



## **FIBERME Communications LLC.**

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FCM5404 IP PBX – User Manual

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# Chapter 1 Product Introduction

Thank you for choosing the FIBERME IPPBX Series products which provide excellent VoIP solutions for Enterprise Unified Communications, Customer Service Center, Hotel Voice Communications, etc.

## 1.1. Typical Application

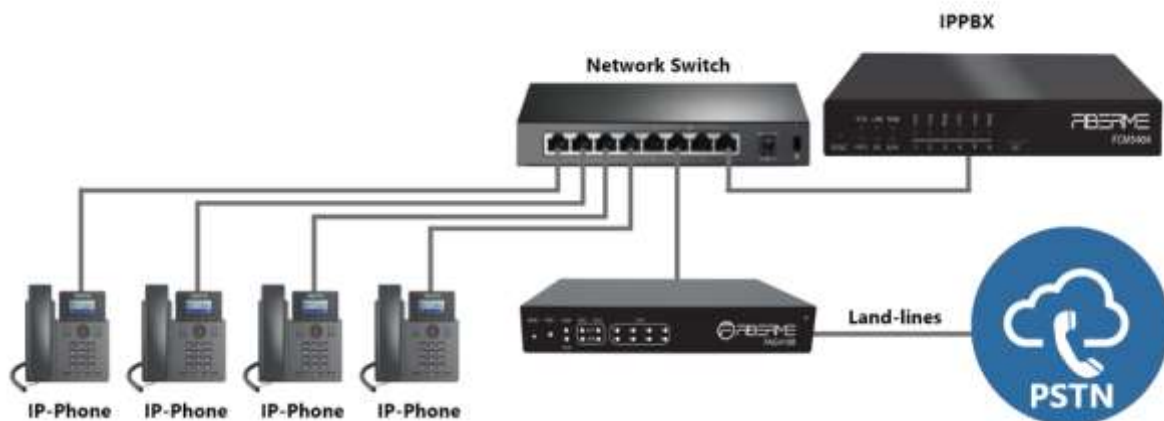


Figure 1-1IPPBX Typical Application

Main functions in typical applications:

**Enterprise Unified Communications:** Extension, trunk, route, CDR, call recording, IVR, voicemail, teleconference, call forwarding, time condition, monitoring, mobile extension, intercepting, etc.

**Customer Service/Call Center:** Extension, trunk, route, CDR, call recording, queue, monitoring, call forwarding, time condition, etc.

**Hotel Communications:** Extension, trunk, switchboard, wakeup service, paging, etc.



## 1.2. Feature List

Basic Features	Description
<b>Extension</b>	Allow users to make calls from extension to extension after registering SIP extensions to IPPBX.
<b>Trunk</b>	Allow extension users to make incoming and outgoing calls by SIP and FXO trunks with the help of inbound and outbound routes.
<b>Inbound Routes</b>	Enable forwarding calls from SIP or FXO trunks to internal extensions, IVR, conference, call center, DISA, callback systems, etc.
<b>Outbound Routes</b>	Enable making calls from extensions to external PSTN users.
<b>CDR</b>	Allow users to query and download detailed call records by condition on the webpage.
<b>Call Recording</b>	Record extensions, trunks, conferences, call centers; query, play and download the recording.
<b>Call Forwarding</b>	Extensions can be forwarded on different conditions such as 'Always', 'On Busy', 'No Answer', or 'Not Registered'. Meanwhile, time condition settings are supported.
<b>Call Waiting</b>	This feature allows an FXS extension to receive another call while on the phone. It will make the feature of transfer on busy invalid.
<b>Hotline</b>	If an extension on the FXS port doesn't dial out within the set time after it is picked up, the preset number will be called automatically.
<b>Do Not Disturb</b>	Reject all incoming calls to this extension.
<b>Mobile Number</b>	Multiple mobile numbers can be set for an extension to avoid missing any call to it.
<b>Monitor</b>	Support monitoring modes All, Listen, Whisper, Barge-in and monitoring authorities Disable, Enable All, Extensions to set for an extension.
<b>Voicemail</b>	Each extension supports an independent voicemail box as well as sending messages to a designated E-mail address.
<b>Fax</b>	Support T.38 fax extension and fax gateway modes.
<b>Extension Security</b>	Guarantee the security of extensions by password, ACL, User Agent, etc.
<b>Communication without Power</b>	Enable a connection of the station which is linked with the FXS port and the trunk which is linked with the FXO port to keep the calls between the FXS and FXO ports uninterrupted during power outage.
<b>IVR</b>	Customize multi-level IVR.
<b>Call Center Queue</b>	Customize call center queues, providing multiple station ringing strategies to satisfy a variety of applications.
<b>Conference</b>	Support teleconferencing with more than 30 parties.
<b>AutoCLIP</b>	Redirect call to original extension.
<b>CC Routes</b>	When the extension is busy, the call will be recorded. After the callback interval, the call will be dialed back.



<b>Ring Groups</b>	Set a group of extensions into a ring group. When the callers call the ring group, all available extensions will ring simultaneously or sequentially (up to different ringing strategies).				
<b>Intercept Groups</b>	Support interception of inside calls in a group and calls of specified extensions.				
<b>Call Paging</b>	Meet such requirements as paging system.				
<b>Call Parking</b>	Allow users to “park” a phone call with a parking extension number, placing it on hold to be answered on a softphone or any other phone in the office. The caller is put on hold while users switch phones.				
<b>Blacklist</b>	Numbers in the blacklist will be blocked to call in, or called, or both. It supports two modes: Exact Match and Regex Match.				
<b>DISA</b>	Enable outside users using PBX service just like the system extensions to make calls.				
<b>Callback</b>	Hang up the specified callers and let the PBX call them back.				
<b>Speed Dial</b>	Customize a short number that allows fast dialing of your frequently used numbers so that you can place a call by pressing a reduced number of keys without having to look up his/her phone number.				
<b>Emergency Number</b>	You can call the emergency number through the designated relay.				
<b>Wakeup Service</b>	You can customize the time and the extension, and then the PBX will initiate a call to the extension				
<b>Multicast</b>	You can customize the time and the multicast address, and then the PBX will play a prompt to the corresponding multicast address.				
<b>Time Condition</b>	This feature is supported for inbound routes, call forwarding, mobile extensions, etc.				
<b>PIN Code</b>	This feature is supported for outbound routes, DISA, conference, voicemail, etc.				
<b>Signaling &amp; Protocol</b>	<b>Description</b>				
<b>SIP Signaling</b>	Supported protocol: SIP V1.0/2.0, RFC3261				
<b>Voice</b>	<table border="1"> <tr> <td>CODEC</td> <td>G.711A, G.711U, G.729, G722, G723, G726-16, speex@8000h@20i, speex@16000h@20i, AMR, PCMA@30i, PCMA@40i, PCMU@30i, PCMU@40i</td> </tr> <tr> <td>DTMF Mode</td> <td>RFC2833, RFC4733, SIP INFO, INBAND</td> </tr> </table>	CODEC	G.711A, G.711U, G.729, G722, G723, G726-16, speex@8000h@20i, speex@16000h@20i, AMR, PCMA@30i, PCMA@40i, PCMU@30i, PCMU@40i	DTMF Mode	RFC2833, RFC4733, SIP INFO, INBAND
CODEC	G.711A, G.711U, G.729, G722, G723, G726-16, speex@8000h@20i, speex@16000h@20i, AMR, PCMA@30i, PCMA@40i, PCMU@30i, PCMU@40i				
DTMF Mode	RFC2833, RFC4733, SIP INFO, INBAND				
<b>Network</b>	<b>Description</b>				
<b>Network Protocol</b>	Supported protocol: TCP/UDP, TLS, SSH, HTTPS, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.				
<b>Static IP</b>	IP address modification support.				
<b>DHCP</b>	IP address dynamic allocation support.				
<b>DNS</b>	Domain Name Service support.				
<b>Security</b>	<b>Description</b>				
<b>ACL</b>	This feature is supported for extension registration and WEB access, etc.				





<b><i>Auto Defense</i></b>	Allow users to customize dynamic firewall strategies to guarantee the security of system and network.
<b><i>TLS&amp;SRTP</i></b>	Guarantee the security of signaling and voice communications.
<b>Maintain &amp; Upgrade</b>	<b>Description</b>
<b><i>WEB Configuration</i></b>	Support of configurations through the WEB user interface.
<b><i>Language</i></b>	Chinese, English.
<b><i>Software Upgrade</i></b>	Support of user interface, IPPBX service, kernel and firmware upgrades based on WEB.
<b><i>Tracking Test</i></b>	Support of Ping and Tracert tests based on WEB.
<b><i>SysLog Type</i></b>	ERROR, WARNING, NOTICE, INFO, DEBUG, CONSOLE



# Chapter 2 Quick Guide

This chapter is intended to help you grasp the basic operations of the FCM IPPBX product in the shortest time.

## Step 1: Confirm that your packing box contains all the following things.

- FCM5404 \*1
- FCM5404: External 12V Power Adapter \*1
- Warranty Card \*1
- Installation Manual \*1

## Step 2: Connect the network cable.

Connect the LAN port of FCM5404 with the network cable of the PC, or connect it to the router or PBX. Configure the IP address of the PC to 192.168.0.200 and then you can go <https://192.168.0.101> to visit the webpage of FCM5404.

Go to the page [Network Settings](#) to configure the actual IP address, subnet mask, gateway, etc. Then use the modified IP to visit the webpage of FCM5404.

## Step 3: Add and configure SIP extensions.

Go to the page [Extensions](#) to add SIP extensions. Modify extension settings and enable necessary functions according to your requirements. After that, you can perform a dial from extension to extension.

## Step 4: Add and configure SIP trunks.

Go to the page [Trunks](#) to add SIP trunks and modify trunk settings according to your requirements.

## Step 5: Add call features.

Go to the page [Call Features](#) to add necessary call features, such as IVR menus, conference rooms, call center queues, ringing groups, etc.

## Step 6: Add inbound routes.

Go to the page [Inbound Routes](#) to add inbound routes and set route destinations, such as extensions, IVR menus, conference rooms, call center queues, ringing groups, etc.

## Step 7: Add outbound routes.

Go to the page [Outbound Routes](#) to add outbound routes and set member extensions for each route.

## Special Instructions:

- The chassis of the FCM5404 IPPBX product must be grounded for safety reasons, according to standard industry requirements. A simple way is earthing with the third pin on the plug or the grounding studs on the machine. No or improper grounding may cause instability in operation as well as decrease in lightning resistance.
- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes are never jammed.
- During runtime, if the SYS indicator doesn't flash regularly and you cannot figure out and solve the problem by yourself, please contact our technicians for help. Otherwise, it may lead to a drop in performance or unexpected errors.



# Chapter 3 WEB Configuration

## 3.1. System Login

Make sure the LANs of PC and IPPBX are in the same network segment. Enter the default IP address of IPPBX <https://192.168.0.101> to log in the web interface.

The original username and password are both admin. After login, you can add users and set users' access authority, as well as modify the username and password.

Note: We suggest you use those browsers Chrome 67, Firefox60, IE11 or above versions to ensure the normal access of the management interface.

WANIP: 192.168.1.101;

LANIP: 192.168.0.101.

## 3.2. System Status

It includes eight parts: Dashboard, System Info, PBX Monitor, Active Call Queue, Active Calls, Active Conferences, Multicast Monitor and View Voicemail.

### 3.2.1. Dashboard

The FCM5404 monitors the status for Trunks, Interfaces, Storage Device Life. It presents administrators the real-time status in different sections

### 3.2.2. System Info

#### 3.2.2.1. General Info

Item	Description
<b>System Time</b>	Current system time of IPPBX
<b>Up Time</b>	Running time of IPPBX since startup
<b>Product</b>	FCM5404
<b>Serial Number</b>	Unique identifier of the device
<b>Max Sessions</b>	The default value for FCM5404 is 30. It can be authorized.
<b>Max Extensions</b>	The default value for FCM5404 is 150. It can be authorized.
<b>uboot</b>	Version information of the current uboot
<b>kernel</b>	Version information of the kernel
<b>version</b>	Version information of the current software



### 3.2.2.2. Network

#### 3.2.2.2.1. LAN

Item	Description
<b>TYPE</b>	Static IP
<b>MAC</b>	MAC address of LAN
<b>IP Address</b>	IP address of LAN
<b>Gateway</b>	Gateway address which displays only when LAN is the default network interface
<b>Subnet Mask</b>	Information about subnet mask
<b>Preferred DNS Server</b>	Information about preferred DNS server
<b>Alternate DNS Server</b>	Information about alternate DNS server
<b>Alternate DNS Server</b>	When the network cable is well connected and the network goes normal, here displays connection. If the network cable is not connected or the network is unreachable, here displays disconnection.

#### 3.2.2.2.2. WAN

Item	Description
<b>TYPE</b>	Static IP, DHCP or PPPoE
<b>MAC</b>	MAC address of WAN
<b>IP Address</b>	IP address of WAN
<b>Gateway</b>	Gateway address which displays only when WAN is the default network interface
<b>Subnet Mask</b>	Information about subnet mask
<b>Preferred DNS Server</b>	Information about preferred DNS server
<b>Alternate DNS Server</b>	Information about alternate DNS server
<b>Network State</b>	When the network cable is well connected and the network goes normal, here displays connection. If the network cable is not connected or the network is unreachable, here displays disconnection.

