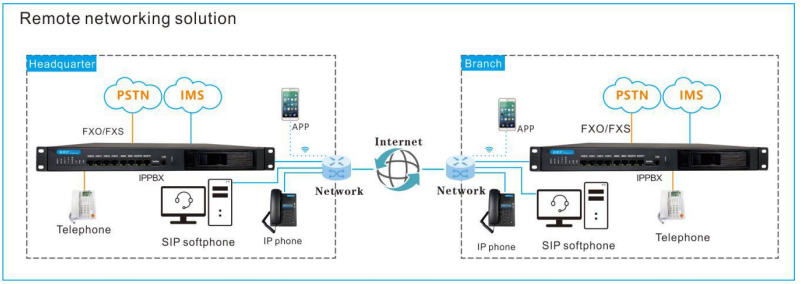
1.2.3 Hotel



1.2.4 Multi-building



1.2.5 Remote networking



1.3 KEY FEATURES

◆Multi-function, cost-effective, and support E1/FXO/FXS/SIP/SIP phones
◆Adapt to complex usage scenarios
◆Supports SNMP/TR069 standard network management protocols
◆Support both IPv4 and IPv6, facilitating future network upgrades
◆Support G.711, G.729, G.723, G.726, AMR and other codes
◆It adopts standard SIP protocol and is perfectly compatible with IMS/NGN and mainstream soft switch platforms

1.4 VOICE FEATURES

◆Voice codes: G.711a/ μlaw, G.723.1, G.729A/B, G.726
◆Silence suppression
◆Comfortable noise generation (CNG)
◆Voice Activity Detection (VAD)
◆Echo cancellation (G.168), maximum 128ms
◆Dynamic jitter cache
◆Flashing hook detection
◆Automatic gain control
◆Fax: T.38 and Pass-through
◆Modem/POS
◆DTMF mode: RFC2833/Signal/Inband
◆VLAN 802.1P/802.1Q

1.5 VOIP PROTOCOL

◆Protocol: SIP V2.0 (UDP/TCP), RFC3261, SDP, RTP(RFC2833), RFC3262, RFC3263,
◆RFC3264, RFC3265, RFC3515, RFC2976, RFC3311
◆RTP/RTCP, RFC2198, RFC1889
◆SIP over TLS

◆RFC4028 Session Timer
◆RFC3266 IPv6 in SDP
◆RFC2806 TEL URL1 Product Overview
◆RFC3581 NAT.rport
◆Active and standby SIP servers
◆External proxy server
◆DNS query /A query /NATPR query
◆SIP trunks can be created
◆Early media/early response

1.6 SOFTWARE FEATURE

◆Ring group
◆Web access control rules
◆Telnet access control rules
◆Action URL
◆IPv4/IPv6
◆Matching rule: digitmap
◆Voice bandwidth optimization, encryption, and compression
◆Flexible call routing policy
◆The calling and called number change function
◆CRBT function
◆Message Leaving function
◆Recording function
◆IVR function

1.7 PBX FEATURE

◆Call forwarding (directly/No answer/Busy)
◆Call waiting/Call hold
◆Call Transfer (directly transfer/Inquiry transfer)
◆Answer within the group
◆Hotline
◆Do not disturb
◆Three party conference
◆Morning call service
◆Alarm function

1.8 MANAGEMENT AND MAINTENANCE

◆Web Management configuration page
◆Automatic upgrade/configuration
◆SNMP V1/V2/V3

◆TR069
◆Configure backup/restore
◆Upgrade the HTTP/TFTP/FTP program
◆Query and export call CDRS
◆Querying and exporting syslogs
◆Ping/Tracert test
◆Line Diagnosis (GR909)
◆NTP or Daylight Saving Time
◆IVR voice maintenance
◆Centralized maintenance in the cloud
◆Remote Web function (reliable transmission, upload and download)

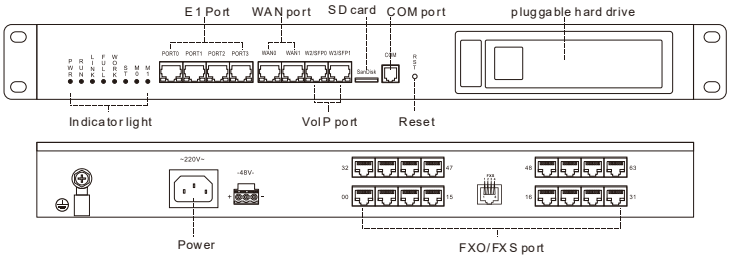
1.9 PRODUCT SPECIFICATION

Parameter	PX256	PX512	PX1024	PX400	PX800
SIP	256	512	1024	128	256
FXS+FXO	80	80	80	4FXO	8FXO
SIP Trunk	256	256	256	128	256
E1(NO. 7/PRI)	0	2	4	0	0
concurrent call	16	32	64	8	16
temperature(℃)	0-50	0-50	0-50	0-50	0-50
Weight	<7KG			<1KG	
power consumption	<50W			<10W	
Size	440mm*475mm*45mm			248mm*156mm*36mm	

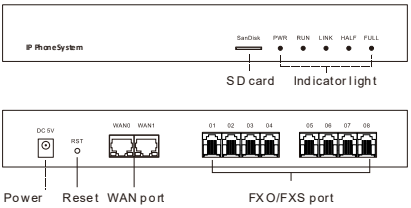
2 INSTALLATION GUIDE

2.1 DEVICE INTERFACE

PX256/512/1024 Connection



PX400/800 connection



Note: WAN0 is connected to WAN1 because there is only one network port. You can use either port to connect to the Internet

The FXS and FXO ports of the PXU model are RJ45 ports, with 4 ports out of 1. Yellow ports

The PXZ model only has FXO ports, which are blue RJ11 ports, while the FXS ports planned for later production are grey RJ11 ports

Reset key: After the machine is reset, add the default IP address (192.168.123.100) temporarily and restore the account and password; After the machine is restarted, the IP address is restored to the previous IP address

2.2 DEVICE INDICATOR

PWR indicator: Always on means the power supply is normal

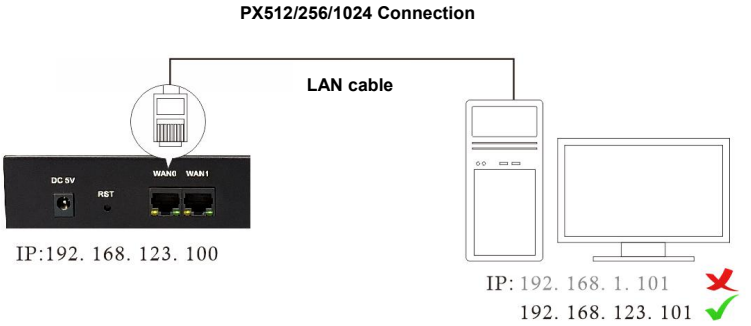
RUN indicator: Flashing quickly(3 to 4 times every second) means the device is being started. Flashing slowly (1 time every second) means the device is running properly

LINK indicator: Always on means detecting the hard disk or SD card is properly connected

HALF indicator: Always on means the hard disk /SD card usage exceeds the reserved threshold

FULL indicator: Always on means the hard disk /SD card usage exceeds the alarm threshold

2.3 NETWORK CONNECTION DIAGRAM



Caution: The PC and 192.168.123.100 must be on the same network segment

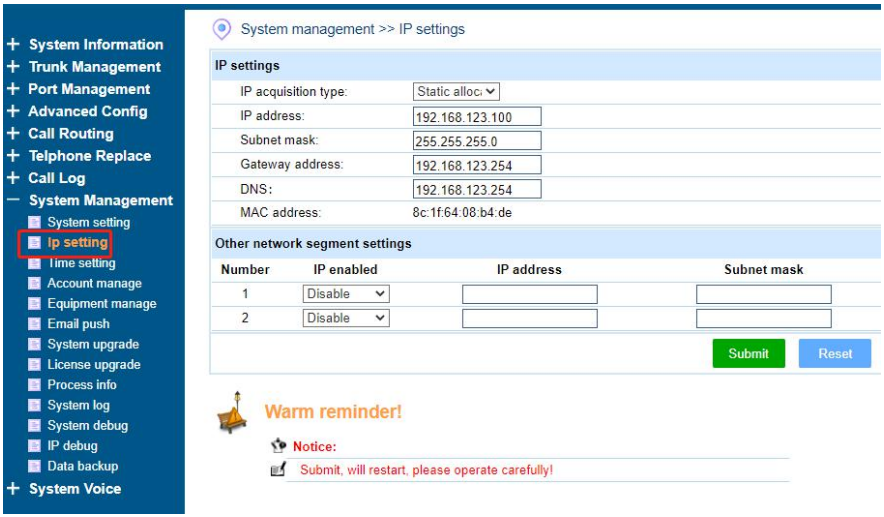
2.4 LOGIN WEB

Open a browser on the PC (recommended to use Google browser), enter http://192.168.123.100 in the address bar, administrator user name: admin, default password: admin



2.5 MODIFY IP ADDRESS

Change the IP address of IPPBX to the actual network environment (for example, set the IP address to 192.168.123.100, click submit, the system will automatically restart, after the restart is complete, enter 192.168.123.100 in the browser to log in), and then connect it and the computer to the LAN environment for use.



3 BASIC OPERATION

3.1 MAKE A CALL

Method 1: Wait 4 seconds after dialing the called number (waiting for the system dialing timeout)
Method 2: Add # to the called number when dialing

3.2 DIAL OPERATOR

By default, the operator is the first SIP extension and dial 0 to call the operator

3.3 CALL PORT GROUP

Pick up the phone and dial the number of the Port Group. All members in the group ring according to the preset ringing mode
The default port group is 8888. The member of the port group is the first SIP extension

3.4 INCOMING CALL

Connect the PSTN CO line to the FXO port. Users can dial the number of this CO line directly. On default setting, the first SIP extension is ringing extension.

3.5 MAKE OUTGOING CALL

By default, all CO lines are enabled. On each outgoing call dialing, system will choose CO line 1 on priority. Therefore, you need to disable the ports which are not connected to CO lines
The default outgoing code is 9, which can be modified
Dial 9+ CO line number

3.6 CALL PICKUP

Method 1: Pick up the phone, dial 9 + the ringing extension number
Method 2: Pick up the phone, dial 9# to pick up the earliest calling incoming call

3.7 CALL ON HOLD

You can dial *# to hold a current call, and dial *# again to restore the holding call. This operation can be also replaced by pressing the Flash button..