#### 3.2.2.3 Performance

Item	Description
<b>CPU</b>	Real-time display of current CPU utilization
<b>MEMORY</b>	Real-time display of current memory utilization
<b>LAN</b>	Real-time display of current rate of LAN
<b>WAN</b>	Real-time display of current rate of WAN

#### 3.2.2.4. Storage Usage

Item	Description
<b>Flash</b>	Display of total and used storage of the built-in flash card as well as the utilization
<b>TF</b>	Display of total and used storage of the outer TF card as well as the utilization
<b>USB</b>	Display of total and used storage of the outer USB card as well as the utilization
<b>NetDisk</b>	Display of total and used storage of the network disk space as well as the utilization



### 3.2.3. PBX Monitor

#### 3.2.3.1. Extension

Item	Description
<b>Status</b>	For a SIP trunk, display of status: unregistered/registered/ringback/ringing/talking; for an FXO trunk, display of status: idle/ringback/ringing/talking.
<b>Extension</b>	Extension number
<b>Name</b>	Name of the extension user
<b>Type</b>	Extension type, FXS or SIP
<b>IP and Port</b>	For a SIP trunk, display of IP address and port number; for an FXS trunk, display of physical port number.

#### 3.2.3.2. Trunk

Item	Description
<b>Trunk Name</b>	User-defined name of the trunk
<b>Type</b>	Trunk type, FXO or SIP
<b>Trunk Status</b>	For an FXO trunk, display of status: unusable/idle/in use; for a SIP peer trunk, display of status: unmonitored/unusable/usable; for a SIP Register trunk, display of status: fail to register/registered.
<b>Domainname/IP/Port</b>	For a SIP extension, display of domain name/IP address of the registered IP/Soft phone; for an FXO extension, display of physical port number.

### 3.2.4. Active Call Queue

Item	Description
<b>Active Call Center</b>	Statistics for queue connection rate, total number of calls, number of connected calls, number of waiting calls, number of abandoned calls, average waiting time, average talking time.
<b>Agents</b>	Click to view the statistics information about the agents in the queue (Extension, Total Calls, Answered, Missed, Caller Hangup While Agent Ringing, Login Time, Talk Time, Agent Type, Agent Status)
<b>Queue</b>	Click to view the call status information (Status, Caller, Called, Location, Wait Time, Talk Time), and you can hang up, monitor and transfer the current call.



### 3.2.5. Active Calls

Item	Description
<b>Eavesdrop</b>	Click to monitor the current call
<b>Kill</b>	Click to end the current call compulsively

### 3.2.6. Active Conferences

#### 3.2.6.1. Conference List

Item	Description
<b>Conference Room Name</b>	Display the name of the conference room.
<b>Conference Room Number</b>	Show the number of the conference room.
<b>Member Count</b>	Show the number of people online in the current conference room. It is empty when there is no conference in progress.
<b>Conference Run Time</b>	Show the duration of the current conference. It is empty when there is no conference in progress.
<b>Operation</b>	Click the operation icon of the current conference to display the participating members in the current conference room, and to further ban, mute, invite, kick out or add members, etc.

#### 3.2.6.2. Conference Contacts

Item	Description
<b>Group Name</b>	The name of the conference contacts on the phone book.
<b>Type</b>	Two types are available: <i>Extension</i> and <i>Custom</i> .
<b>Extension</b>	You can select the extension to add. When the type is Extension, the extension will be displayed.
<b>CID Name</b>	The name of the conference contact. This item is displayed when the type is set to Custom.
<b>CID Number</b>	The number of the conference contact. This item is displayed when the type is set to Custom.

### 3.2.7. Multicast Monitor

Item	Description
<b>Multicast Name</b>	Display the multicast name.
<b>Multicast Number</b>	Display the multicast number.
<b>Multicast Type</b>	Display multicast types, including <i>Triggle Multicast</i> , <i>Timed Recast</i> and <i>Live Multicast</i> .
<b>Multicast Address</b>	Display the multicast address.
<b>Enabled</b>	Whether to enable the multicast feature,
<b>Status</b>	Display the state of multicast monitor.



## 3.2.8. View Voicemail

Click to view the voicemail configuration of all extensions. Then click 'Messages' following each extension to view the exact voicemail messages.

## 3.3. Extensions/Trunks

### 3.3.1. Extensions

#### 3.3.1.1. Extensions

##### 3.3.1.1.1. Basic

General	Description
<b>Type</b>	Extension type, SIP or FXS
<b>Extension</b>	Extension number consists of all digits, with the default value range of 1000~5899 which can be modified in 'PBX->Preference->Extension Preferences'.
<b>Password</b>	It is generated randomly during the creation of a SIP extension and can be modified by users.
<b>Enabled</b>	Set whether to enable the extension or not. By default it is set to true.
<b>Max Registrations</b>	Maximum amount of registrations of this SIP extension, with the default value of 3.
<b>Effective Caller ID Number</b>	The callerID number for this extension to call outbound, i.e. the UserName field.
UserInfo	Description
<b>Name</b>	The callerID number for this extension to call outbound, i.e. the DisplayName field.
<b>User Password</b>	The password for this extension user to log into the system. Username is Name, while the default password is 'Pass' plus the extension number.
<b>Voicemail Mail To</b>	The email address to send voicemail to
<b>Mobile Number</b>	Fill in the mobile phone number of this extension user.
<b>Prompt Language</b>	The language of voice prompts. Three options available: System Default, Chinese and English. System Default means to use the same language as set in Voice Prompts.

##### 3.3.1.1.2. Features

Voicemail	Description
<b>Voicemail Enabled</b>	Once this feature is enabled, the call to this extension will enter the voicemail if failed. By default, the setting is True.
<b>Voicemail Password</b>	The password to enter the extension voicemail which is a randomly generated value by default and can be modified by users.
<b>Voicemail Keep Local</b>	Set whether to save the voicemail at IPPBX after it is sent with a specified email. By default, the setting is True.
<b>Voicemail File</b>	Set the way to send the voicemail. Audio File Attachment: Send the voice message via email attachment; Download Link: Send the voice message via link. The latter is the default setting.
Monitor	Description
<b>Allow being</b>	Set if this extension can be monitored or not.



<b>monitored</b>	<p>*Disable: Not allow to be monitored, as default.</p> <p>*Enable All: Allow all extensions to monitor.</p> <p>*Extensions: Select extensions to monitor.</p>
<b>Monitor Mode</b>	<p>Set the mode in which this extension monitors other ones. The default setting is None</p> <p>None: You will not be allowed to monitor calls;</p> <p>All: All the following 3 modes will be available for use;</p> <p>Listen: You can only listen into the call, but cannot talk (default feature code: *54)</p> <p>Whisper: You can talk to the extension you are monitoring without being heard by the other parties (default feature code: *55)</p> <p>Barge-in: You can talk to both parties (default feature code: *56)</p>
<b>Call Forwarding</b>	<b>Description</b>
<b>Always</b>	Always redirect calls to the designated destination within the period set by the following time condition select box. The default setting is Disabled.
<b>On Busy</b>	Redirect calls to the designated destination if the extension is busy within the period set by the following time condition select box. The default setting is Disabled.
<b>No Answer</b>	Redirect calls according to call source to the designated destination if not answered within the period set by the following time condition select box. The default setting is Disabled.
<b>Not Registered</b>	Redirect calls to the designated destination if the extension is not registered within the period set by the following time condition select box. The default setting is Disabled.
<b>Follow Me</b>	<b>Description</b>
<b>Follow Me</b>	Bind a target number (internal extension or external number) to this extension. When there is an incoming call, both original and bind numbers will ring at the same time so that the agent could pick up the call in different locations. The external number will go out through SIP trunks.
<b>Do Not Disturb</b>	<b>Description</b>
<b>Do Not Disturb</b>	When DND is enabled for an extension, it will reject all incoming calls. The default setting is Disabled.
<b>Mobility Extension</b>	<b>Description</b>
<b>Enable Mobility Extension</b>	If you enable this setting, when the user's mobile number dials into the system, the phone will have the same user permission as the desktop extension. So the mobile number will be able to connect with the other extension, dial out with the trunk and play voicemail.
<b>Prefix</b>	It is the same with the user's mobile number. A prefix matching the outbound route also needs to be filled in.
<b>Ring Simultaneously</b>	When the extension has an incoming call, it rings on the mobile number simultaneously and the 'Follow Me' function will be invalid.

### 3.3.1.1.3. Advanced

<b>RTP Settings</b>	<b>Description</b>
<b>Enable SRTP</b>	When this feature is enabled, the RTP stream is encrypted, sharing the same certification with TLS. The default setting is False.





<b>SIP Bypass Media</b>	Set whether to send the media stream point to point or in transparent proxy mode. Proxy Media: The media stream will pass IPPBX (default); Bypass Media: The media stream will be transported point to point. Do not enable recording when set to 'Bypass Media' to avoid problems.
<b>RTP Codec String</b>	Set RTP Codecs. So far G711A, G711U, G729, G722, PCMU, PCMA, GSM, G723, G726-16, speex@8000h@20i, speex@16000h@20i, AMR, PCMA@30i, PCMA@40i, PCMU@30i, PCMU@40i, VP8, VP9, H264, H263, H263-1998, H263-2000are supported.
<b>Register Settings</b>	<b>Description</b>
<b>AuthACL</b>	Once enabled, only the IP address or IP segment that matches the setting will be able to register this extension number. For example, 192.168.1.235/24 means all IP addresses in the segment of 192.168.1 are allowed to register; 192.168.1.235/32 means only the address 192.168.1.235 is allowed to register. By default, it is null.
<b>Online Detection</b>	Send the OPTIONS message to this extension to check if it is registered and reachable. The default setting is False.
<b>SIP Force Expires</b>	Calculated by second. The default value 0 means using the registration validity of SIP extensions while other values mean compulsively using the registration validity of IPPBX. Range: 0~3600.
<b>SIP Expires Max Deviation</b>	Reply to new REGISTER messages with time difference. This item should work with <b>SIP Force Expires</b> . For example, if <b>SIP Force Expires</b> is set 1800 seconds and this item is set 600 seconds, the value of Expires in the 200ok message which is returned by IPPBX upon successful registration will be a random value within the range of 1200s-2400s. By default, it is 0.
<b>UserAgent Filter</b>	It is null by default, which means not to verify the UserAgent field in the Register message. When it is not null, a SIP extension can register successfully as long as the UserAgent field in the Register message sent by the SIP extension is consistant in the prefix.
<b>SIP Force Contact</b>	Set whether to rewrite the contact port, or rewrite both the contact IP and port. This function will not take effect until the registration is refreshed. It is null by default.
<b>Call Settings</b>	<b>Description</b>
<b>Call Timeout</b>	Set the maximum ringing duration in seconds for every call of this extension. The default value is 30s.
<b>Max Call Duration</b>	Set the maximum call duration in seconds for every call of this extension, the call will be terminated once it exceeds the time. This item is only valid for calling external numbers. The default, value is 6000s.
<b>Wait DialTone Time Out</b>	The maximum time to wait for the dial tone, calculated by ms.The default value is 0.
<b>Wait Dial Timeout</b>	Wait Dial Timeout, with the default value of 10000 in milliseconds. The value range is 6000-15000ms.
<b>Digit Timeout</b>	Digit Timeout, with the default value of 2000 in milliseconds. The value range is 1000-10000ms.
<b>Outbound Restriction</b>	When this feature is set to True, this extension cannot call out except for emergency numbers. The default setting is False.



<b>Extension Trunk</b>	When this feature is enabled, the remote SIP trunk devices can use this extension and its password to register to this IPPBX and call in without any configuration. You can find this extension in the outbound trunk list and select it as a trunk to call out. The default setting is False.
<b>Outbound CallerIdName</b>	CallerID name of this extension trunk displayed in an outbound call once the Extension Trunk feature is enabled, having a higher priority than similar settings in Extensions.
<b>Outbound CallerIdNumber</b>	CallerID number of this extension trunk displayed in an outbound call once the Extension Trunk feature is enabled, having a higher priority than similar settings in Extensions.
<b>Call Permission</b>	Set the call permission of an extension, four options available: No Call: Block any calls from the extension. Internal Call: Only internal calls are allowed Local Call: Allow the calls without 0 as the start number Long-distance Call: Allow the calls with only one 0 at the beginning. International Call (default): Allow the calls with two 0 at the beginning.
<b>Hotline Settings</b>	<b>Description</b>
<b>Hotline Number</b>	After FXS hookoff, if you do not have any operations, it will call the hotline number which is 2 to 30bits in length. By default, it is null.
<b>Hotline Timeout</b>	Hotline timeout, with the default value of 3000 in milliseconds. The value range is 0~10000(ms) and this value must be less than Wait Dial Timeout.
<b>Other Settings</b>	<b>Description</b>
<b>RX Volume</b>	Set the volume in the direction from the SIP phone to the SIP port. The value range is -4~4 and the default value is 0.
<b>TX Volume</b>	Set the volume in the direction from the SIP port to the SIP phone. The value range is -4~4 and the default value is 0.
<b>FXS Settings</b>	<b>Description</b>
<b>Min Flash Detection</b>	Set the minimum amount of time, in milliseconds, that a hook flash must remain depressed in order for the system to consider it as a valid flash event. The default value is 300ms.
<b>Max Flash Detection</b>	Set the maximum amount of time, in milliseconds, that a hook flash must remain depressed in order for the system to consider it as a valid flash event. The default value is 1000ms.
<b>RX Volume</b>	Set the volume in the direction from the analog phone to the FXS port. The value range is -7~7 and the default value is 0.
<b>TX Volume</b>	Set the volume in the direction from the FXS port to the analog phone. The value range is -7~7 and the default value is 0.
<b>Echo Cancellation Level</b>	The default value is 64ms
<b>Enable Cut DTMF</b>	Set the length of the in-band DTMF voice to cut. Do not set it too large lest normal voice signals be cut. The default value is 25.
<b>Enable DTMFPassthrough</b>	Enable the DTMF passthrough during the conversation. By default it is unticked.