3.8 CALL WAITING

When the call waiting function is enabled, if you hear the call waiting tone (beep) during a call, it indicates you that a new call is incoming. You can switch between an incoming call and a current call using a Flash button.

3.9 CALL FORWARD

The Call forward function allows users to forward an incoming call after inquiring to the third party whether to answer the call.

Assume that user A dials user B's phone number and User B wants to transfer the call to user C. The procedure is as follows:

Step 1 User A dials user B's phone number.

Step 2 User B hears the phone ring and picks up the phone. User A talks with User B.

Step 3 User B presses the Flash button and dial user C's phone number (ending with a #).

Then there will be two cases:

(1) If user C answers the call, user B hangs up, then user A and user C enter the talking state.

(2) If user C does not answer the call or does not allow the transfer, User B press the FLASH button again to restore the call with User A

3.10 FLASHING HOOK OPERATION

User A talk with user B. User B flashing the hook and dials user C's number. User B holds the call with user A (User A hears the wait tone). In this case, B can perform the following three operations:

(1) Press 1 to switch to the call with user A

(2) Press 2 to switch to the call with C

(3) Press 3 to enter into three-party conference

3.11 FUNCTION CODE

The following is the phone function access code (function key). After picking up the phone, user can make call.

Advanced Configuration -> Function code

Advanced Config >> Function key

Function key settings

Function:	Function key value:	Default\Personality:	Open\Close:
Query WAN port IP address:	*158#	Default	Open
Query phone number:	*114#	Default	Open
Query port ID:	*115#	Default	Open
Check registration status:	*168#	Default	Open
Set IP address:	*152*	Default	Open
Set IP subnet mask:	*153*	Default	Open
Set up IP gateway:	*156*	Default	Open
Port volume increase:	*170#	Default	Open
Port volume down:	*171#	Default	Open
Break-in:	*14*	Default	Open
Break-out:	*15*	Default	Open
Monitor:	*13*	Default	Open
Direct call IP:	*47*	Default	Open
Enable call waiting:	*51#	Default	Open
Disable call waiting:	*50#	Default	Open
Enable unconditional transfer:	*72*	Default	Open
Disable unconditional transfer:	*73#	Default	Open
Enable transfer on busy:	*90*	Default	Open
Disable transfer on busy:	*91#	Default	Open
Enable non-answer transfer:	*92*	Default	Open
Disable no answer transfer:	*93#	Default	Open
Turn on do not disturb:	*78#	Default	Open
Turn off Do Not Disturb:	*79#	Default	Open
Access voice mailbox:	*200#	Default	Open
Self-ringing:	*142#	Default	Open
Play system music:	*128#	Default	Open
Play system time:	*123#	Default	Open
Call Agent:	*9#	Default	Open
Appoint Call Agent:	*9*	Default	Open

◆ *158# Query the IP address of the WAN port on the device

◆ *114# Query the phone number of an FXS port

◆ *115# Query the phone number of an FXS port group

◆ *168# Query the registration status of an FXS port

◆ *152* To set IPv4 address, dial *152*192*168*123*100# to set IPv4 address to 192.168.123.100

◆ *156* To set the IPv4 gateway, dial *156*192*168*123*254# to set the IPv4 gateway to 192.168.123.254

◆ *153* To set the subnet mask, dial *153*255*255*255*0# to set the subnet mask to 255.255.255.0

◆ *170# Set the volume of an FXS port to increase

◆ *171# Set the volume of the FXS port to decrease

◆ *14* Number # Barge in an original call and talk with the caller. Hold the original call (for FXS ports)

◆ *15* number # Force disconnect an original call and talk with the caller.Disconnect the original call (for FXS ports).

◆ *111# Restart the device

◆ *# Call hold (for FXS ports, press the Hold button on an IP phone)
◆ *47* Direct IP address call
◆ *51# Enable call waiting service
◆ *50# Turn off call waiting service
◆ *72* Enable directly call forward service(For example, *72*8000, set the call transfer number to 8000).
◆ *73# Disable directly call forward service
◆ *90* Enable call forward on busy (for example, *90*8000, set the call transfer number to 8000).
◆ *91# Disable call forward on busy
◆ *92* Enable the call forward service on no answer (For example: *92*3*8000, set the call to be transferred to 8000 if no one answers after three rings)
◆ *93# Disable call forward service on no answer
◆ *78# Enable DND
◆ *79# Disable DND service
◆ *200# Access voicemail

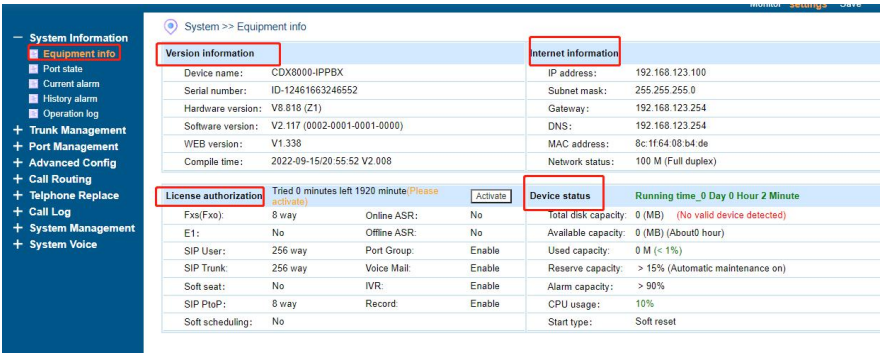
4 WEB CONFIGURATION

Reminder: The parameters take effect immediately after click “Submit” on the WEB page. After all the parameters have been modified, click "Save" at the end. If you do not click Save, the modified parameters will become invalid after the power is off.

4.1 SYSTEM INFORMATION

The system information bar contains five interface. The Equipment info and port status allows user to quickly learn about the basic Version information and Interface information. Current alarm, History alarm, and Operation log provide the administrator with logs about the running status of the device

4.1.1 Equipment Information



The device information page contains four sections: **Version information**, **Internet information**, **License authorization**, and **Device status**

Version information: Lists device names, serial numbers, and software and hardware versions. When upgrading the system, user needs to provide the device serial number (ID) code

Internet information: Default parameter of the device network port

License authorization: View the functions of the device. Including:

Device status: If the device is not activated, there is an activation button on the right of License Authorization, indicating that the device is in trial state before delivery. After the customer clicks "Device Activation", the device is in official state

The number of users or users (loop) refers to the number of FXS ports or FXO ports.

SIP user: specifies the maximum SIP extension

SIP trunk: specifies the maximum number of SIP trunks supported

SIP P to P: Indicates the maximum number of devices connected to FXO, FXS, and IPPBX

Device status: View the system running time and disk usage. The disk is used to store recording files. User can replace the disk by themselves. After replacing the disk, format the disk first

4.1.2 Current Alarm

In the current alarm bar, you can view the current alarm information about the device. You can handle the alarm according to the actual situation

4.1.3 History Alarm

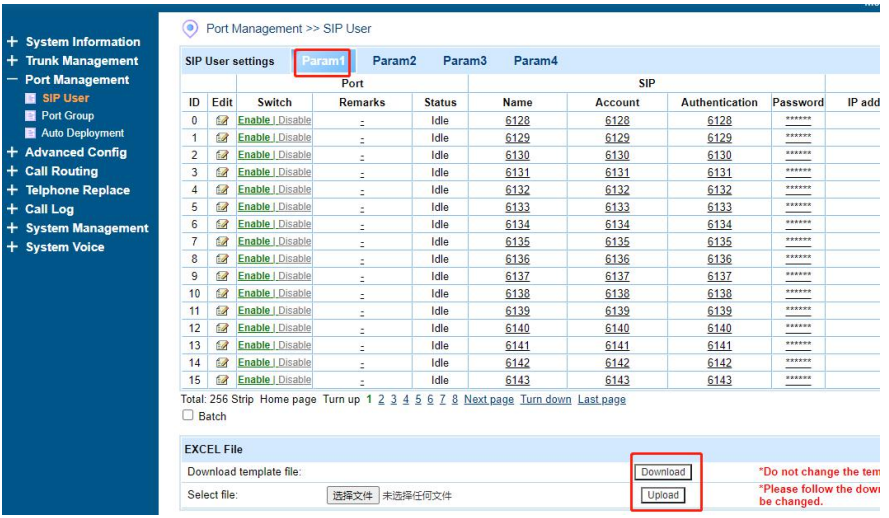
In the history alarm bar, you can view the cleared alarms. (A report was generated in a certain period), which can be used as the basis for troubleshooting

4.1.4 Operation Log

In the operation log bar, you can view all records of web operations (including the account, IP address, operation content, and operation time).

4.2 SIP USER

Port Management -> SIP User -> Param 1



By default, all SIP accounts are enabled. The default account is 6128-6XXX, and the default SIP password is 123456

Switch: Open or Stop the account

Remarks: You can remark whose user uses this account

Status: Displays the status of the current status, such as calling, ringing, and talking. You need to manually refresh the web page to display

Name: Enter the user name of the account or the same as the SIP account. You can leave this parameter blank. Generally, enter this parameter on the IP terminal

Account: IP extension number (The IP terminal must be the same as this parameter when initiating registration)

Authentication: It must be the same as the SIP account. (This parameter must be the same when the IP terminal initiates the registration.)

Password: Set the SIP account password. (This parameter must be the same when the IP terminal initiates the registration.)

IP address: records the IP address that has registered the account

SIP port: indicates that the port with an IP address registers the account

SIP state: indicates the current registration status. If an IP terminal has registered the account, it is registered; otherwise, it is unregistered. You can check the registration status of IP terminals

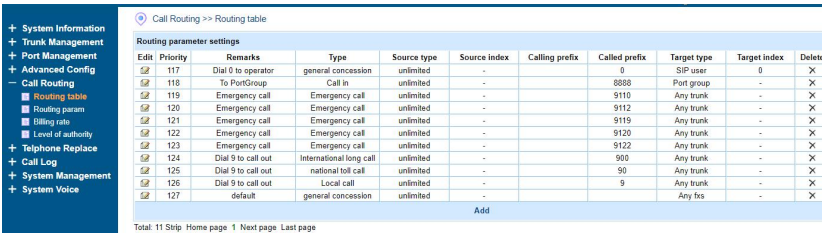
Tip: The SIP account is the extension number. General users will change the extension number according to their own needs. It is suggested to download an EXCEL module on your own computer (the "Download" button in the picture above), and then edit it on the computer through EXCEL. It is also more convenient to make a backup of the extension number. Edit and upload to IPPBX (" Upload "button in the picture above)

Note: When modifying a SIP account, ensure that it does not conflict with the items in the routing table. For example, if the item 8888 is in the routing table, the SIP account cannot be set to 8888. If the SIP account must be set to 8888, you can adjust the parameters in the routing table

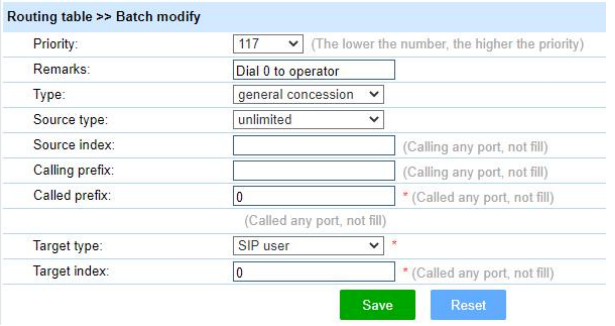
4.3 OPERATOR SETTING

By default, dial 0 to operator 6128.

Call Routing -> Routing table



Click "Edit" and the following interface pops up



If the called prefix is changed to 1, user dial 1 to the operator. The Target type can be an analog user or SIP user. The Target index corresponds to the port number of an analog user or SIP user. In the figure, port 0 of the SIP user is 6128. If you want to change the operator number to 6129, change the target index to 1. Submit and save

4.4 PORT GROUP SETTING

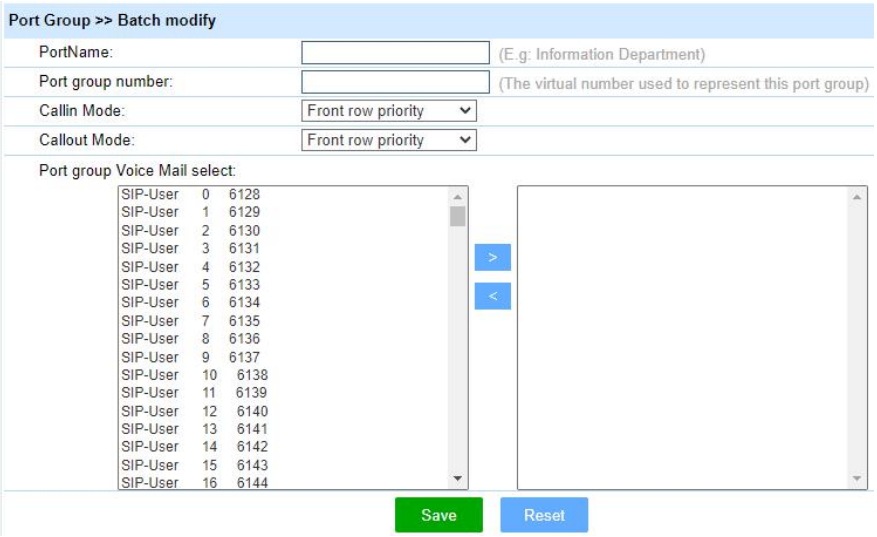
Port Management -> Port Group

The port group is also called operator group. It is mostly used as Extension group; When set in the loop trunk, it is also used for trunk line grouping.



The system has a port group 8888 by default. The member of the port group is 6128. The default ringing mode is group ringing

Click "Edit", the following interface pops up



You can add multiple ringing extensions.

The port mode is the ringing mode, which can be:

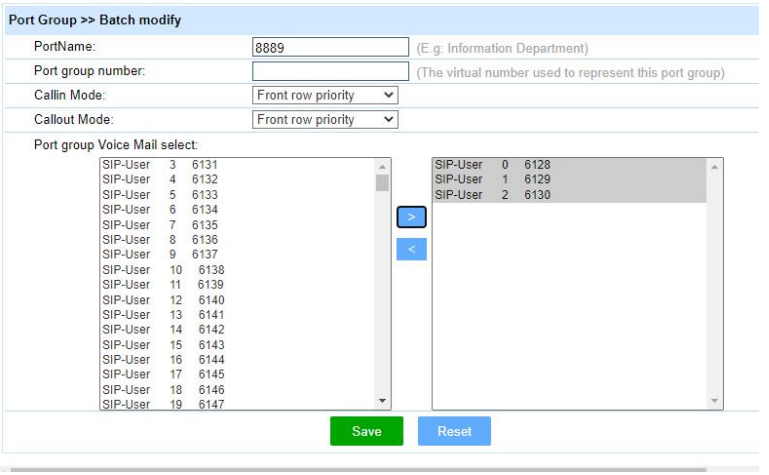
Group ringing: All members of the extension ring simultaneously

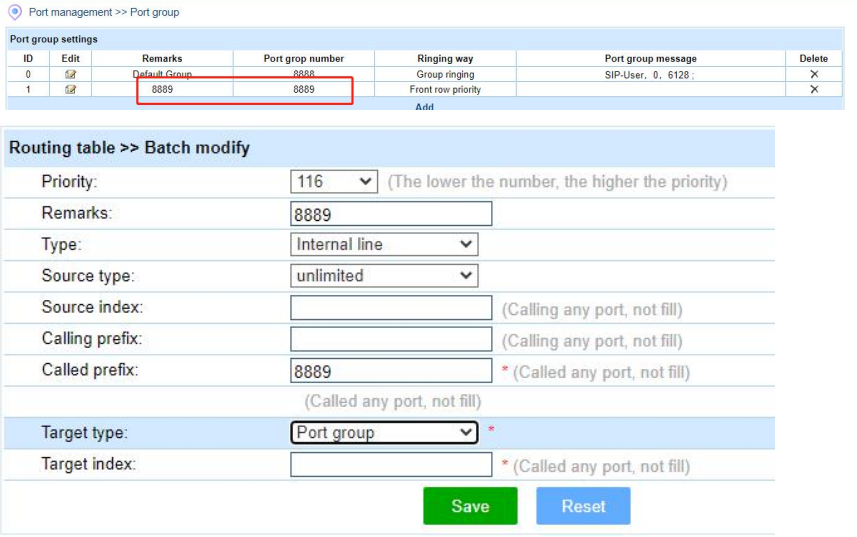
Front row priority: Ring the front of idle extensions on priority.

Cycle increment: The first incoming call rings the extension in the first row, the second incoming call rings the extension in the second row, the third incoming call rings the extension in the third row, and so on

Cycle decreasing: For the first incoming call, the extension in the last row rings; for the second incoming call, the extension in the second row rings; and so on

If you need to add one more ring group, for example, 8889, the ring extension number is 6128, 6129, 6130. then click add



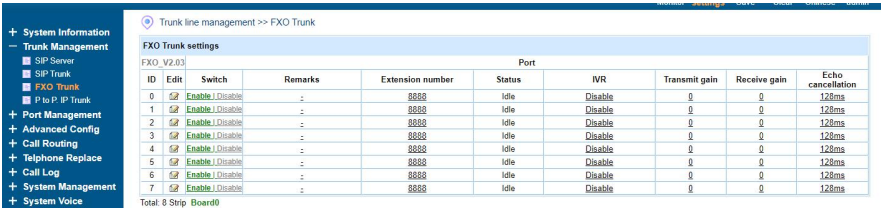


After click Submit to save, an additional record is displayed in the port group list. At the same time, go to the Call Routing -> Routing table to add a route to port group 8889

4.5 ENABLE/DISABLE TRUNK LINE

4.5.1 Analog Trunk

Trunk management -> FXO Trunk

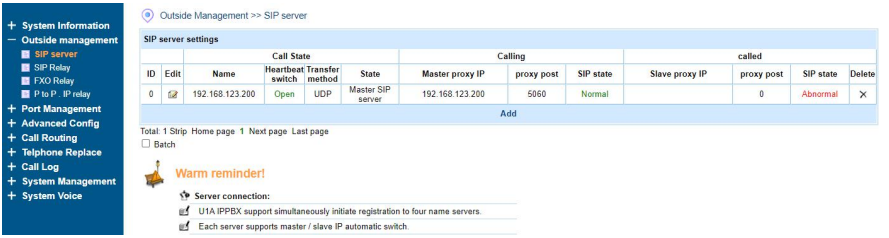


The system has a certain number of analog trunk. By default, all trunk lines are enabled. When making outgoing calls, the trunk lines in channel 0 are preferentially accessed. Please disable the trunk line channel which is unused.

4.5.2 SIP Trunk

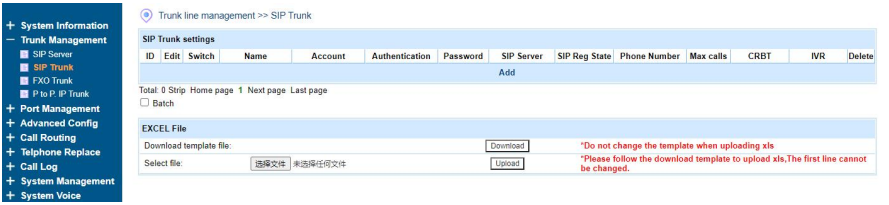
Generally, the SIP trunk should be applied from service provider/carrier, and enter the information provided by the provider on the IPPBX

(1) Trunk management -> SIP server



Enter the IP address of server and port number provided by the carrier. After the server is connected, the connection status is Normal. Four SIP servers can be connected at the same time, that is, system support four SIP trunks. Enter the server information and one or more trunk numbers provided by the carrier

(2) Trunk Management -> SIP trunk

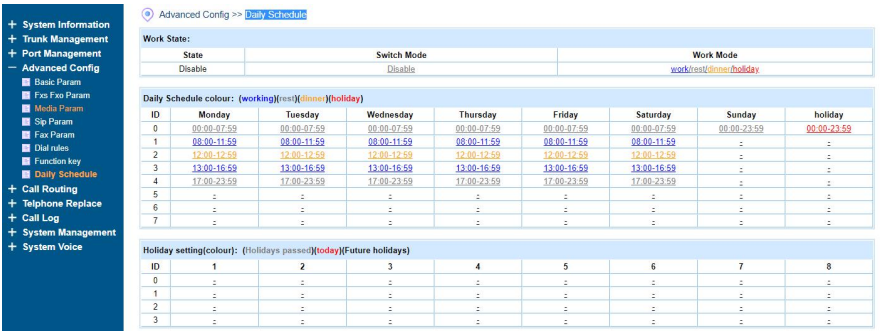


Enter the account, authentication code, and password of a trunk provided by the carrier. If multiple trunks are provided, add multiple records. After you add it, save and use it.