<b>Flash Event</b>	Press the hook flash on the analog phone during a call to direct this call to 3-way calling or call forwarding. The corresponding options are 3 Way (default) and Call Swap.
<b>DTMF Duration</b>	Set the length of the DTMF tone sent by FXS, calculated by ms. The value range is 20-800 and the default value is 100ms.
<b>DTMF Gap</b>	Set the interval for FXS to send DTMF tones, calculated by ms. The value range is 20-800 and the default value is 100ms.
<b>Tone Country</b>	Two options available: USA (default) and China.
<b>Call Waiting</b>	Enable the Call Waiting feature for this extension. By default it is unticked.

### 3.3.2. Extension Groups

Item	Description
<b>Name</b>	The name of the extension group. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Member</b>	Select one or more extensions to become members of the extension group. It is null by default and must be filled in; otherwise the configuration will fail to be saved.

### 3.3.3. Trunks

#### 3.3.3.1. Basic

Item	Description
<b>Trunk Type</b>	Trunk type, SIP or FXO.
<b>Trunk Name</b>	User-defined, consisting of letters and digits.
<b>Record</b>	Set whether to save the recording data. The default setting is False.
<b>Enabled</b>	Enable or disable the trunk. The default setting is True.
<b>Transport</b>	Three options available: UDP, TCP, TLS. TLS goes valid only if it is enabled in <b>SIP Settings</b> . The default setting is udp.
<b>Register</b>	Set whether to register the SIP trunk, which is determined by the trunk provider. The default setting is False.
<b>Profile</b>	Two options available: LAN (default).
<b>Trunk IP/Domain</b>	IP address or domain name of the SIP trunk plus port number.
<b>Username</b>	Username of the registered SIP trunk
<b>Auth Username</b>	Used for SIP authentication. In most cases, it is the same with the username.
<b>Password</b>	The registration password of the SIP trunk.
<b>Expire Seconds</b>	The default value is 800 seconds.
<b>RegFail Retry</b>	The default value is 30 seconds.
<b>Keep Inbound CallerID</b>	In case of unregistration, use the transparent extension as the caller by default; in case of registration, use the registered account as the caller by default.
<b>Enable Proxy</b>	Support of proxy mode for trunks like IMS. By default, it is unticked.



<b>Outbound CallerID Name</b>	CallerID name of this trunk displayed in an outbound call, having a higher priority than similar settings in <b>Extensions</b> . By default, it is null.
<b>Outbound CallerID Number</b>	CallerID number of this trunk displayed in an outbound call, having a higher priority than similar settings in <b>Extensions</b> . By default, it is null.

### 3.3.3.2. CODEC

Item	Description
<b>Codec Preferences</b>	Set the RTP codec for SIP trunk outbound calls. G711A, G711U, G729, G722, PCMU, PCMA, GSM, G723, G726-16, speex@8000h@20i, speex@16000h@20i, AMR, PCMA@30i, PCMA@40i, PCMU@30i, PCMU@40i, VP8, VP9, H264, H263, H263-1998, H263-2000 are supported at present. If none is selected, all Codecs in SDP will be used by default; otherwise, only the selected ones will be assigned.

### 3.3.3.3. Advance

VoIP Settings	Description
<b>Get CalledID Type</b>	Set to null to get called ID from the request field; set to auto to user to get called ID from the to field. By default, it is null.
<b>Send CID Type</b>	<ul style="list-style-type: none"> <li>* NONE (default): Put the CID information only in the From field;</li> <li>* Remote-Party-ID: Add the Remote Party-ID field to the CID information;</li> <li>* P-Asserted-Identify: Add the P-Asserted-Identify field to the CID information.</li> </ul>
<b>OPTIONS Interval</b>	The interval to send the OPTIONS message to check if this SIP trunk is available, calculated by second. The default setting null means no sending. By default it is null, which means not to send.
<b>Send Privacy ID</b>	When this item is set to True, the header field Privacy: id will be added to the INVITE message. By default, it is set to False.
<b>TX Volume</b>	Set the volume for voice from the SIP line to the IP port. The value ranges from -4 to 4 and the default value is 0.
<b>RX Volume</b>	Set the volume for voice from the IP port to the SIP line. The value ranges from -4 to 4 and the default value is 0.
<b>From User</b>	Use the value of this item to override the UserName field in the From header field while sending the INVITE message. By default, it is null.
<b>From Domain</b>	Use the value of this item to override the Domain field in the From header field while sending the INVITE message. By default, it is null.
<b>SIP Force Contact</b>	When you select 'rewrite-contact-IP', the host field representing contact is overwritten by the source IP address.
<b>PAIField Source</b>	Use the registered account/transparent extension as the caller. Use the transparent extension as the caller by default.
<b>Enable SRTP</b>	When it is ticked, the RTP stream is encrypted and the certificate is the same as TLS. By default it is unticked.
Other Settings	Description
<b>Limit Max Calls</b>	Set the maximum number of concurrent calls for this SIP. The default value is 0 which means no limit.



<b>Enable Early Session</b>	Add the message header <i>P-Early-Session: supported</i> to the SIP message.
<b>Enable Early Media</b>	Add the message header <i>P-Early-Media: supported</i> to the SIP message.
<b>User Phone</b>	You can add content to the To field of the INVITE message.
<b>Call Timeout</b>	Set the maximum response time of the trunk for a call out from it. The default value is 30s.
<b>DNIS</b>	Dial Number Identification Service is used to identify which trunk a call comes in. It allows users to define the display name of an incoming call instead of the called number so that the phone will display the DNIS name when a call comes in on the corresponding trunk. It is unticked by default.
<b>DNIS</b>	<b>Description</b>
<b>DNIS</b>	Dial Number Identification Service is used to identify which trunk a call comes in. It allows users to define the display name of an incoming call instead of the called number so that the phone will display the DNIS name when a call comes in on the corresponding trunk. It is unticked by default.
<b>DNIS Name</b>	The name of the caller ID displayed for the incoming call through this SIP trunk.
<b>DNIS Number</b>	The number of the callee ID of the incoming call through this SIP trunk according to which users determine the value of DNIS Name.
<b>FXO</b>	<b>Description</b>
<b>TX Volume</b>	Set the volume in the direction from the FXO port to the analog phone. The value range is -7~7 and the default value is 0.
<b>RX Volume</b>	Set the volume in the direction from the analog phone to the FXO port. The value range is -7~7 and the default value is 0.
<b>Hangup Detection</b>	<b>Description</b>
<b>Hangup Detection Method</b>	Two methods available: Busy Tone (default) and Polarity.
<b>Busy Count</b>	Specify how many busy tones to wait for before hangup. The default value is 4.
<b>Busy Freq</b>	Set the frequency of busy tones detected. The default value is 450Hz.
<b>Delay Detect Busy Tone</b>	Set the delay time before detecting the next busy tone, calculated by 20ms. The default value is 25.
<b>Busy Tone Detection Cycle</b>	Set the cycle to detect the busy tone, calculated by 20ms. The default value is 200.
<b>Answer Detection</b>	<b>Description</b>
<b>Answer Detection Method</b>	Set whether to use the Polarity method to detect if the remote end picks up the call and answers. None (default): Once an FXO outbound number is successfully sent, the call will be regarded as answered by the callee and the IPPBX will send 200ok message in the direction to the caller. Polarity: When an FXO outbound number is successfully sent and the polarity reversal signal is detected as 1 on the line, the call will be regarded as answered by the callee and the IPPBX will send 200ok message in the direction to the caller.
<b>DID Number</b>	Set the DID number for the incoming call through this FXO port. By default it is null.
<b>Caller ID Settings</b>	<b>Description</b>



<b>Caller ID Detection</b>	Set whether to detect the Caller ID of an incoming call. By default, it is ticked.
<b>Polarity Delay</b>	Set the minimum time interval for the answer polarity detection and the hang-up polarity detection. The default value is 600ms.
<b>On Ring Count</b>	Set the ring count. If Caller ID Detection is enabled, the value must be greater than 1. The default value is 0.
<b>DTMF Letter Clear</b>	Clear the letters A, B, C, D in the CallerID.
<b>Enable Debug Caller ID</b>	Enable the caller number debugging feature for recording and analyzing the caller.
<b>Set Record Time for Caller ID</b>	Set the recording timing, after the first ring or ahead of the first ring.
<b>Other Settings</b>	<b>Description</b>
<b>Enable Cut DTMF</b>	Set the length of the in-band DTMF voice to cut. Do not set it too large lest normal voice signals be cut. The default value is 25ms which means not to cut.
<b>Enable DTMFPassthrough</b>	Enable the DTMF passthrough during the conversation. By default it is unticked.
<b>DTMF Duration</b>	Set the length of the DTMF tone sent by FXO, calculated by ms. The value range is 20-800 and the default value is 100ms.
<b>DTMF Gap</b>	Set the interval for FXO to send DTMF tones, calculated by ms. The value range is 20-800 and the default value is 100ms.
<b>Wait Dialtone Timeout</b>	The maximum time to wait for the dial tone, calculated by ms. The default value is 0.
<b>Delay Dial Timeout</b>	The maximum time to delay for the dial, calculated by ms. The default value is 500. If you wish to customize, enter the value in the text box directly. The valid range is greater than or equal to '0'.
<b>Echo Cancellation Level</b>	The default value is 64ms.
<b>Tone Country</b>	Two options available: USA (default) and China.
<b>Fast Release upon Inbound Call Failure</b>	Set whether to enable this feature for a trunk. It is disabled by default. If enabled, inbound calls will be released quickly as they are failed.
<b>FXO Flash Time</b>	Set the flash time of the FXO port.

### 3.3.3.4. DOD

Item	Description
<b>DOD</b>	This feature allows users to set the caller ID and number of associated extensions displayed when dialing out which have the higher priority than the caller ID and number configured in basic settings.
<b>DOD Name</b>	The caller ID name of an outbound call.
<b>DOD Number</b>	The caller ID number of an outbound call.
<b>Bind Extension</b>	Determine which extensions are selected to bind the DOD number

### 3.3.3.5. Adapt Caller ID

Item	Description
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<b>Adapt Caller ID</b>	Adapt the incoming caller ID number by cutting or adding the prefix in order to facilitate the use of the callback feature for the SIP extension.
<b>Match Mode</b>	<p>Use regular expression to match.</p> <p>(1) ^123\$: "^" starts matching, "\$" ends matching, ^123\$ indicates strict matching</p> <p>(2) 123^(123 456)\$: " " means or, ^(123 456)\$ means to match 123 or 456</p> <p>(3) ^123 456\$: It means to match the string beginning with 123 or the string ending with 456</p> <p>(4) ^123[4-6]\$: "[" means to match any of the characters within, [4-6] is equivalent to [456], ^123[4-6]\$ means matching 1234, 1235, 1236</p> <p>(5) ^123\d\$: "\d" means any number from 0 to 9, ^123\d\$ is equivalent to ^123[0-9]\$</p> <p>(6) ^123\d+\$: "+" means to match one or more characters in front of it, ^123\d+\$ means matching at least 4 digit strings beginning with 123, such as: 1234, 12300, ..., 123456789, etc.</p> <p>(7) ^123\d*\$: "*" means to match 0 or more digits in front of it, ^123\d*\$ means matching at least 3 digit strings beginning with 123, such as: 123,1234,12300 ,...,123400000, etc.</p> <p>(8) ^123: It means any string beginning with 123, such as: 123, 12345, 123abc, etc.</p> <p>(9) 123\$: It means any string ending with 123, such as: 123, 666123, abc123, etc.</p> <p>(10) ^123\d{3}\$: "\d{3}" means to match 3 digits, ^123\d{3}\$ means matching any 6-digit string beginning with 123</p> <p>(11) ^(13[4-9] 147 15[0-2,7-9]): It indicates to match the string beginning with 134, 135, 136, 137, 138, 139, 147, 150, 151, 152, 157, 158, 159</p> <p>(12) .*: "." represents any character, .* means to match any string</p>
<b>Strip</b>	Remove the prefix of an incoming call number.
<b>Prepend</b>	Add the prefix content after removing the prefix.

## 3.4. Call Control

### 3.4.1. Inbound Routes

Item	Description
<b>Name</b>	User-defined name of this inbound route. It must be filled in; otherwise the configuration will fail to be saved.
<b>Enabled</b>	Set whether to enable this route. The default setting is True.
<b>DID Pattern</b>	<p>Use regular expression to match.</p> <p>(1) ^123\$: "^" starts matching, "\$" ends matching, ^123\$ indicates strict matching</p> <p>(2) 123^(123 456)\$: " " means or, ^(123 456)\$ means to match 123 or 456</p> <p>(3) ^123 456\$: It means to match the string beginning with 123 or the string ending with 456</p> <p>(4) ^123[4-6]\$: "[" means to match any of the characters within, [4-6] is equivalent to [456], ^123[4-6]\$ means matching 1234, 1235, 1236</p>





	<p>(5) <math>\wedge 123\d\\$</math>: "d" means any number from 0 to 9, <math>\wedge 123\d\\$</math> is equivalent to <math>\wedge 123[0-9]\\$</math></p> <p>(6) <math>\wedge 123\d+\\$</math>: "+" means to match one or more characters in front of it, <math>\wedge 123\d+\\$</math> means matching at least 4 digit strings beginning with 123, such as: 1234, 12300, ..., 123456789, etc.</p> <p>(7) <math>\wedge 123\d*\\$</math>: "*" means to match 0 or more digits in front of it, <math>\wedge 123\d*\\$</math> means matching at least 3 digit strings beginning with 123, such as: 123,1234,12300 ,...,123400000, etc.</p> <p>(8) <math>\wedge 123</math>: It means any string beginning with 123, such as: 123, 12345, 123abc, etc.</p> <p>(9) <math>123\\$</math>: It means any string ending with 123, such as: 123, 666123, abc123, etc.</p> <p>(10) <math>\wedge 123\d\{3}\\$</math>: "\d{3}" means to match 3 digits, <math>\wedge 123\d\{3}\\$</math> means matching any 6-digit string beginning with 123</p> <p>(11) <math>\wedge (13[4-9] 147 15[0-2,7-9])</math>: It indicates to match the string beginning with 134, 135, 136, 137, 138, 139, 147, 150, 151, 152, 157, 158, 159</p> <p>(12) <math>\cdot</math>: "." represents any character, <math>\cdot</math> means to match any string</p> <p>Set the DID number to a group of consecutive extension numbers. Please fill in the DID number range, select "Destination" as "Extension Range" and fill in the corresponding extension range. For example, DID range: 5503301-5503305, extension range: 1001-1005.</p>
<b>Caller ID Pattern</b>	Same as the item <b>DID Pattern</b> . By default, it is null.
<b>Destination</b>	Multiple options available, such as Extensions, IVR Menus, Ring Groups, Conference Rooms, Call Center, etc. By default it is null and must be filled in; Otherwise, the configuration will fail to be saved.
<b>Enable Inband DTMF Detection</b>	Enable the inband DTMF detection, and the default setting is False.
<b>Enable Fax Detection</b>	Set whether to enable the fax detection. The default setting is False. *False: Neither detect Fax tone nor send Fax. *True: Proceed to send Fax if Fax tone detected.
<b>Fax Destination</b>	In case the fax detection is enabled and the property of the SDP field in the INVITE message is detected as fax, it is necessary to set a route to the corresponding fax destination. By default, it is null.
<b>Send Ring Tone</b>	Local: When an incoming call to the trunk is transferred to the extension and the extension is ringing, the IPPBX will send the 183 messages well as the ringback tone to the calling party. Remote: When an incoming call to the trunk is transferred to the extension, the IPPBX will transmit the 180/183 message and the ringback tone from the extension transparently to the calling party. The default setting is <i>Remote</i> .
<b>Enable Time Condition</b>	The feature is disabled by default. Once enabled, it is required to set a destination corresponding to this time condition.
<b>Distinctive Ring</b>	Send the INVITE message with the Alert-Info header field to the called extension to





<b>Tone</b>	let it select different ring tone files based on the Alert-Info header field. By default, it is null.
<b>Priority</b>	Used to adjust the priority of multiple inbound routes. The default value is 100. The smaller the value, the higher the routing priority.
<b>Member Trunks</b>	Select the trunks that can use this route. It must be filled in; otherwise, the configuration will fail to be saved.
<b>Enable Mobility Extension</b>	If you enable this setting, when the user's mobile number dials into the system, the phone will have the same user permission as the desktop extension. So, the mobile number will be able to connect with the other extension, dial out with the trunk and play voicemail.
<b>Enable T.38</b>	Whether to enable the T.38 feature. The default setting is No.

### 3.4.2. Outbound Routes

Item	Description
<b>Name</b>	User-defined name of this outbound route. It must be filled in; otherwise the configuration will fail to be saved.
<b>Enabled</b>	Set whether to enable this route. The default setting is True.
<b>Dial Patterns</b>	<p>Use regular expression to match.</p> <p>(1) <math>\wedge 123\\$</math>: "<math>\wedge</math>" starts matching, "<math>\\$</math>" ends matching, <math>\wedge 123\\$</math> indicates strict matching</p> <p>(2) <math>123\wedge(123 456)\\$</math>: " " means or, <math>\wedge(123 456)\\$</math> means to match 123 or 456</p> <p>(3) <math>\wedge 123 456\\$</math>: It means to match the string beginning with 123 or the string ending with 456</p> <p>(4) <math>\wedge 123[4-6]\\$</math>: "[]" means to match any of the characters within, [4-6] is equivalent to [456], <math>\wedge 123[4-6]\\$</math> means matching 1234, 1235, 1236</p> <p>(5) <math>\wedge 123\d\\$</math>: "d" means any number from 0 to 9, <math>\wedge 123\d\\$</math> is equivalent to <math>\wedge 123[0-9]\\$</math></p> <p>(6) <math>\wedge 123\d+\\$</math>: "+" means to match one or more characters in front of it, <math>\wedge 123\d+\\$</math> means matching at least 4 digit strings beginning with 123, such as: 1234, 12300, ..., 123456789, etc.</p> <p>(7) <math>\wedge 123\d*\\$</math>: "*" means to match 0 or more digits in front of it, <math>\wedge 123\d*\\$</math> means matching at least 3 digit strings beginning with 123, such as: 123,1234,12300 ,...,123400000, etc.</p> <p>(8) <math>\wedge 123</math>: It means any string beginning with 123, such as: 123, 12345, 123abc, etc.</p> <p>(9) <math>123\\$</math>: It means any string ending with 123, such as: 123, 666123, abc123, etc.</p> <p>(10) <math>\wedge 123\d\{3\}\\$</math>: "\d{3}" means to match 3 digits, <math>\wedge 123\d\{3\}\\$</math> means matching any 6-digit string beginning with 123</p> <p>(11) <math>\wedge(13[4-9] 147 15[0-2,7-9])</math>: It indicates to match the string beginning with 134, 135, 136, 137, 138, 139, 147, 150, 151, 152, 157, 158, 159</p> <p>(12) <math>\cdot</math>: "." represents any character, <math>\cdot^*</math> means to match any string</p>
<b>Strip</b>	The number of digits to be removed from the prefix. By default, it is null.
<b>Prepend</b>	The digits to be added to the prefix. The default setting is null.



<b>Suffix</b>	The digits to be added to the suffix. The default setting is null.
<b>Delay</b>	The delay time before dial, calculated by ms. The default setting is null.
<b>Member Extensions</b>	Add member extensions for controlling the outbound call authority. Only those extensions selected have the authority to use this route. It must be filled in; otherwise the configuration will fail to be saved.
<b>Member Trunks</b>	Select the trunks that can use this route. It must be filled in; otherwise the configuration will fail to be saved.
<b>SIP Code</b>	Set the route to the next trunk or route when receiving a specific SIP cause code.
<b>Hangup Cause</b>	Set the route to the next trunk or route when receiving a specific hangup cause, and if the SIP code was set at the same time, the SIP code could not work. By default, it is null.
<b>Next Route</b>	If enabled, when the route is successfully matched and the call is not established normally, the next route will continue to be matched. By default, it is ticked.
<b>Password</b>	Set if you need a password for using this outbound route. The default setting is none. *None: The call goes out directly *Pin List: The gateway will require Password for outgoing calls, and will check the entered PIN with the selected PIN list in Call Features - Pin Numbers. The call will be preceded while the entered PIN matches any in the PIN list. *Single Pin: Manually set password .The gateway will require Password for outgoing calls, and the call will be preceded only if the entered PIN is correct.
<b>Rrmemory Hunt</b>	Round robin with memory, remember which trunk was used last time, and then use the next available trunk to call out.
<b>Description</b>	Description of the outbound route. By default, it is null.
<b>Priority</b>	Used to adjust the priority of multiple outbound routes. Smaller Number means higher Priority. The default value is 1000. The smaller the value, the higher the routing priority.
<b>Time Condition</b>	Set which time period to use this route. Untick any option by default, which means no time limits on outbound calls.



### 3.4.3. Outbound Restrictions

Item	Description
<b>Name</b>	Name of this user-defined outbound restriction. It must be filled in; otherwise the configuration will fail to be saved.
<b>Time Limit</b>	Set a time limit for calls. The default value is 5 minutes.
<b>Number of Calls Limit</b>	Set how many calls are allowed in the limited time. For example, if <b>Time Limit</b> is set to 5 minutes and this item is set to 5, it means the designated extension can only make 5 calls in 5 minutes. When this extension makes the 6 <sup>th</sup> call, it will be locked.
<b>Auto Cancel Restriction</b>	The setting of True means the designated extension can make more calls after the time limit even if it is locked; the setting of False means this extension, once it is locked, cannot make outbound calls any more until it is unlocked manually.
<b>Enabled</b>	Set whether to enable this outbound restriction rule. The default setting is True.
<b>Member Extensions</b>	Select the extensions that use this restriction rule. It must be filled in; otherwise the configuration will fail to be saved.

### 3.4.4. AutoCLIP

Item	Description
<b>AutoCLIP</b>	AutoCLIP can redirect call to original extension. The IPPBX automatically stores information about outgoing calls to the AutoCLIP routing table. When the same person calls back, the call will be routed directly to the original extension that made the former mentioned outgoing call.
<b>View AutoCLIP List</b>	A list of extension outbound calls.
<b>Delete Used Records</b>	If enabled, when an AutoCLIP record is matched, it will be automatically deleted afterwards. By default, it is unticked.
<b>Record Keep Time</b>	Set how long each record will be kept in the AutoCLIP list. The default value is 8 hours.
<b>Only Keep Missed Call Records</b>	If enabled, the system will only keep records of outbound calls that are not answered by the called party in the AutoCLIP list. By default, it is ticked.
<b>Match Outgoing Trunk</b>	If enabled, only the calls that come in through the same trunk as the last call go out from will match against the AutoCLIP list. By default, it is ticked.
<b>Record PSTN Trunk</b>	If enabled, calls that go out through PSTN will be recorded to the AutoCLIP list. By default, it is ticked.
<b>Digits Match</b>	Define how many digits from the last digit of the incoming call number will be used to match the AutoCLIP record. If the number has fewer digits than the value defined here, it will be matched in full length. The default value is 7.
<b>Enabled</b>	Set whether to enable the AutoClip routing. The default setting is False.
<b>Member Trunks</b>	Select the trunk on which outgoing calls will be recorded. It is required; otherwise the configuration will fail to be saved.



### 3.4.5. CC Routes

Item	Description
<b>CC Routes</b>	When the extension is busy, the call will be recorded. After the callback interval, the call will be dialed back.
<b>View CC List</b>	View the list of calls which are recorded upon the extension is busy and need to be dialed back.
<b>CC Interval Time</b>	The callback interval for calls in the CC record. The default value is 1 minute.
<b>Record Keep Time</b>	The time to keep a CC record. The default value is 8 hours.
<b>Enabled</b>	Set whether to enable the CC routes. By default, it is False.
<b>Member Extensions</b>	Add the extensions which have the authority to control the CC routes.

ps: When the extension calls back the caller and the caller picks up the call, it will hear the prompt 'This is a call back, please do not hang up' and then call back the extension. If the call comes from the FXO port, the caller may not hear the prompt or the prompt may be incomplete. In such case, it is recommended to turn on the Polarity Reversal function for the FXO port.

### 3.4.6. Time Condition

Item	Description
<b>Time Condition</b>	It can be set for such features as outbound routes, inbound routes, call forwarding, and follow me.
<b>Name</b>	User-defined name of a time condition. It must be filled in; otherwise the configuration will fail to be saved.
<b>Type</b>	Three options available: WorkTime (default), Holiday, Custom.
<b>Settings</b>	Define worktime and custom conditions necessary to execute the destination selection.
<b>Advance</b>	If ticked, more settings will appear for you. By default, it is unticked.
<b>WorkTime</b>	Multiple times allowed to set, including day of week, hour and minute by default. If you need to set year, month, day of month, tick the above item <i>Advance</i> .
<b>Holiday</b>	Multiple times allowed to set, including year, month and day of month by default. If you need to set day of week, tick the above item <i>Advance</i> .
<b>Custom</b>	Multiple times allowed to set, including month, day of month, week of month, day of week, hour, minute, as well as exclude holiday.