4.6 DAY/NIGHT MODE

There are 24 hours in a day, which can be divided into multiple time period. You can define whether the system is in working, resting, dining, or holiday status in a certain time period. For outgoing call, you can set whether directly ring the extension or enable IVR in different working status. If you don't need to use the day/night service, then no need to set here.

Advanced Config->Daily Schedule



Current Status: Display the current work and rest mode of the system

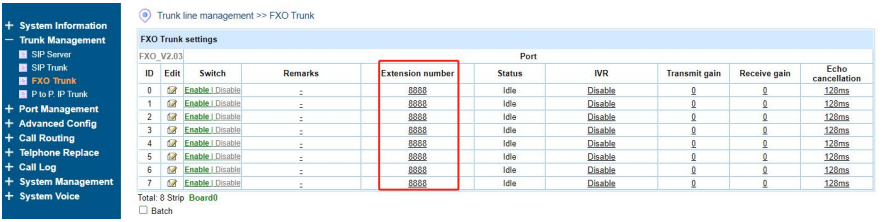
Switching mode: If the day/night service is disabled, the system will run according to the daytime working status; If the day/night service is enabled, automatic switching and manual switching mode are supported. In the automatic switching mode, the system switches the current status according to the parameter. The day/night service is disabled by default.

Work and rest mode: There are various work and rest modes for work/rest/dining/holiday. The related setting in "Work and Rest Setting" and "Holiday Setting" take effect only after the corresponding mode is enabled.

4.7 CO LINE INCOMING CALL PROCESSING

4.7.1 Ring Extension directly

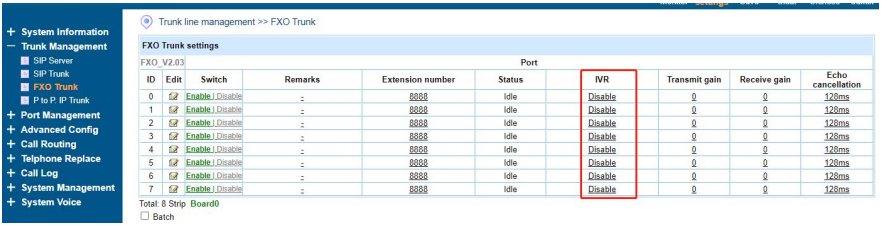
Trunk Management -> FXO Trunk -> Extension number



By default, the trunk line incoming call ringing port group is group 8888. The member in port group 8888 is only 6128. Therefore, when an CO line incoming call arrives, Ext.6128 will ring. If you want to change the ringing extension and ringing mode,please refer to the section <Port Group Setting>

4.7.2 Auto Attendant Mode(IVR)

Trunk Management -> FXO Trunk -> IVR



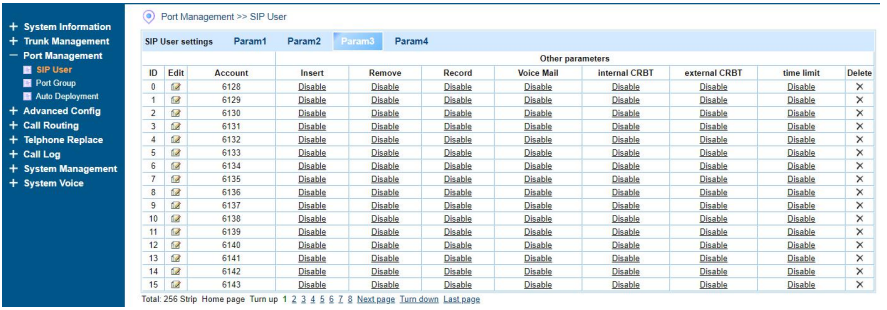
The default voice operation is to dial 0 to the operator, and dial the extension number directly to ring the extension. For details about IVR Settings, please refer to <System Voice>

If the night/day service is enabled, you can set the working mode according to the mode described in Section 4.6. For example, in the following figure, IVR is disabled in working mode. But enable the IVR in rest mode.

4.7.3 CRBT Mode

When CO line call in, the caller hear voice message, meanwhile system ring internal extension

Go to Port management -> SIP User -> Parameter 3 -> FXO CRBT, select the voice on "external CRBT" item of port 0 (default operator), and save it (how to upload the CRBT to IPPBX, please refer to section "System Voice").

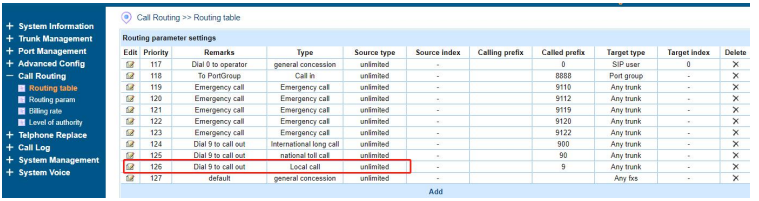


This system also supports the FXS CRBT, that is, when an extension call another extension which was set CRBT function, the waiting tone heard by the calling extension is internal CRBT message.

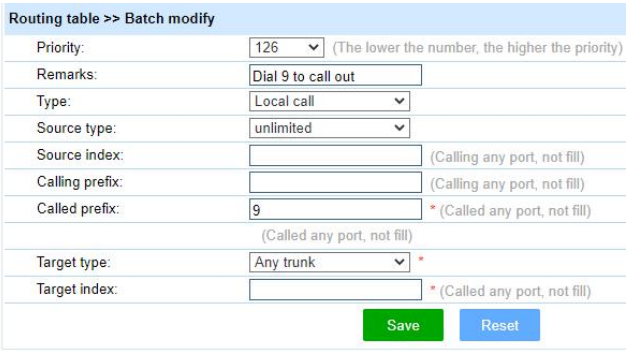
4.8 OUTGOING CALL

By default the outgoing call code is 9

Call Routing -> Routing table



Click "Edit" and the following interface pops up



If you want to change the outgoing code, change the Called prefix and change the item related to the outgoing code

Target Type:

Any trunk Indicates that a call can be made through any trunk by dialing 9

FXO trunk Indicates that dialing 9 can be made only through an external FXO port

SIP trunk Indicates that the user can dial 9 only through the SIP line provided by the carrier

E1 trunk Indicates that dialing 9 can be made only through the E1 trunk line

P to P. IP Trunk Indicates that dial-9 can be made only through point-to-point terminals.

For example, if the outgoing code is changed to 1, the prefix of the national toll number, international toll number, emergency number definition, and the setting when the call is transferred to an outside line in the routing table must be changed from 9 to 1.

After modifying the outgoing code, go to the Number Change -> Replacement table



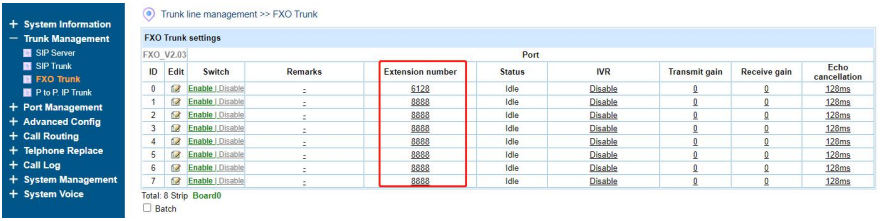
Change the called prefix 9 to 1 in the record of priority 127. After save, dial 1 for outgoing call

4.9 EXTENSION ACCESS CO LINE AUTHORITY

The system can set certain extensions to use certain CO lines

Dedicated CO line for exclusive extension: The ringing extension of the CO line is an extension. And this extension can only use this CO line to make outgoing call.

Trunk Management -> FXO Trunk -> Extension number

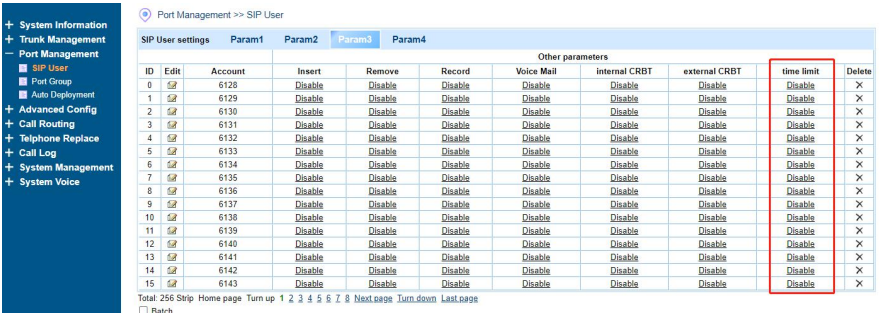


When the item "extension number" is entered with the specific extension number, that is, the CO line is bound with this extension, that is, the dedicated line for this extension only.

When the item "extension number" is entered with port group, that is setting CO line group, as shown in the following figure.

4.10 CALL DURATION CONTROL

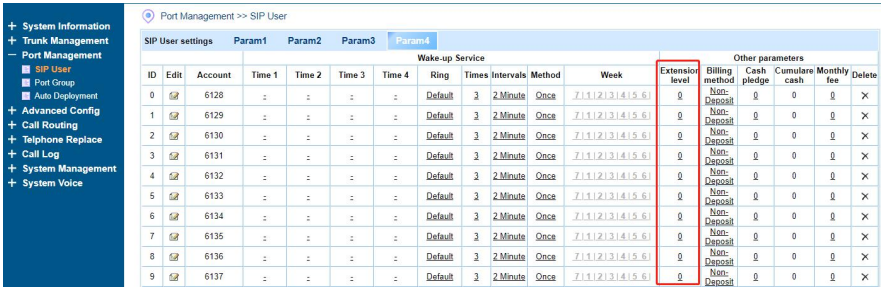
Port Management -> SIP User -> Parameter 3 -> time limit



The call duration control is only valid for outgoing call. As shown in the figure above, if the time limit of port 0 (extension 6128) is set to 1 minute, the outgoing call will be automatically disconnected after 1 minute

4.11 OUTGOING CALL LEVEL SETTING

Port Management -> SIP User -> Parameter 4 -> Extension Level



Set the extension to have local call and toll call permissions. The system can set a maximum of 16 levels, 5 default levels:

Level 0 is unrestricted

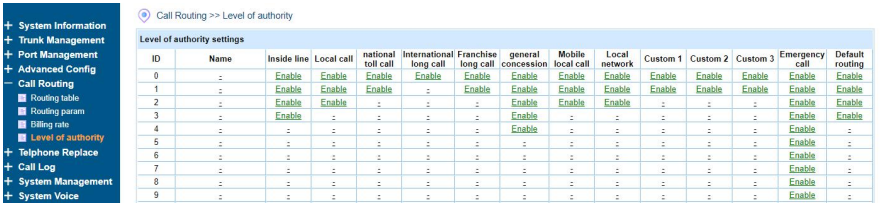
Level 1 is restricted to international call

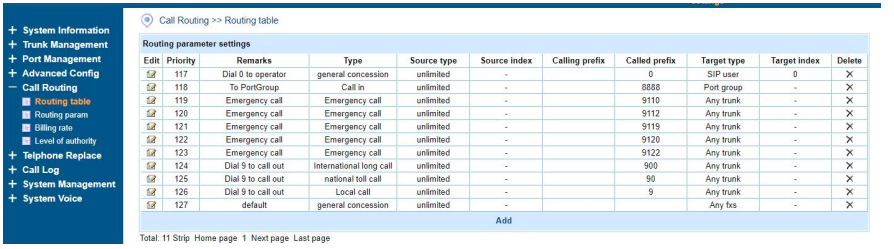
Level 2 is restricted to domestic toll calls

Level 3 is restricted to local call and can only call internal or emergency numbers (i.e.110,112,119,120,122).

Level 4 is restricted for intercom call. You can only dial 0 to the operator (6128) or dial an emergency number

If you need to configure more level or have more requirement for the level setting, please go to Call Routing-> Level of Authority





From above table we can see:

Priority 117: Dial 0 to the operator (the first SIP user) for general concession

Priority 119-123: Dial 9110,9112,9119,9120,9122 for emergency call

Priority 124: Dial 900+ CO line number to make an international long call

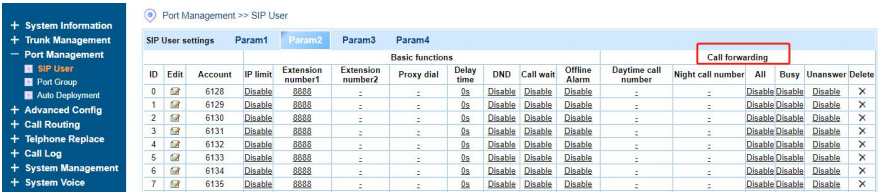
Priority 125: Dialing 90+ CO line number is a national toll call

Priority 126: Dial 9+ CO line number is a local call

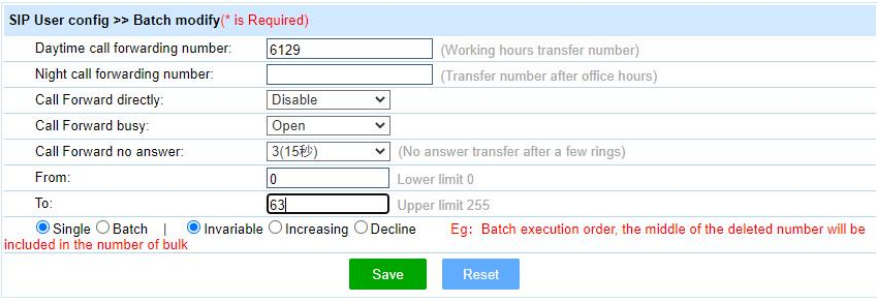
Note: The lower the priority, the higher the level. When make outgoing call, the prefix with the highest rank is matched first.

4.12 CALL FORWARD SETTING

1. You can input command on phone to set this feature directly. For details, please refer to Function Code in Basic Operations
2. On the WEB page, choose Port Management > SIP User > Parameter 2 > Call Forwarding



Click any position under Call Forward, will pop up the setting about “Call Forward”



There are three forwarding mode can be set: Call Forward directly, Call Forward busy and Call Forward no answer

For example as above figure, if port 0 is set no one answer after three rings, the call will be forwarded to extension 6129

If you need to forward to CO line, enter 9+ the CO line number (9 is the default outgoing code)

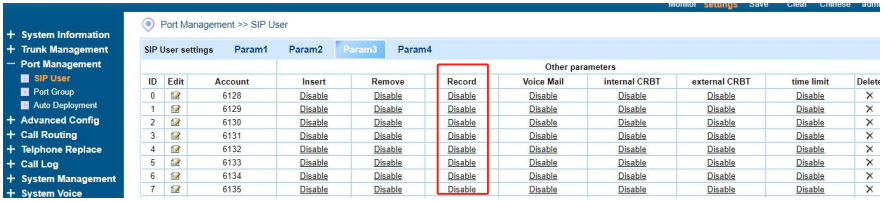
Note:

Call forward directly can be set maximum three times. There is no limit for busy or no answer call forward.

The system ringing interval is 5 seconds. So if set three ring, then the ringing time is 15s.

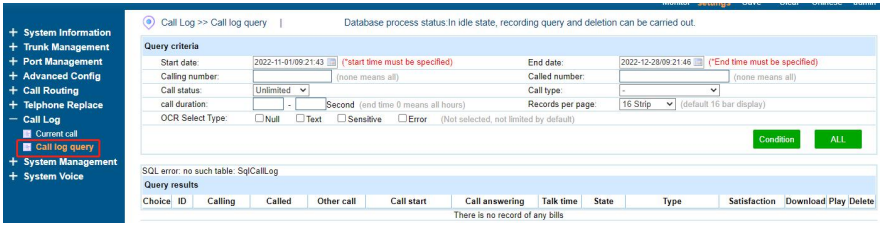
4.13 CALL RECORDING

Port Management -> SIP User -> Parameter 3 -> Record



After enable record function, the call can be recorded, and the recording file is stored in SD

Go to Call Log > Call log query to query call logs and recordings. You can play, download, and delete recording files online



4.14 VOICE MAIL

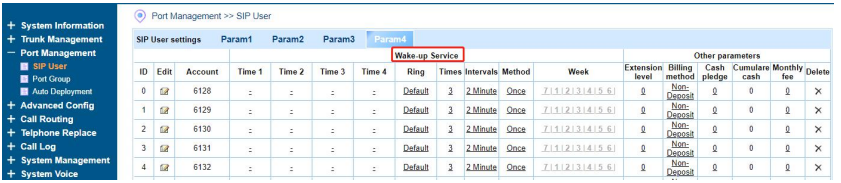
Port Management -> SIP User -> Parameter 3 -> Voice Mail



Voice mail: If the phone rings several times and no one answer it, the message is forwarded to the system. The message is stored in the SD card as prompted. If you have a leave message, enter *200# on the phone to listen to the message

4.15 ALARM CLOCK

Port Management -> SIP User -> Parameter 4 -> Wake-up Service



Wake-up service: Alarm clock function. You can set alarm clock for four time points. When the alarm time is up, the corresponding extension will ring for 60 seconds. The wake-up message can be the default voice of the system or the customized voice (you can go to the system voice -> CRBT interface to upload the customized voice).

Number and interval: If the user does not respond to the alarm within 60 seconds, the system rings again at the set interval. If the user does not respond for a period of time, the alarm ends after the set number of times. If the user responds to the call when the alarm is ringing, the alarm clock ends.

4.16 NETWORKING SETTING

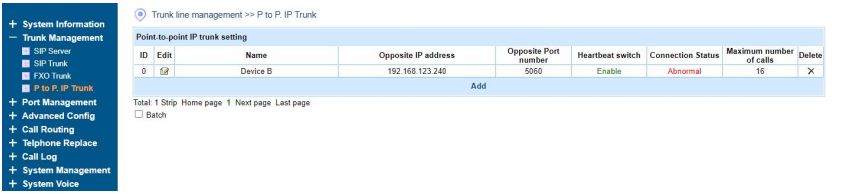
Two or more IPPBX devices are generally networked in point-to-point mode

For example, networking between device A and device B

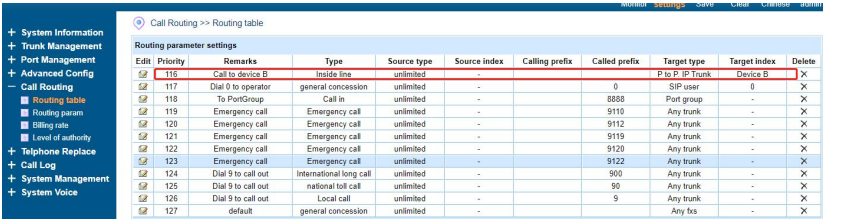
Device A: 192.168.123.240. The extension number is 6128-6XXX

Device B: 192.168.123.100; The extension number is 8128-8XXX

Step 1: Log in to device A and go to Trunk line Management > P to P. IP trunk. Add the IP address of device B. After the connection is successful, the connection status is Normal .

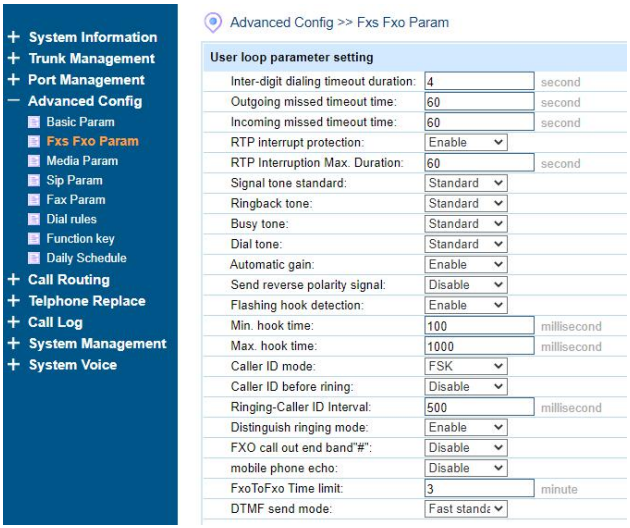


Step 2: Go to Call Routing -> Routing table. Add a route to device B. After that pick up extension of device A, dial 8xxx, can call the extension of device B.