## 3.5. Call Features

### 3.5.1. IVR

#### 3.5.1.1. IVR

Basic	Description
<b>Name</b>	User-defined IVR name. It must be filled in; otherwise, the configuration will fail to be saved.
<b>IVR Number</b>	The extension number that can be routed to this IVR, with the default value range of 6500~6599 which can be modified in 'PBX->Preference->Extension Preferences'. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Greet Long</b>	It is played as the first prompt for entering the IVR menu. The default setting is Default.
<b>Greet Short</b>	It is played when the user doesn't enter any key or enters a wrong key. By default, it is null.
<b>Response Timeout</b>	The time waiting for a digit input after prompt. The default value is 10000ms.
<b>Inter-Digit Timeout</b>	The maximum time between your entering of two adjacent DTMF digits. The default value is 3000ms.
<b>Max Timeouts</b>	Maximum number of timeouts before exit. The default value is 3.
<b>Password</b>	Set the IVR password. The default value is 0000.
<b>Max Failures</b>	Maximum number of retries before exit. The default value is 3.
<b>Digit Length</b>	Maximum number of digits allowed for the caller ID. The default value is 4.
<b>Enabled</b>	Set whether to use the IVR. By default, it is True.
<b>Direct Extension</b>	Set whether the user can dial directly to extensions after hearing the IVR prompt.
<b>FXO Flash Transfer</b>	Set whether to allow the current FXO to flash transfer. By default, it is False.
<b>Direct Outbound</b>	Set whether the user can dial directly out after hearing the IVR prompt. By default, it is unticked.
Advanced	Description
<b>Invalid Sound</b>	The prompt played in case of invalid keypress. The default setting is Default.
<b>Exit Sound</b>	The prompt played upon exiting the IVR menu. The default setting is Default.
<b>Exit Action</b>	The destination selected to enter after exiting the IVR menu. By default, it is null.
<b>Caller ID Name Prefix</b>	The prefix of the caller ID name sent upon the call passing from IVR to an internal extension. By default, it is null.



<b>Ring Back</b>	The ring back tone the caller will hear upon the call passing from IVR to an internal extension. The default setting is Default.
<b>Key Press Event</b>	<b>Description</b>
<b>Keypress Event</b>	Set the destinations respectively according to 0~9, *, # and invalid keys.

### 3.5.1.2. Conference

#### 3.5.1.2.1. Basic

Item	Description
<b>Room Name</b>	User-defined name of a conference room. It must be filled in; otherwise the configuration will fail to be saved.
<b>Conference Center Number</b>	The number dialed to reach this conference room, with the default value range of 6400~6499 which can be modified in 'PBX->Preference->Extension Preferences'. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Greeting</b>	The greeting played upon joining this conference room. The default setting is Default.
<b>Schedule</b>	Set the start and end time for this conference room.
<b>Pin</b>	Set whether a password is needed for entering this conference room. The default setting is True.
<b>Record</b>	Set whether to enable the recording. By default, it is False.
<b>Max Members</b>	The maximum number of members allowed in this conference room.
<b>Announce</b>	If set to True, other members will hear prompts upon a member enters or exits this conference room; if set to False, there will be no prompt for a member's entering or exiting. The default setting is False.
<b>Enabled</b>	Sets whether to use the conference room. The default setting is True.
<b>Moderator Member</b>	Specify the moderator extension for this conference. It must be filled in; otherwise the configuration will fail to be saved.

#### 3.5.1.2.2. Advanced Settings

Participant Settings	Description
<b>Wait for Moderator</b>	If set to True, the participants could not hear each other until the moderator joins the conference. The default setting is True.
<b>Say your name</b>	If set to True, you will hear a prompt 'Please say your name' upon you enter a conference room, and other members will hear a prompt 'xxx enters the conference' upon you successfully join in the conference. The default setting is True.
<b>Mute Participant</b>	If set to True, the participants expect for the moderator are not allowed to speak in this conference room. The default setting is False.
<b>Allow Participant to Invite</b>	If set to True, all participants could press *0 to invite other users to enter this conference room, press *1 to launch an invitation with confirmation request and press *2 to kick the member they invited out of this room. The administrator could press *3 to kick out all participants in the conference. The default setting is True.

#### 3.5.1.2.3. Appointment Meeting Settings

Item	Description
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<b>Enable Appointment</b>	If set to Yes, Appointment Meeting will be enabled; if set to No, Appointment Meeting will be disabled. The default setting is No.
<b>Meeting Group Name</b>	Choose the members who will be invited to the conference meeting.
<b>Custom Date</b>	To custom the date of Appointment Meeting. The default setting is Yes.
<b>Date</b>	Choose the date of Appointment Meeting.
<b>Time</b>	Choose the time of Appointment Meeting.

### 3.5.1.3. Call Queue

#### 3.5.1.3.1. Basic

Item	Description
<b>Queue Name</b>	User-defined name of a call center queue. It must be filled in; otherwise the configuration will fail to be saved.
<b>Queue Number</b>	The number dialed to reach this call center queue, with the default value range of 6700~6799 which can be modified in 'PBX->Preference->Extension Preferences'. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Pin</b>	Set whether a password is needed for dynamic agents to enter this queue. The default setting is False.
<b>Agent Password</b>	Set the password for dynamic agents to enter this call center queue. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Ring Strategy</b>	Ring All: All available agents ring. Longest Idle Agent (default): The agent keeping idle for the longest time rings first. Round Robin: All available agents ring in rotation. Random: All available agents ring randomly. Agent with Least Talk Time: The agent whose total call time is shortest rings first. Agent with Fewest Calls: The agent with fewest calls rings first. Top Down: The agents ring from top to down in the order already configured. Sequentially by Agent Order: The agents ring in the order of their numbers.
<b>Agent Call Timeout</b>	The maximum time for each agent to ring. The default value is 15 seconds.
<b>Max No Answer</b>	The allowed number of consecutive unanswered calls. 0 means no limit and the default value is 0.
<b>Max Wait Time</b>	The maximum time a caller can wait in a queue before being pulled out, calculated by second. 0 means no time limit.
<b>Discard Abandoned</b>	Set the recovery time from disposal.
<b>Timeout Action</b>	Select the destination to enter when the call in the queue doesn't be answered in the maximum waiting time. By default, it is null.
<b>Agent Answer Announce</b>	Announcement played upon the agent answers the call. The default setting is null.
<b>Agent Retry Time</b>	The interval time between the failed and new calls of an agent. The default value is 30 seconds.
<b>Wrap Up Time</b>	The interval time between the answer of an incoming call and the allocation of a



	new one.
<b>Max Queue Length</b>	Set how many callers are allowed in the queue.
<b>Caller ID Name Prefix</b>	The prefix of a caller ID name sent when the queue allocates a call to the agent. By default it is null.
<b>Overflow Action</b>	Transfer to the destination when the number of calling parties in the queue exceeds the upper limit.
<b>Agents Initial Status</b>	Set the initial status of the static agents.
<b>Alert Info</b>	Set the content of the Alert-Info field. By default, it is null.
<b>Agents</b>	Set one or several extensions to be the fixed station of the current queue.

### 3.5.1.3.2. Caller Experience Settings

Item	Description
<b>Music on Hold</b>	Select the music on hold to play when the caller enters this queue. The default setting is Default.
<b>Join When No Agent</b>	If enabled, callers can join a queue that has no agents. By default, it is unticked.
<b>Max Wait Time with No Agent</b>	The maximum waiting time for a caller in the queue that has no agents. The default value is 90s.
<b>Join Announce</b>	Announcement played to callers upon joining the queue. The default setting is Default.
<b>Queue Busy Resume Offer</b>	Set whether to assign incoming calls to other stations if the current station is already in call. The default setting is True.
<b>Join Announce Playtime</b>	Set the playtime of Join Announce. The default value is 0 which means it is invalid,
<b>Answer Type</b>	Set the way to answer. When it is set to Answer, 180 will be returned; when it is set to Pre-answer, 183 will be returned. The default setting is Answer.
<b>Transfer Prompt</b>	When the calling party enters the queue, the system prompt tone will be heard periodically. If the agent rings, the prompt tone will be played. For example: "Call forwarding, please wait for a moment". The default setting is empty.
Caller Position Announcements	Description
<b>Announce Position</b>	Announce the current position of the caller in the queue. By default, it is ticked.
<b>Announce Hold Time</b>	Announce how long the caller shall wait in the queue. By default, it is ticked.
<b>Call Duration</b>	The average call length estimated by users based on actual situations, used to calculate the waiting time for the caller. The default value is 60s.
<b>Announce Frequency</b>	Set how often to announce the queue position and the hold time. The default value is 30s.
Periodic Announcements	Description
<b>Announce Sound</b>	The system prompt that will be played periodically to callers in the queue, such as 'All agents are busy. Please wait a minute. To leave a message, press 1; to end the call, just hang up'. The default setting is Default.
<b>Announce</b>	How often the system prompt is played. The default value is 0s.





<b>Frequency</b>	
<b>Busy Callback</b>	<b>Description</b>
<b>Enable Busy Callback</b>	When this feature is enabled, the caller can choose to hang up the call while hearing a corresponding voice prompt and this call still waits in line. Then once it is the caller's turn to transfer the call to an agent, IPPBX will start a call to this agent and wait for answers before dialing back to the caller to establish a connection. The default setting is False.
<b>Agent Busy Announce</b>	Select a voice file as the prompt for agent busy. The default setting is Default.
<b>Agent Busy Callback Key</b>	Press this key to enter the flow of dialing back upon agent busy. The default value is 2.

### 3.5.1.3.3. Key Press Event

<b>Key Events</b>	<b>Description</b>
<b>Option</b>	The keys that might be pressed after the caller hears the system prompt.
<b>Destination</b>	The destination the call will be transferred to after the caller's keypress.

### 3.5.1.4. Pickup Group

<b>Item</b>	<b>Description</b>
<b>Name</b>	User-defined name of an intercept group. Users can set intercept groups by service requirements, facilitating the members in a group to answer calls for each other. It must be filled in; otherwise, the configuration will fail to be saved.
<b>Member</b>	Select members for this group. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.

### 3.5.1.5. Ring Group

<b>Item</b>	<b>Description</b>
<b>Name</b>	User-defined name of a ring group. It must be filled in; otherwise the configuration will fail to be saved.
<b>Ring Group Number</b>	The number dialed to reach this ring group, with the default value range of 6200~6299 which can be modified in 'PBX->Preference->Extension Preferences'. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Ring Strategy</b>	Three options available: Simultaneous, Sequence, Random. *Simultaneous (default): All extensions ring at the same time. *Sequence: Ring one by one. Timeout by Second. *Random: Random select extensions, none-repetitive.
<b>Timeout Destination</b>	Select the destination to enter when agents in this ring group are all not answered. By default it is null.
<b>Ring Timeout(s)</b>	The timeout time to ring next extension, and also the timeout time to enter Timeout Destination if all extensions are unavailable. The default value is 30s.
<b>Enabled</b>	Set the status of the ring group. The default setting is True.
<b>Alert Info</b>	Set the content of the Alert-Info field. By default, it is null.



<b>Ring Back Scheme</b>	The ringback tone sent to the caller. The default setting is us-ring.
<b>CID Name Prefix</b>	The prefix of a caller ID name sent to the extension. By default, it is null.
<b>Extension Answer Confirm</b>	If set to Yes, the extension user will hear the following prompts upon picking up the call: Press 1 to answer; press 2 to reject. The default setting is No.
<b>Member</b>	Select members for this group. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.

### 3.5.1.6. Blocked List

Item	Description
<b>Name</b>	User-defined name of a blacklist. It must be filled in; otherwise, the configuration will fail to be saved.
<b>Match Mode</b>	Set the mode to match the caller number coming in through the trunk with the blacklist, two options available: Exact Match(default) and Regex Match.
<b>Blocked List Direction</b>	Set the direction of the blocked list. The default setting is Inbound.
<b>Blocked List Number</b>	An exact number in the blacklist.
<b>Regular Expression</b>	<p>Fill in following the rule of <a href="#">Regular Expression</a>.</p> <ol style="list-style-type: none"> <li>(1) ^123\$: "^" starts matching, "\$" ends matching, ^123\$ indicates strict matching</li> <li>(2) 123^(123 456)\$: " " means or, ^(123 456)\$ means to match 123 or 456</li> <li>(3) ^123 456\$: It means to match the string beginning with 123 or the string ending with 456</li> <li>(4) ^123[4-6]\$: "[" means to match any of the characters within, [4-6] is equivalent to [456], ^123[4-6]\$ means matching 1234, 1235, 1236</li> <li>(5) ^123\d\$: "\d" means any number from 0 to 9, ^123\d\$ is equivalent to ^123[0-9]\$</li> <li>(6) ^123\d+\$: "+" means to match one or more characters in front of it, ^123\d+\$ means matching at least 4 digit strings beginning with 123, such as: 1234, 12300, ..., 123456789, etc.</li> <li>(7) ^123\d*\$: "*" means to match 0 or more digits in front of it, ^123\d*\$ means matching at least 3 digit strings beginning with 123, such as: 123,1234,12300 ,...,123400000, etc.</li> <li>(8) ^123: It means any string beginning with 123, such as: 123, 12345, 123abc, etc.</li> <li>(9) 123\$: It means any string ending with 123, such as: 123, 666123, abc123, etc.</li> <li>(10) ^123\d{3}\$: "\d{3}" means to match 3 digits, ^123\d{3}\$ means matching any 6-digit string beginning with 123</li> <li>(11) ^(13[4-9] 147 15[0-2,7-9]): It indicates to match the string beginning with 134, 135, 136, 137, 138, 139, 147, 150, 151, 152, 157, 158, 159</li> <li>(12) .*: "." represents any character, .* means to match any string</li> </ol>
<b>Enabled</b>	Set whether to enable the black list feature. By default, it is True.