Step 3: Log in to device B and repeat the step 1 and Step 2. Enter the information about device A in device B. A. After that pick up extension of device B, dial 6xxx, can call the extension of device A.

4.17 FXS FXO PARAMETERS



Inter-digit dialing timeout duration: indicates the inter-digit dialing timeout duration. If the value is set to 4S, it indicates that the inter-digit dialing duration exceeds 4S. The system completes dialing by default

Outgoing missed timeout time: If no answer time more than this value, the call will be defined as timeout.

Incoming missed timeout time: If no answer time more than this value, the call will be defined as timeout

RTP interrupt protection: If RTP interrupt protection is enabled and no RTP packets are found within the specified time range, the call will be disconnected.

RTP Interruption Max. Duration: Specifies the maximum duration for interrupting RTP packets. If no RTP packets are found within this period, the call will be disconnected

Dial tone, ring back tone, busy tone: standard/multi-frequency

Automatic gain: Select whether to enable automatic gain control(AGC)

Send reverse polarity signal: Select whether to enable sending reverse polarity signal. The main purpose of enabling reverse polarity signal is for call billing.

Flashing hook detection: there is an active button on the position of the phone handle, which is called the hook. In the off-hook state, the quick action of pressing this button is called "flash hook". It is a process of quickly pressing and releasing the hook, essentially cutting off the DC channel for about 100 to 400ms.

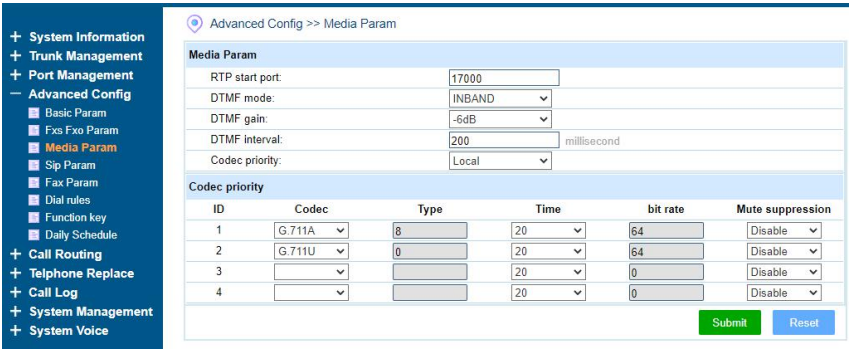
Caller ID mode: You can select DTMF and FSK as the CDR sending mode. Generally, the default configuration is FSK. The message type can be MDMF or SDMF; The message format can be display name and calling number, calling number only, or display name only.

Caller ID before Ringing: Select whether to enable CID before Ringing. If this function is enabled, the incoming call number is displayed before the call rings. Otherwise, the call number is displayed after the call rings

Ring - Caller ID interval: If Sending CID before ringing function is not enabled, you need to set the delay for sending CID after ringing. The default value is 500ms.

Distinguish ringing mode: Distinguish internal and external ringing mode during incoming call.

4.18 MEDIA PARAMETER



RTP Start port: You need to configure an RTP start port. The default RTP start port is 17000

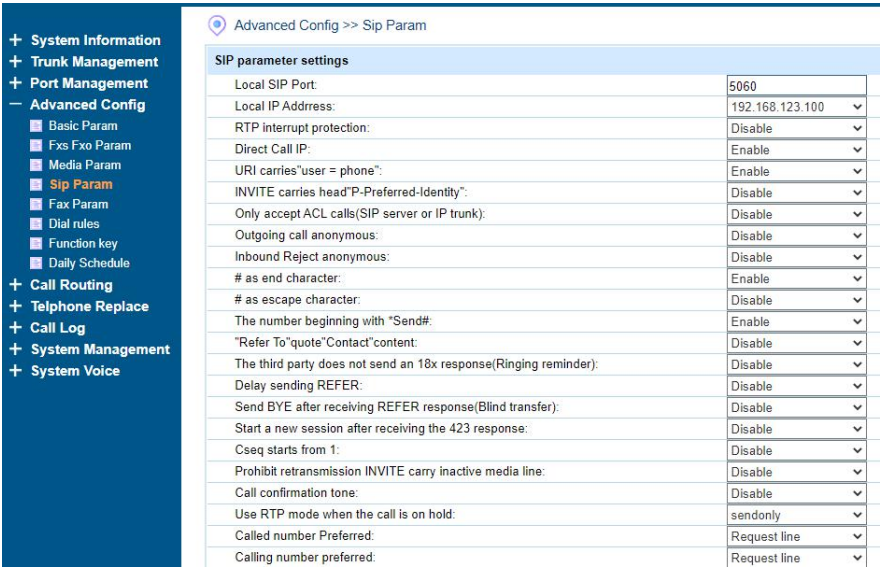
DTMF mode: SINGAL, INBAND, and RFC2833. The default is INBAND

DTMF gain: The gain of DTMF signal. The default value is 0 DB

DTMF interval: The default is 200ms

Codec priority: Select the local codec or the peer codec

4.19 SIP PARAMETER



Local SIP port: 5060 by default

Local IP address: specifies the default IP address 0 of the local IP address

Direct call IP: If this parameter is enabled, you can directly call another device using its IP address

URI carries"user = phone": If this parameter is enabled, the called number is extracted from the user name when making calls to the PSTN network

INVITE carries head"P-Preferred-Identity": The p-preferred-identity header can be used to Identify a user in anonymous calls

Only accept ACL calls(SIP server or IP trunk) : If this parameter is enabled, the device accepts only incoming calls through the SIP server or point-to-point IP trunk

Outgoing call anonymous: If this parameter is enabled, outgoing calls carry anonymous in the from header field, and the calling number is not displayed

Inbound Reject anonymous: Enable this parameter to reject incoming calls carrying Anonymous from the header field

 # as end character: If this parameter is enabled, the # indicates the end of a dial. When the device detects the # sign, it sends a call

 # as escape character: If this parameter is enabled, # is used as part of the call number

The number beginning with *Send #: If this parameter is enabled, the number starting with * ,# is also sent

"Refer To"quote"Contact"content : Refer is a SIP extension method that has been included in the RFC and its function is to ask the recipient to contact a third party by using the contact address information provided in the request. When this parameter is enabled, the Refer to field of the SIP message will be filled with the contact header

The third party does not send an 18x response(Ringing reminder): If this parameter is enabled, the third party does not send an 18x response during the inquiry transfer

Delay sending REFER: Enable this parameter. During blind transfers, the transfer operator does not REFER until it receives 200OK from a third party

Send BYE after receiving REFER response(Blind transfer): If this parameter is enabled, the third party sends BYE after receiving the REFER response

Start a new session after receiving the 423 response: Enable this parameter and automatically update the expires header after receiving a 423 response

Cseq starts from 1: If this parameter is enabled, the first value of Cseq starts at 1

Prohibit retransmission INVITE carry inactive media line: If this parameter is enabled, disallow the REINVITE to carry inactive media rows

Call confirmation tone: If this parameter is enabled, the ring tone is played back when a call does not receive a 180x response

Use RTP mode when the call is on hold: Select the RTP mode in call hold. The value can be inactive or sendonly

Called number Preferred: The p-call-party-ID header or request line is optional

Calling number preferred: The P-Asserted-Identity or from header is optional

Report SDP at any time: If this parameter is enabled, the SDP is reported at any time

18x response preferences: You can select the 18x response with SDP, the last received 18x response, or play only the local ringback tone

Hook mode: There are three hook service operation modes, namely Mode 1, Mode 2, and Mode 3

Call transfer trigger mode: The polling and transfer trigger modes include hang up and 'flash +4

Domain name resolution mode: There are three domain name query modes: Class A, SRV, and NAPTR

DNS cache: Select whether to use the DNS cache

Domain name resolution interval: Specifies the interval for domain name resolution. The value ranges from 0 to 3600 seconds. 0 indicates that the domain name is not refreshed

Early media: If this parameter is enabled, early media is supported (media interactions within early conversations are early media)

Early reply: If this parameter is enabled, early reply is supported

Provisional Response ACKnowledgemen (PRACK): If this parameter is enabled, the device supports temporary response reliable retransmission

Only 18x with SDP enabled PRACK: Enable this parameter to enable reliable retransmission only for 18x with SDP

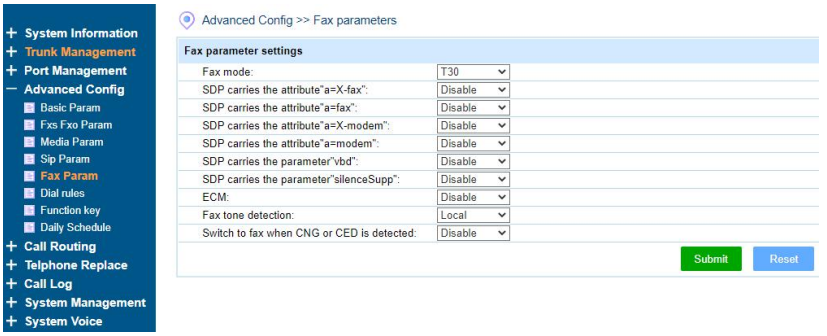
Session Timer(RFC4028): Enables this parameter. The system supports session survival detection

Session refresh interval: Specifies the session refresh interval. The default value is 1800 seconds

Min. Session refresh interval: Specifies the minimum session refresh interval

Session refresh request method: Session refresh request method. The default value is INVITE

4.20 FAX PARAMETER



Fax Mode:

1) VDB Pass-through:

Sometimes called VBD mode, the media gateway does not do any processing on the fax signal, but packages it as voice directly in the RTP packet

In this paper, it is sent by RTP stream (in order to reduce the damage to the fax signal, Pass-through fax, speech encoding

The code mode is G711A or G711U.