### 3.5.1.7. PIN Numbers

Item	Description
<b>Name</b>	User-defined name of a PIN number. It must be filled in; otherwise the configuration will fail to be saved.
<b>PIN List</b>	Multiple PIN numbers are allowed and should be separated by ','. This feature is used for such applications as conference, outbound routes which require entering the PIN number to verify authorities. It is null by default and must be filled in; Otherwise, the configuration will fail to be saved.
<b>Enabled</b>	Set whether to enable or disable the PIN list. It is null by default and must be filled in; otherwise the configuration will fail to be saved.

### 3.5.1.8. Speed Dial

Item	Description
<b>Name</b>	User-defined name of a speed dial, which must be unique. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Speed Dial Number</b>	Number of a speed dial, unique. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Destination</b>	Destination number that the speed dial number corresponds to. It is null by default and must be filled in; otherwise the configuration will fail to be saved.

### 3.5.1.9. Paging

Item	Description
<b>Name</b>	User-defined name of a call paging, which shall be unique. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Number</b>	Number of a call paging. The default value range is 6300~6399. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Type</b>	Two options available: Unilateralism(default) and Bidirectional.
<b>CallerID Name Prefix</b>	The prefix of a caller ID name of the call started by the call paging. It is null by default.
<b>Member</b>	Select members for this group. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.

### 3.5.1.10. DISA

Item	Description
<b>Name</b>	User-defined name of a DISA, which must be unique. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Response Timeout</b>	The maximum time waiting for the caller to press digits after prompt.
<b>Digit Timeout</b>	The maximum time permitted between two digits in dialing an extension number.
<b>Second Dial</b>	Set whether to enable the two-stage dial. The default setting is Enable.
<b>PIN Type</b>	Three options available: None, Single Pin and Pin List. If set to Single Pin or Pin List, the caller in DISA will hear the prompt for entering a password before inputting



	the callee number to dial.
<b>Outbound Routes</b>	Select an outbound route for DISA call out. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.

### 3.5.1.11. Call Back

Item	Description
<b>Name</b>	User-defined name of a callback, which must be unique. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Delay</b>	The delay time to call back after rejecting an incoming call.
<b>Strip</b>	Set how many digits will be stripped from the call number before the callback is placed. It is null by default.
<b>Prepend</b>	Set the digits to prefix the callback number before the callback is placed. It is null by default.
<b>Destination</b>	The destination which the callback will direct the call to. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Through</b>	*Auto (default) *From Come in *Select SIP Trunk

### 3.5.1.12. Emergencies

Item	Description
<b>Emergency Number</b>	The emergency number users fill in by actual requirements, such as 110, 911. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Trunk</b>	Choose trunks for dialing the emergency number. All extensions can make emergency calls through these trunks regardless of the Time Condition setting. When all the trunks are busy, the system will terminate an ongoing call to make sure the emergency call can be put through. The default setting is None.
<b>Announce</b>	When an emergency number is dialed, the system will make a notification call to the selected extension with a prompt. Multiple extensions are allowed. The default setting is None.



### 3.5.1.13. Multicast

<b>Multicast Type</b>	The default type is Triggler Multicast.
<b>Multicast Name</b>	The user-defined name of the multicast which must be unique.
<b>Multicast Number</b>	The multicast number, with the default range of 6800~6899.
<b>Multicast Address</b>	The multicast address, just keep one multicast address at the specific time. The default setting is 224.168.168.168.
<b>Multicast Port</b>	The multicast port, just keep one multicast port at the specific time. The default value is 34567.
<b>Prompt</b>	Set the multicast prompt tone.
<b>Enabled</b>	Set whether to enable the multicast feature. The default setting is Yes.

### 3.4.4.14. Wakeup Services

Item	Description
<b>Name</b>	User-defined name of the wakeup service, unique. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Prompt</b>	The alarm prompt. The default setting is Default.
<b>Custom Date</b>	User-defined alarm date, including day of week, etc. By default, it is unticked.
<b>Date</b>	Set the year, month and day information. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Time</b>	Set the time which is by default 00:00.
<b>Snooze Time</b>	Set the time interval for retry. The value range is $\geq 60$ and the default value is 600, calculated by second.
<b>Wakeup Times</b>	The wakeup times must be within 0 to 20. 0 means unlimited.
<b>Enabled</b>	Set whether to enable the wakeup service. The default setting is True.
<b>Wakeup Member</b>	The extension members that need the wakeup service. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.

## 3.6. Billing

### 3.6.1. Extensions

Item	Description
<b>Status</b>	Show if the extension is available.
<b>Extension</b>	Display the extension number.
<b>Name</b>	Display the name of the extension user.
<b>Billing Mode</b>	Show the billing method.
<b>Total Recharge</b>	Display the total extension call charges.
<b>Balance</b>	Display the balance of the extension call charge.
<b>Credit Limit</b>	Display the extension call charge limit
<b>Payment</b>	Show the extension payment method
<b>Operation</b>	Use the edit box to edit the extension, and click Top-up to recharge the extension.



### 3.6.2. Rates

Item	Description
<b>Description</b>	Enter the description.
<b>Order</b>	The order in which the rate matches. The smaller the value, the higher the priority.
<b>Match Pattern</b>	Match using regular expressions.
<b>Rate</b>	Cost deducted per billing cycle.
<b>Number Length</b>	To match the rate, the length of dialed number must be shorter or equal to this value. '0' means matching numbers of any length.
<b>Billing Increment</b>	The billing cycle, with the default value of '60s'.
<b>Time Condition</b>	Set the time period for rate matching. The default setting is 'All'.
<b>Extension</b>	Select the extensions that will use this rate.

### 3.6.3. Top-up History

Item	Description
<b>Top-up Date</b>	Display the top-up date.
<b>Extension</b>	Display the extension number.
<b>Balance before Top-up</b>	Show the balance before recharge.
<b>Top-up Amount</b>	Show the top-up amount.
<b>Top-up Balance</b>	Display the balance after recharge.

### 3.6.4. Statistics

Item	Description
<b>Time Range</b>	You can query the extension's billing statistics based on the call start and end time.
<b>Screening</b>	Filter based on call data.
<b>Statistics Type</b>	The statistic type.
<b>Trunk</b>	Filter by trunk.
<b>Extension</b>	Filter by extension.



### 3.6.5. Billing Settings

Item	Description
<b>Enable Billing</b>	Check it to enable the billing function.
<b>Decimal Places</b>	The decimal places to retain in calculating the amount.
<b>Insufficient Balance Prompt</b>	The prompt will be played when the extension do not have sufficient balance for calling outbound.
<b>Extension Locked Prompt</b>	The prompt will be played when the extension is locked.
<b>Low Balance Threshold</b>	When the balance reaches the threshold, the system will play the 'Top-up Prompt'. The value must be equal to or greater than '0'. The default value is '0'.
<b>Top-up Prompt</b>	The prompt will be played when the balance reaches the threshold.
<b>Hangup when the balance is Insufficient</b>	When the account balance is less than the credit limit, the call will be hung up automatically.

## 3.7. FCM Setting

### 3.7.1. SIP Settings

Item	Description
<b>Enable Session Timer</b>	Enable the timer for a SIP session which should be refreshed in a designated time. It is ticked by default.
<b>Session Timeout</b>	Set the maximum refresh interval for the session timer. The default value is 1800 seconds.
<b>UserAgent</b>	The content of the User-Agent field which is defined by users. The default setting is IPPBX.
<b>RTP Range</b>	Set the range of the RTP port used by the PBX. The default setting is 16384-32768.
<b>RTP AutoFix Timing</b>	RTP AutoFix Timing. The default setting is True.
<b>Nat Options Ping</b>	When it is set to True by default, the PBX will send the options message to all the terminals which register after NAT to keep the active connection to the terminal.
<b>DSCP Enabled</b>	In order to guarantee the communications Qos, the eight identification bytes in the IP header of the data packet are encoded to classify the service classes and differentiate the service priorities.
<b>Enable RTPAlways Adjust</b>	When this feature is enabled, UC will update the RTP port of the local end as long as the PTP port changes. The default setting is No.
<b>Trunk Profile Setting</b>	<b>Description</b>
<b>Enable External_LAN</b>	By ticking this option, you can create SIP trunks on the LAN port. It is ticked by default.
<b>Enable External_WAN</b>	By ticking this option, you can create SIP trunks on the WAN port. It appears only when the network mode is set to Double or Route. It is ticked by default.
<b>SIP IP</b>	The IP address to be monitored by using the SIP protocol. By default, it is the IP address of this network port.
<b>SIP Port</b>	The port to be monitored by using the SIP protocol. By default, it is 5080.
<b>Public SIP IP</b>	The SIP IP used for NAT traversal when the PBX stays in the LAN.
<b>Public RTP IP</b>	The RTP IP used for NAT traversal when the PBX stays in the LAN.



<b>100rel Enable</b>	Add <i>Supported:100rel</i> to the INVITE message.
<b>Enable TLS</b>	If this option is ticked, the SIP trunk will support UDP, TCP, TLS at the same time. It is unticked by default.
<b>TLS Only</b>	If this option is ticked, the calls on this SIP trunk will only support TLS.
<b>TLS SIP Port</b>	The default value is 5081.
<b>TLS Version</b>	The TLS version used by the SIP trunk. The default value is tlv1.
<b>TLS Certificate</b>	The certificate needed in case the PBX works as the client. It will be renamed to agent.pem after it is uploaded.
<b>Extension Profile Setting</b>	<b>Description</b>
<b>Enable Internal_LAN</b>	By ticking this option, you can create SIP extensions on the LAN port. It is ticked by default.
<b>Enable Internal_WAN</b>	By ticking this option, you can create SIP extensions on the WAN port. It appears only when the network mode is set to Double or Route. It is ticked by default.
<b>SIP IP</b>	The IP address to be monitored by using the SIP protocol. By default it is the IP address of this network port.
<b>SIP Port</b>	The port to be monitored by using the SIP protocol. By default, it is 5060.
<b>Public SIP IP</b>	The SIP IP used for NAT traversal when the PBX stays in the LAN.
<b>Public RTP IP</b>	The RTP IP used for NAT traversal when the PBX stays in the LAN.
<b>Enable WebRTC</b>	When it is enabled, users are allowed to register to IPPBX using page.
<b>WS Port</b>	Set the WS port. By default, it is 5066.
<b>WSS Port</b>	Set the WSS port. By default, it is 7443.
<b>Enable TLS</b>	If this option is ticked, the SIP extension will support UDP, TCP, TLS at the same time. It is unticked by default.
<b>TLS Only</b>	TLS If this option is ticked, the calls on this SIP extension will only support TLS.
<b>TLS SIP Port</b>	The default value is 5061
<b>TLS Version</b>	The TLS version used by the SIP extension. The default value is tlv1.
<b>Create CA Certificate</b>	When the PBX works as the Server, the CA certificate is used to generate Client and Server certificates, with the filename cafile.pem.
<b>Create Server Certificate</b>	It is a certificate needed when the PBX works as the Server, with the filename agent.pem.
<b>Create Client Certificate</b>	It is a certificate provided by the PBX for other clients to use, generated by using the same CA certificate of the Server certificate, with the filename client.pem.
<b>RTP CODEC</b>	<b>Description</b>
<b>RTPCodec</b>	Global RTP Codecs as list: G711A, G711U, G729, G722, GSM, G723, G726-16, speex@8000h@20i, speex@16000h@20i, AMR, PCMA@30i, PCMA@40i, PCMU@30i, PCMU@40i, VP8, VP9, H264, H263, H263-1998, H263-2000
<b>Certificate</b>	<b>Description</b>
<b>Create Certificate/Download Certificate</b>	Click it to generate and download the certificate.





## 3.7.2. Record Settings

Item	Description
<b>Internal Call Being Recorded Prompt</b>	The prompt that will be played to both the caller and the callee before the recording of internal calls. The default setting is None.
<b>Outbound/Inbound Calls Being Recorded Prompt</b>	The prompt that will be played to both the caller and the callee before the recording of outbound/inbound calls. The default setting is None.
<b>Record Start</b>	Set the timing to start recording, two options available: <i>After Answer</i> and <i>After Media</i> . The default setting is <i>After Answer</i> .
<b>Record Mode</b>	Set the recording mode, two options available: <i>Recording on One Side</i> and <i>Recording on Both Sides</i> . The default setting is <i>Recording on One Side</i> .
<b>Record Direction</b>	Set the recording direction, three options available: <i>Incoming Recording</i> , <i>Outgoing Recording</i> , <i>Incoming</i> and <i>Outgoing Recording</i> . The default setting is <i>Incoming Recording</i> .
<b>Record Sample Rate</b>	Set the sampling frequency of the recording, 8000 or 16000. The default value is 8000.
<b>Recording File Format</b>	Set the recording file format, WAV or MP3. The default format is WAV.
<b>Record Trunks</b>	Select trunks on which the calls will be recorded. By default, it is null.
<b>Record Extensions</b>	Select extensions on which the calls will be recorded. By default, it is null.
<b>Record Conferences</b>	Select conference rooms in which the calls will be recorded. By default, it is null.

## 3.7.3. Voice Prompts

### 3.7.3.1. Voice Prompts

Item	Description
<b>Music On Hold</b>	The music catalog to play when a call is being held. The default setting is default catalog.
<b>Play Call Forwarding Prompt</b>	If enabled, the system will play a prompt before transferring a call. By default, it is unticked.
<b>Music On Hold</b>	Set what to play when a call is being held during call forwarding. The default setting is Music On Hold.

### 3.7.3.2. System Prompt

Item	Description
<b>Upload System Prompts</b>	The supported compression format is zip. Please make sure of the integrity of voice packages to guarantee the normal use.
<b>Prompts List</b>	Display all the voice packages in IPPBX and allow you to select one as the system prompt.
<b>Language</b>	Three options are available: English (default), Chinese and Turkish.



### 3.7.3.3. Music on Hold

Item	Description
<b>Catalogue</b>	Select a catalogue of music on hold or press the following button + to create a new catalog.
<b>File Path</b>	Select a new music file and upload it to the list.
<b>File List</b>	Music files in the list can be played or removed.

### 3.7.3.4. Custom Prompt

Item	Description
<b>Upload</b>	The file to be uploaded should be: 8000Hz sampling rate, 16bit, single channel, wav or mp3 format.
<b>Record</b>	Define the name of a wav file, select an extension to record, then click the RECORD button. When the extension rings, pick up the call and say what you want to record.

## 3.7.4. Storage

### 3.7.4.1. Preference

#### 3.7.4.1.1. Storage Locations

Item	Description
<b>Voicemail</b>	A location to store your voicemail. It is Local Flash by default. If you plug TF or USB storage cards to the PBX, or add network disks, there will be more options: TF/SD, USB or the network disk (user-defined name).
<b>Recordings</b>	A location to store your recordings. It is Local Flash by default. If you plug TF or USB storage cards to the PBX, or add network disks, there will be more options: TF/SD, USB or the network disk (user-defined name).
<b>OTR</b>	A location to store your One Touch Recordings. It is Local Flash by default. If you plug TF or USB storage cards to the PBX, or add network disks, there will be more options: TF/SD, USB or the network disk (user-defined name).
<b>Logs</b>	A location to store your logs. It is Local Flash by default. If you plug TF or USB storage cards to the PBX, or add network disks, there will be more options: TF/SD, USB or the network disk (user-defined name).

#### 3.7.4.1.2. Storage Devices

Item	Description
<b>LOCAL</b>	Display the total storage, available size, usage of the local flash card, providing a reference for storage setting.
<b>TF/SD</b>	Display the total storage, available size, usage of the external TF card, providing a reference for storage setting.
<b>USB</b>	Display the total storage, available size, usage of the external USB card, providing a reference for storage setting.
<b>NETDISK</b>	Display the total storage, available size, usage of the added network disk, providing a reference for storage setting.



### 3.7.4.2. Auto Cleanup

#### 3.7.4.2.1. CDR Auto Cleanup

Item	Description
<b>Max Number of CDR</b>	Set the maximum number of CDR that should be retained. The default value is 10 and the value 0 means no limit. If the threshold is reached, the oldest CDR will be deleted.
<b>CDR Preservation Duration</b>	Set the maximum number of days when CDR should be retained. The default value is 0 which means no limitation. If the threshold is reached, the oldest CDR will be deleted.
<b>Max Number of Conference Sessions</b>	Set the maximum number of conference records allowed to save. If it is exceeded, the oldest record will be deleted. The default value is 5000, and 0 means no limitation.

#### 3.7.4.2.2. Voicemail and One Touch Recording Auto Cleanup

Item	Description
<b>Max Number of Files</b>	Set the maximum number of voicemail and one touch recording files that should be retained respectively for each extension. The default value is 30. If the threshold is reached, the oldest data will be deleted.
<b>Preservation Duration</b>	Set the maximum number of days for voicemail and one touch recording files to be retained respectively for each extension. The default value is 0 which means no limitation. If the threshold is reached, the oldest data will be deleted.
<b>Files Preservation Duration</b>	Set the maximum number of minutes for voicemail and one touch recording files to be retained respectively for each extension. The default value is 0 which means no limitation. If the threshold is reached, the oldest data will be deleted.

#### 3.7.4.2.3. Recordings Auto Cleanup

Item	Description
<b>Max Usage of Device</b>	Set the maximum storage percentage of recording files for the device. The default value is 80% and the value range is 30%~90%. If the threshold is reached, the oldest data will be deleted.
<b>Rec Preservation Duration</b>	Set the maximum number of days for recording files to be retained. The default value is 0 which means no limitation. If the threshold is reached, the oldest data will be deleted.

#### 3.7.4.2.4. Logs Auto Cleanup

Item	Description
<b>Max Size of Total Logs</b>	Set the maximum size of logs that can be saved per file. The default value is 50MB. If the threshold is reached, the oldest data will be deleted.
<b>Logs Preservation Duration</b>	Set the maximum number of log files to be saved. The default value is "7", and "0" means no limit.



### 3.7.4.2.5. Backups

Item	Description
<b>Auto Upload FTP</b>	After the information of the FTP server is configured, the recording file will be uploaded automatically. The default setting is False.
<b>FTP Address</b>	FTP server address, format: xxx:xxx:xxx:xxx or xxx:xxx:xxx:xxx:xx. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Username</b>	User name used on the FTP server
<b>Password</b>	Password used on the FTP server
<b>Upload Time</b>	Real Time (default): upload every 5 minutes Timing: Upload at a fixed time every day. If you select this item, you need to set the startup time and the default setting is 00:00.
<b>Delete Source File</b>	Set whether to delete the original recording file after it is uploaded. The default setting is False.
<b>FTP Test</b>	After the above configurations are set, you can test whether the FTP connection goes normal.

### 3.7.4. Preference

Item	Description
<b>Max Duration</b>	The maximum time length permitted for a call. The default value is 6000 seconds. 0 means no limit.
<b>Attended Transfer Caller ID</b>	The Caller ID that will be displayed on the recipient's phone. There options available: Auto, Transferor(default), Transferee. Example: 1002calling 1003, 1003transfers this call to 1004. * Transferor: Show 1003 on the extension 1004. * Transferee: Show 1002 on the extension 1004. * Auto: When 1003istransferring the call to 1004, the displayed number is 1003; when 1003 hangs up the call after its successful transfer and1002 is talking to 1004, the displayed number is 1002.
Extension Preferences	Description
<b>User Extensions</b>	The number range of user extensions. By default, it is 1000~5899.
<b>Ring Group Extensions</b>	The number range of user extensions in a ring group. By default, it is 6200~6299.
<b>Paging Group Extensions</b>	The number range of user extensions in a paging group. By default, it is 6300~6399.
<b>Conference Extensions</b>	The number range of user extensions in a conference room. By default, it is 6400~6499.
<b>IVR Extensions</b>	The number range of IVR extensions. By default, it is 6500~6599.
<b>Queue Extensions</b>	The number range of user extensions in a call center queue. By default, it is 6700~6799.
<b>Multicast Number</b>	The multicast number must be within this range. The default setting is 6800~6899.



API	Description
<b>Enable API</b>	Enable and disable the API interface. By default, it is disabled.
<b>Username</b>	API username. The default value is api.
<b>Password</b>	API user password. The default value is api.
<b>Event Information Address</b>	The address for the API to download the recording file. The default setting is the IP address set by the system.
<b>Incoming Call Control Timeout</b>	The timeout period in second for incoming calls controlled by the API.

### 3.7.5. Voicemail

Message Options	Description
<b>Max Messages per Folder</b>	The maximum number of messages to store in a single folder of voicemail. The default value is 100.
<b>Max Message Time</b>	The maximum length of a single piece of message. The default value is 300 seconds.
<b>Min Message Time</b>	The minimum length of a single piece of message. The default value is 3 seconds.
<b>Press 5 to leave a message</b>	If this option is ticked, you will hear the prompt: The phone you dial is unavailable now. Please press 5 to leave your message; if it is unticked, you will hear the prompt: The phone you dial is unavailable now. By default, it is ticked.
<b>Operator Breakout from Voicemail</b>	If this option is ticked, you will hear an extra prompt: Press 0 for operator. By default, it is unticked.
Greeting Options	Description
<b>Busy Prompt</b>	Select the greeting that will be played when the extension is busy. The default setting is Default.
<b>Unavailable Prompt</b>	Select the greeting that will be played when the extension is unavailable. The default setting is Default.
Playback Options	Description
<b>Announce Message Caller ID</b>	If this option is ticked, the extension number of the caller who left the message will be announced before the content of this message. By default, it is unticked.
<b>Announce Message Duration</b>	If this option is ticked, the duration of the message will be announced before the content of this message. By default, it is unticked.
<b>Announce Message Arrival Time</b>	If this option is ticked, the arrival time of the message will be announced before the content of this message. By default, it is unticked.
<b>View Voicemail</b>	Click to view the voicemail configuration of all extensions. Then click 'Messages' following each extension to view the exact voicemail messages.



## 3.8. System Setting

### 3.8.1. Security Strategy

#### 3.8.1.1. Security Strategy

Static Defense	Description
<b>Enable Firewall</b>	It is ticked by default.
<b>Enable Ping</b>	If it is unticked, the ping will be forbidden. By default, it is ticked.
<b>Drop All</b>	By default, it is unticked. Add at least one rule that allows TCP to connect to HTTPS:443, telnet:23 before ticking.
<b>Add</b>	The way to add a static security strategy is the same as adding a firewall rule for Linux.
Auto Defense	Description
<b>Port</b>	Enter the port for auto defense. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Protocol</b>	Select a protocol for auto defense, including TCP (default) and UDP.
<b>Number of IP Packets</b>	The allowed number of packets received within the 'time interval'. If the amount of data from a certain IP packet within the 'time interval' exceeds this threshold, the IP will be blacklisted. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Time Interval</b>	Time interval for receiving packets, calculated by second. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Blacklist</b>	Those calls which meet the above set conditions will be blacklisted herein. It can be manually deleted.
Fail2ban	Description
<b>Setup Fail2ban</b>	Enable Fail2ban. It is unchecked by default
<b>Ban Time</b>	The time in second when it is banned by firewall.
<b>Time Span</b>	Within the specified time range (s), after the host reaches the maximum number of matches, it will be banned.
<b>Max Retry</b>	The maximum number of times the host is matched
<b>Blocked List</b>	Those meeting the above setting conditions will be listed in the blacklist which can be deleted manually.



## 3.8.2. Network Settings

### 3.8.2.1. Basic Settings

Item	Description
<b>Hostname</b>	The default value is IPPBX.
<b>Mode</b>	Three options available: Dual, Bridge, Route. The default mode is Dual. Dual: Use Both Eth to communicate. Bridge: Working as switch with LAN address activate. Route: Working as a router, Only WAN used to communicate, LAN supports DHCP server in Router Mode.
<b>Default Interface</b>	When the IPPBX is in the Dual network mode, users should make an interface selection from LAN and WAN. The default setting is WAN.
IPv4	Description
<b>Network Mode</b>	Three options available: IP, DHCP, PPPoE, which are the same as the PC in settings.

### 3.8.2.2. OPENVPN

Item	Description
<b>Enable OpenVPN</b>	Choose whether to enable VPN.
<b>Server Address</b>	Enter the server address of OpenVPN.
<b>Server Port</b>	Enter the server port of OpenVPN. The default value is 1194.
<b>Protocol</b>	Select the protocol type. The client and server must use the same setting.
<b>Device Mode</b>	Select the network device. The client and server must use the same setting. TUN: A TUN device is a virtual point-to-point IP link; TAP: A TAP device is a virtual Ethernet adapter.
<b>Username</b>	Specify the username.
<b>Password</b>	Specify the password.
<b>Encryption</b>	Select the encryption method. The client and server must use the same setting.
<b>Compression</b>	Enable or disable compression for data stream. The client and server must use the same setting.
<b>CA Certificate</b>	Upload a CA certificate
<b>Certificate</b>	Upload a client certificate
<b>Key</b>	Upload a client key.
<b>TLS Authentication</b>	Enable or disable TLS authentication. If enabled, please upload a TA key.

### 3.8.2.3. Static Routes

Item	Description
<b>Add Routes</b>	The way to add routes is the same as that for the PC.



### 3.8.3. LDAP Server

### 3.8.4. Phone Book

Item	Description
<b>Phone Book</b>	This address book is only applicable for UCTALK APP, the access port is 10600, and the uploaded file must be named contacts.txt.
<b>Choose a File</b>	Select the address book to upload.
<b>Phone Book Template Download</b>	Download a template of the address book.