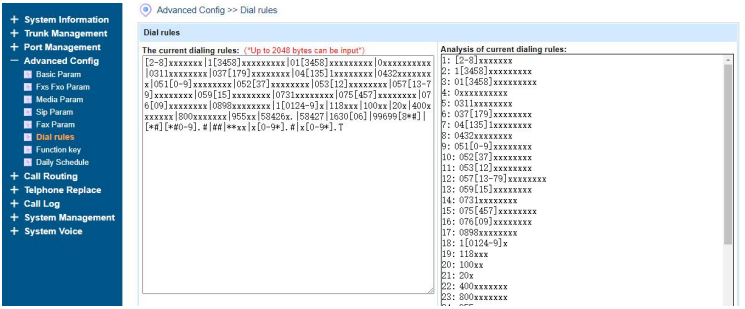
2) T30:

The media gateway recognizes the signaling in the fax signal (analog signal), converts it to the T30 mode, switches to the preset fax channel and sends RTP packets in the RTP stream mode. (To reduce the damage to the fax signal, the voice encoding mode of the fax under T30 is G711A or G711U).

3) T.38

The media gateway recognizes the signaling in the fax signal (analog signal), converts it to digital signaling and transmits it in the IP network, and restores the fax signal tone according to the signaling at the peer end. Fax data is encapsulated into T38 packets for transmission

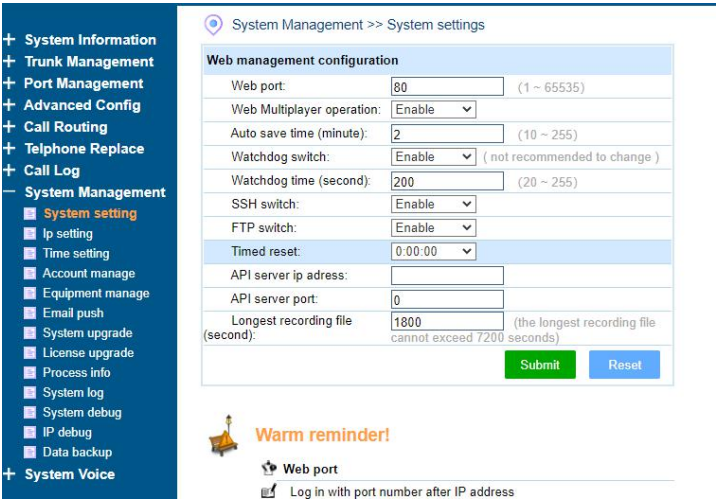
4.21 DIALING RULE



The dialing rule is used to set the dialing rules for FXS calls and supports DigitMap. The general format of a Digit Map is available.The phrasal code strictly represents an expression that contains a list of numeric characters, and the received dialing sequence is only one of those characters

If the characters match, indicates the number has been collected.

4.22 SYSTEM SETTING



Web port: The default value is 1188. You can change the value according to actual condition.

Web Multiplayer operation: Different accounts can log in to the WEB interface at the same time

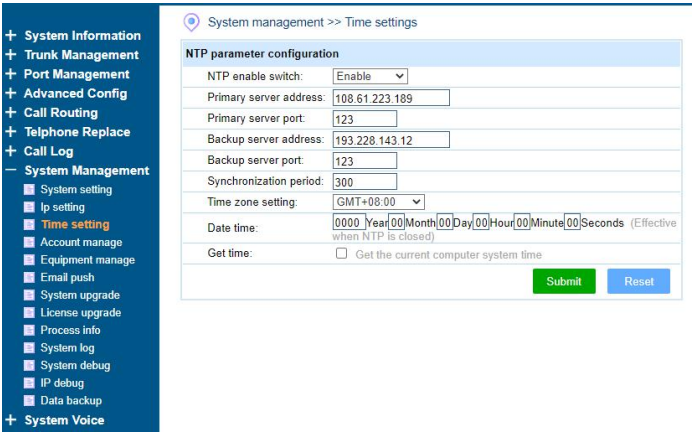
Auto save time (minute): The default value is 30 seconds. If you do not perform any operation after logging into the web for 30 minutes, the system automatically saves the information

Watchdog switch: a protection mechanism of the system. You are not advised to change it

SSH switch: The default value is On.

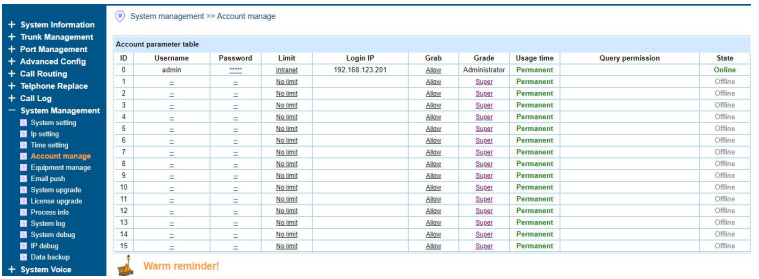
FTP switch: Enable by default

4.23 TIME SETTING



It is used to calibrate the system time of the IPPBX. The IPPBX has built-in storage battery, which can ensure the normal operation of the clock for several days even after power failure. The system automatically synchronizes time with the public time server by default. If it is found that the running time is not on time, you can use manual time synchronization (close NTP, manually set the current time or select "Obtain the current computer system time", click "submit" to update the current system time).

4.24 ACCOUNT MANAGEMENT



User name: User name for logging in to the Web page

Password: The password for logging in to the web

Limit: After this function is enabled, only IP address in the same network segment is allowed to access the system

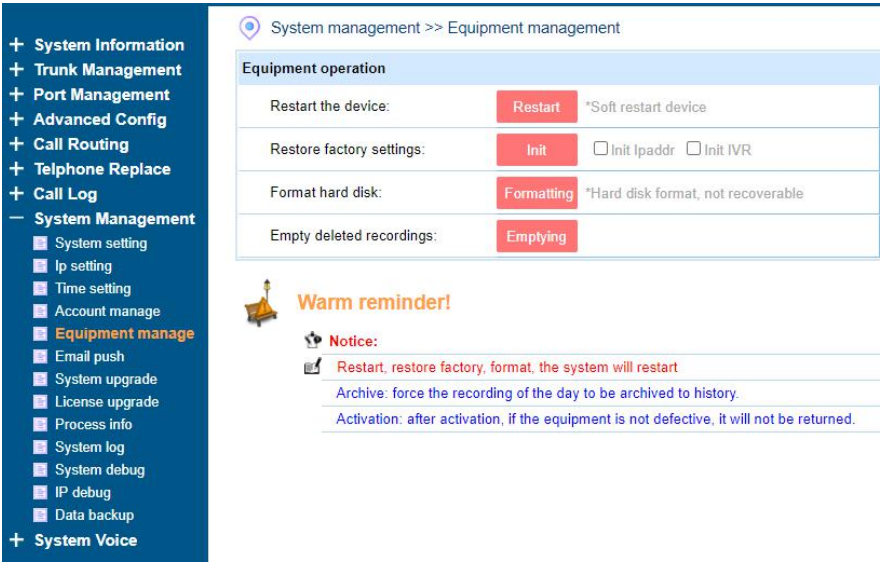
Login IP : Records the current login IP address of the user

Grade: Allowed by default. If you do not want to be login anticipating by other computer user when logging in (you can turn off this function)

Usage time: Account validity time

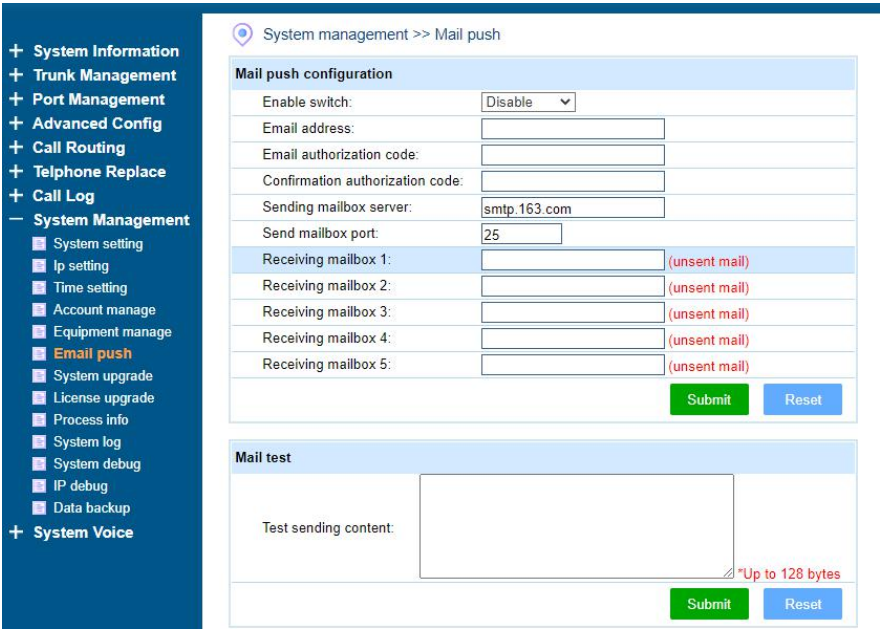
State: Displays the current status of the user

4.25 DEVICE MANAGEMENT



Restart the device: click "Restart", enter "Confirm" in the pop-up box, and click "Confirm";
Restore factory setting: click "initialization" and enter "Confirm" in the pop-up box. Click "Confirm"; the IP address parameters and IVR voice are not restored to default. If you want to restore IP address and IVR voice, select the corresponding item.
Format hard disk: Click "Format" and enter the password "19921063237" in the pop-up box. Click OK. During the formatting process, do not power off. The hard disk will be completely formatted and cannot be recovered. So please caution when performing this operation.
Empty deleted recordings: Click "Clear" and enter "Confirm" in the pop-up box. If you click "Confirm", the recording will be completely deleted.

4.26 EMAIL PUSH



Enable switch: The default value is Off. If this parameter is enabled, the device must support the Internet access function and a correct DNS server must be configured.

Email address: Email 163 is recommended.

Email authorization code: Note that you do not fill in the email password, but the email authorization code, 163 email authorization

To obtain the code, enter email 163, click the "Settings" button, and find "POP3/SMTP/IMAP".

Click, find the "Open client authorization password" on the right side, follow the steps to set (if not clear,

"How to obtain the authorization code of 163 email" can be obtained from Baidu, there are graphic introduction steps);

Send mailbox port Please do not change;

Receiving email address: You can configure five email addresses to be pushed at the same time. The latest sending status of each email address will be attached state;

Mail test: You can edit the test text in "Test sending content" and click Test Send to check the sending status

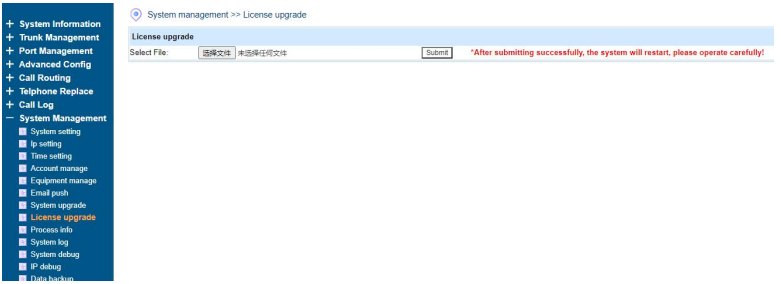
4.27 SYSTEM UPGRADE



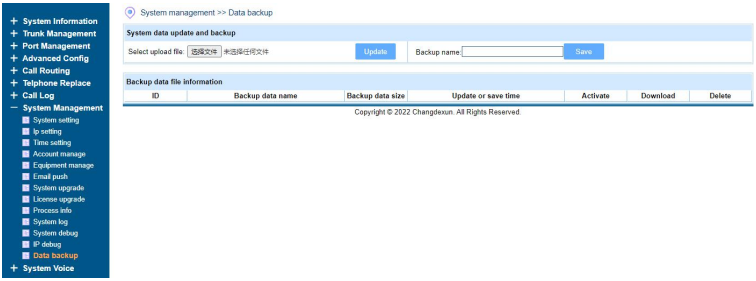
If you need to upgrade the system when the current system has some bug or you have certain function requirements, obtain the new system firmware, select File on this interface, and then upload. After the upload is successful, click Activate. The system automatically restarts

4.28 LICENSE UPGRADE

Select a file: Click Select File, find the License file to be upgraded, and click Submit.



4.29 SYSTEM DATA BACKUP

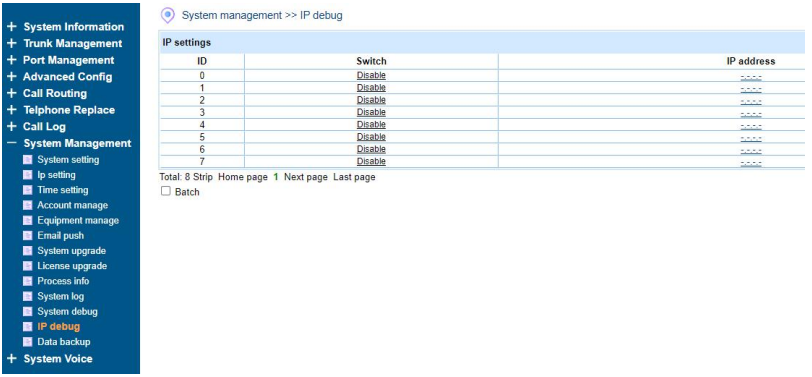


After modifying parameters, you can backup all parameters. Click "SAVE" to automatically back up a copy of the parameters. It is also displayed in the list below. You can download and save the backup file. If you need to recover a parameter. You can click Select File to upload the file. Then select the corresponding backup file point "Activation" in the list to realize the recovery parameters

4.30 TROUBLESHOOTING

Troubleshooting methods when encountering some faults during installation and use:

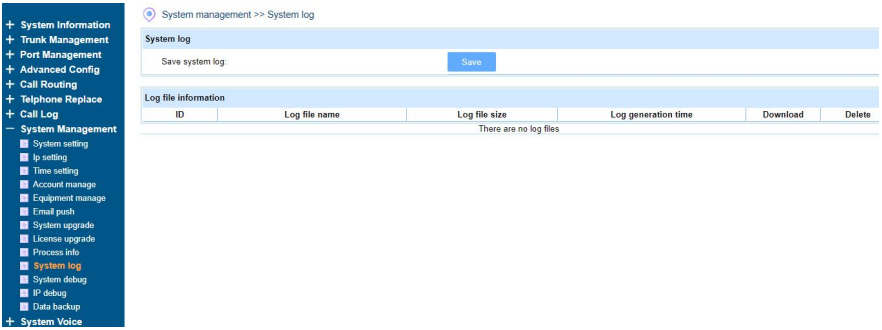
（1）System Management ->IP debug



Enter the ip address of the faulty device. For example, two ip phones cannot communicate with each other or the network between the two devices cannot work. After Enabling, click Save.

(2) Operate between two devices to the fault appear again

(3) Go to System Management -> System Logs



Click "Save" button, the system will save the current run log, save the current log according to the file name (do not change the file name) in the picture above and send it to our technical personnel for fault analysis

4.31 SYSTEM VOICE

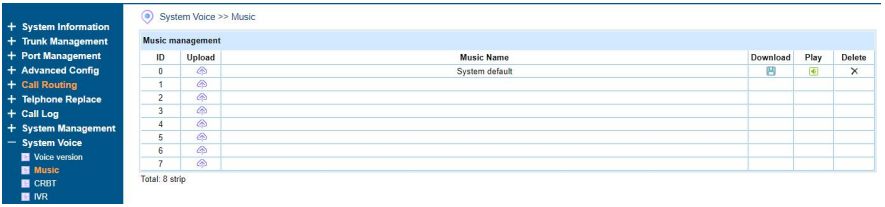
Note: All voice formats in the system must be wav format (format requirements: wave A-Law,8KHz,64kbps, mono)

You can record audio files in any format and then convert them using a format conversion tool

After a user uploads his or her voice, click "Play" on the web page. Sometimes, the voice is not uploaded by the user. You can clear the browser cache and click "Play" to listen to the voice

4.31.1 Music

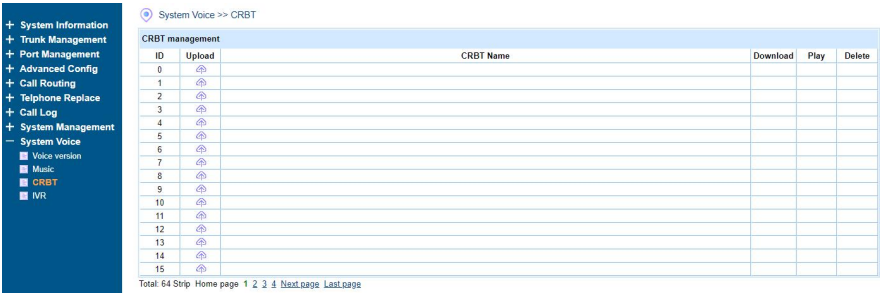
System Voice -> Music



By default, the system provides call holding music. If you want to change it, go to System Voice > Music and upload your own call holding music to position 1

4.31.2 CRBT

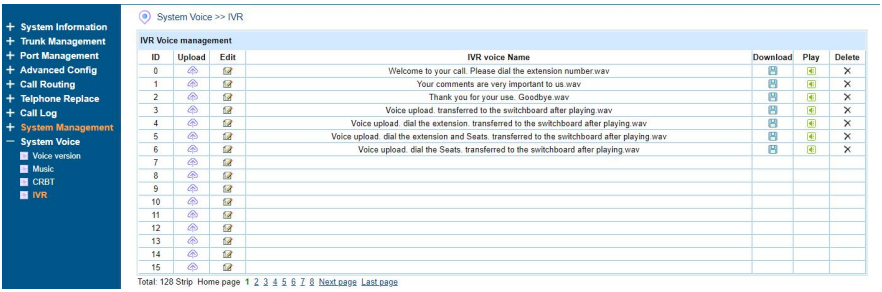
System Voice -> CRBT



CRBT is mainly used when set incoming call ringing mode as CRBT mode, the internal CRBT, and the alarm clock voice notification. Upload the ring back tone here. Then select the CRBT in the corresponding setting. The system does not has built-in CRBT

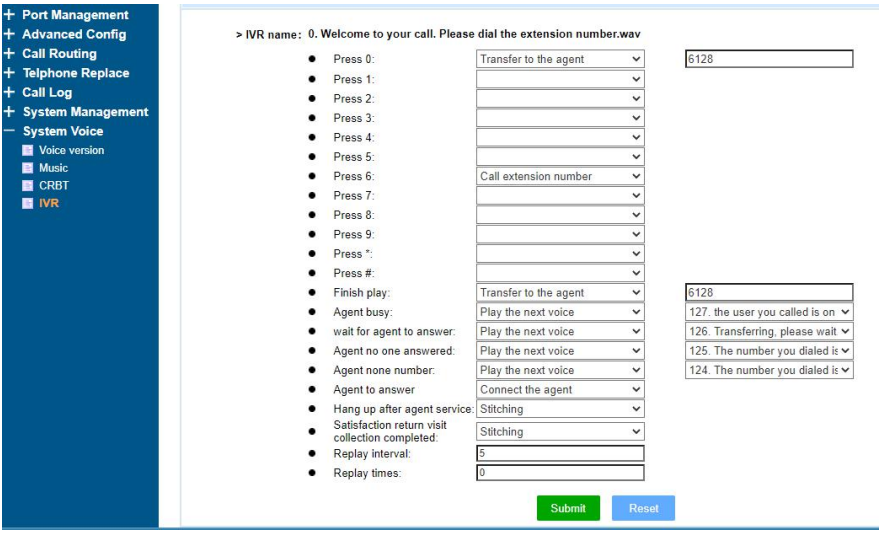
4.31.3 IVR Setting

System Voice ->IVR



The system default voice items as as above figure.. Click "Play" to listen to the voice effect. If necessary, you can replace it or add the voice you need to the blank column.

Click “Edit” on item 0, to enter the IVR voice Settings page



5 TERMINOLOGY

Acronym	Full Name
ARP	Address Resolution Protocol
CID	Caller Identity
DNS	Domain Name System
DND	Do NOT Disturb
DTMF	Dual Tone Multi Frequency
DHCP	Dynamic Host Configuration Protocol
DMZ	Demilitarized Zone
DDNS	Dynamic Domain Name Server
DSP	Digital Signal Process
NTP	Network Time Protocol
PPPOE	Point-to-point Protocol over Ethernet
PSTN	Public Switched Telephone Network
PCM	Pulse Code Modulation
QoS	Quality of Service
VLAN	Virtual Local Area Network
SIP	Session Initiation Protocol
STUN	Simple Traversal of UDP over NAT
SNMP	Simple Network Management Protocol
RTP	Real Time Protocol
UDP	User Datagram Protocol
H.248	H.248
RTCP	RTP Control Protocol