### 3.8.5. Email Settings

Item	Description
<b>Username</b>	The email account which is used to send emails, in the format of <a href="mailto:god@qq.com">god@qq.com</a> . It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Password</b>	The login password of the Email account used to send emails. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Display Name</b>	The display name for the email being sent. It is null by default and must be filled in; otherwise the configuration will fail to be saved.
<b>Send Mail Server</b>	Only the SMTP server is supported now whose format is smtp.qq.com. It is null by default and must be filled in; otherwise, the configuration will fail to be saved.
<b>Port</b>	The port of the SMTP server, with the default setting of 25. It must be filled in; otherwise the configuration will fail to be saved.
<b>Enable SSL/TLS</b>	Depend on if the mail server requires or not. It is ticked by default.
<b>SSL/STARTTLS</b>	Whether to enable SSL/TLS for encryption according to the sending server. The default setting is STARTTLS.
<b>Test Mail</b>	After settings are done, click Test Mail to check if the settings are correct. A test email will be send to the mailbox.





### 3.8.6. Service

Service	Description
<b>Auto Logout Time</b>	Set the automatic logout time of the webpage, up to 120 minutes, the default value is 60, calculated by minute.
<b>Protocol</b>	Select the type for webpage access, the default setting is HTTPS.
<b>Port</b>	Set the port for webpage access, the default value is 443.
<b>Redirect from Port 80</b>	If it is enabled, the access to Port 80 using the HTTP protocol will be automatically redirected to the corresponding port of HTTPS. By default, it is ticked.
<b>Enable Telnet</b>	Set whether to enable Telnet and the corresponding port. By default, it is ticked and the port is 23.
<b>Enable FTP</b>	Set whether to enable FTP and the corresponding port. By default, it is ticked and the port is 21. The FTP login username and password are the same as the admin user. After logging in, you can check the recording data under the storage space such as FLASH, USB mobile hard disk and TF card.
<b>Enable TFTP</b>	Set whether to enable TFTP. By default, it is ticked.

### 3.8.7. Event Setting

### 3.8.8. Feature Code

Digits Timeout	Description
<b>Feature Code Digits Timeout</b>	The maximum time waiting for the next feature code digit. The default value is 5000ms.
Recording	Description
<b>One Touch Record</b>	The feature code that is used to start call recording. The default code is *2.
<b>Agent Free/Busy</b>	Dial the function code+IVR number, you can record the prompt tone, replace the welcome word of IVR or restore the configuration. The default code is *10.
Voicemail	Description
<b>Check Voicemail</b>	The feature code that is used to check the voicemail. Press it and enter your password following the prompt. The default code is *97.
<b>Voicemail Main Menu</b>	The feature code that is used to access the global menu for voicemail. The default code is *98.
<b>Voicemail for Extension</b>	The feature code that is used to leave a voicemail to specified extensions or forward an incoming call to an extension's voicemail directly. The default code is *99.
Transfer	Description
<b>Blind Transfer</b>	Extension A presses this feature code in a call and dials Extension B after hearing the dial tone to transfer the call successfully. The default code is *1.
<b>Attended Transfer</b>	Extension A presses this feature code in a call, dials Extension B after hearing the dial tone, and hangs up the call after communication to transfer the call successfully. Press # to cancel. The default code is *4.
<b>Attended Transfer Timeout</b>	In negotiating a transfer, if the destination is not specified, the call will be transferred back after the set time. The default value is 15 seconds.
Intercept	Description
<b>Group Intercept</b>	By pressing this feature code, an extension can answer the incoming call to another



	extension in the same intercept group. The default code is *8.
<b>Extension Intercept</b>	By dialing this feature code plus an extension number, users can answer incoming calls to this extension. The default code is **.
<b>Intercom</b>	<b>Description</b>
<b>Intercom</b>	By dialing this feature code plus an extension number, users can start an intercom call to this extension. The default code is *80.
<b>Call Parking</b>	<b>Description</b>
<b>Call Parking</b>	Dial this feature code during a call to put the call on hold and park it at an extension number directed by the system. Any other phone can dial this extension number to resume the conversation.
<b>Directed Call Parking</b>	Dial this feature code and an extension number to park the call at the extension. Any other phone can dial this extension number to resume the conversation. The default feature code is #72. Note: If the specified number has been taken, an available number will be assigned by the system for parking.
<b>Park Extension</b>	By dialing this feature code, Extension A will be parked at another extension number. Other extensions can dial this extension number to resume the conversation with Extension A. The default feature code is 5900. Note: If the specified number has been taken, an available number will be assigned by the system for parking.
<b>Park Extension Start/Park Extension End</b>	The range of extensions where the call can be parked at. The default setting is 5901~5999.
<b>Park Timeout</b>	The maximum time for an extension allowed to park. The default value is 90 seconds.
<b>Call Forwarding</b>	<b>Description</b>
<b>Enable Forward All Calls</b>	By dialing this feature code, an extension forwards all calls to its voicemail; by dialing this feature code plus a designated number, an extension forwards all calls to this designated number. The default feature code is *72.
<b>Disable Forward All Calls</b>	Dial this feature code to disable forwarding of all calls. The default feature code is *73.
<b>Toggle Forward All Calls</b>	Dial this feature code to toggle forwarding of all calls. The default feature code is *72.
<b>Enable Forward When Busy</b>	By dialing this feature code, an extension forwards all calls to its voicemail when busy; by dialing this feature code plus a designated number, an extension forwards all calls to this designated number when busy. The default feature code is *90.
<b>Disable Forward When Busy</b>	Dial this feature code to disable call forwarding when busy. The default feature code is *91.
<b>Enable Forward No Answer</b>	By dialing this feature code, an extension forwards all calls to its voicemail when no answer; by dialing this feature code plus a designated number, an extension forwards all calls to this designated number when no answer. The default feature code is *92.
<b>Disable Forward No</b>	Dial this feature code to disable call forwarding when no answer. The default feature



<b>Answer</b>	code is *93.
<b>DND</b>	<b>Description</b>
<b>Enable Do Not Disturb</b>	Dial this feature code to put the extension into the DND state. The default feature code is *77.
<b>Disable Do Not Disturb</b>	Dial this feature code to take the extension out of the DND state. The default feature code is *78.
<b>Toggle Do Not Disturb</b>	Dial this feature code to toggle the DND state. The default feature code is *77.
<b>Call Monitor</b>	<b>Description</b>
<b>Listen</b>	Dial this feature code plus an extension number to monitor the extension. If this feature will work or not is related to the setting of monitor authority. The default value is *54.
<b>Whisper</b>	Dial this feature code plus an extension number to monitor the extension and whisper to it. If this feature will work or not is related to the setting of monitor authority. The default value is *55.
<b>Barge-in</b>	Dial this feature code plus an extension number to enter the call of this extension for monitoring. If this feature will work or not is related to the setting of monitor authority. The default value is *56.
<b>Forcible Hangup</b>	By dialing this feature code in a call, users can disconnect this call forcibly. The default feature code is *6.
<b>Listen Local</b>	Dial this feature code plus an extension number to monitor the conversation from the designated extension and the monitor cannot hear the voice from the other end. The default feature code is *93. Note: To monitor an extension, you need to configure the Monitor Settings for this extension first.
<b>Listen Remote</b>	Dial this feature code plus an extension number to monitor the conversation from the remote end of the designated extension and the monitor cannot hear the voice from the local designated extension. The default feature code is *94. Note: To monitor an extension, you need to configure the Monitor Settings for this extension first.
<b>Force Hangup on Monitor</b>	By dialing this feature code in a call, users can disconnect this call forcibly on monitor. Pressing this feature code+queue number on extension can disconnect this call forcibly. The default feature code is *64.
<b>Agent</b>	<b>Description</b>
<b>Agent Login/Logout</b>	By dialing this feature code plus a queue number, the extension can follow the prompt to log in and out the queue dynamically. The default feature code is *22.
<b>Agent Free/Busy</b>	When the status of the agent in the queue is 'On Break', it is set to 'available'; and when the status is 'available', it is set to 'On Break'.
<b>Agent Status ID</b>	By dialing this feature code plus a queue number, the extension can follow the prompt to add an extension to the designated queue or delete it from the queue. The default feature code is *23.
<b>BlackList</b>	<b>Description</b>



<b>Blacklist Add</b>	By dialing this feature code, the extension can follow the prompt to add a caller Id to the blacklist dynamically. The default feature code is *40.
<b>Blacklist Remove</b>	By dialing this feature code, the extension can follow the prompt to remove a caller Id from the blacklist dynamically. The default feature code is *41.
<b>Blacklist Add Last Call</b>	By dialing this feature code, the extension will add the latest call number to the blacklist. The default feature code is *42.
Query IP	Description
<b>Query LAN IP</b>	By dialing this feature code, the FXS extension can query such information as the IP address of LAN. The default feature code is *60.
<b>Query WAN IP</b>	By dialing this feature code, the FXS extension can query such information as the IP address of WAN. The default feature code is *61.
CC Routes	Description
<b>CC Routes</b>	When the extension is busy, dial this function key to implement the callback feature. The default feature code is *7.
Conference	Description
<b>Multiparty Conference</b>	The feature of multi-party conference, and the default feature code is *0.
Custom Status	Description
<b>Custom Status</b>	Press the feature code+1 or 2 digit-status code to push the API message. The default value is *76.

### 3.8.9. Date Time Settings

Item	Description
<b>Current System Time</b>	Display the current system date and time of the PBX.
<b>Time Zone</b>	The default setting is GMT+8:00 (Beijing).
<b>Daylight Saving Time</b>	Enable it to set the daylight saving time.
<b>Set up Manually</b>	Set the date and time manually. Tick the option <i>System Time</i> below and you can manually set the time.
<b>Synchronized with NTP Server</b>	Fill in the address or domain name of a NTP server and the PBX will synchronize with it in time automatically.
<b>System Time</b>	Set the system time manually.
<b>Enable NTP Server</b>	Tick this option to provide the NTP service for other devices.



## 3.9. Maintenance

### 3.9.1. Upgrade

Item	Description
<b>Manual Upgrade</b>	Use the upgrading file to upgrade the PBX version manually.

### 3.9.2. Users

This interface is used for adding WEB users. The default user and its password are both admin. The admin user can log in to the device through FTP to access the USB, network disk and local recording folder. The initial password is admin and you can modify it via the web page.

Item	Description
<b>Username</b>	User-defined, not allowed to be Admin.
<b>Password</b>	User-defined.
<b>Confirm Password</b>	Confirm your password.
<b>Language</b>	Select a language, Chinese or English.
<b>Groups</b>	Determine the user's authority
<b>Enabled</b>	Set the status of this account.

### 3.9.3. User Group

Item	Description
<b>Admin</b>	By default, an administrator group has the authority to checkstatus, call records and set recordings, as well as PBX, system and all functional modules. The exact authority of corresponding functional modules can be set by requirements.
<b>Public</b>	By default, a public group only has the authority to checkstatus and call records, as well as play and query recordings. The exact authority of corresponding functional modules can be set by requirements.
<b>User</b>	By default, a user group only has the authority to check status and call records, as well as play and query recordings. The exact authority of corresponding functional modules can be set by requirements.

### 3.9.4. Reboot

Item	Description
<b>Reboot</b>	Reboot the IPPBX system.
<b>Auto Reboot</b>	Set auto reboot in a day, a week or a month.



## 3.9.5. Backup and Restore

### 3.9.5.1. Backup

The backup content includes: System Configuration (default), Network Configuration, CDR, Operation Log Record, Customized Voice Prompt Files (default), SystemVoice Prompt Files. Users can customize the backup content.

### 3.9.5.2. Restore

Click the Browse button to select a backup file on your PC to restore your device.

### 3.9.5.3. Backup Lists

Display all lists of files that have been backed up with the backup time. Here you can select a backup file to restore.

## 3.9.6. Factory Reset

Item	Description
<b>Factory Reset</b>	Restore to the factory settings. You can choose not to restore the network settings. You should enter the correct verification code for reset, which is randomly generated

## 3.9.7. Operation Log

Item	Description
<b>Filter</b>	Main WEB operations will all be recorded to operation logs which can be queried by Username, IP Address, Start and End Date.
<b>Display</b>	The log list will display the operation time, the user who operated, the IP address, the type of operation as well as the operation details.

## 3.9.8. PBX LOG

Item	Description
<b>Log Level</b>	Six options available: CONSOLE, INFO, NOTICE, WARNING, ERROR, DEBUG. When DEBUG is ticked, you can set subsequently whether to output 'siptrace' which is the log of SIP messages.
<b>Log List</b>	The system will generate a log file every day which can be downloaded and deleted.

## 3.9.9. Log Viewer

Item	Description
<b>Log Viewer</b>	The key calls will be recorded in logs. On this interface you can filter those logs and sort them in descending order, show their line numbers and set their display size so as to better view the current log information.



### 3.9.10. Trouble Shooting

Item	Description
<b>Ethernet Capture Tool</b>	Set filter conditions for network capture, such as SIP only, both SIP and RTP, etc.
<b>Port Monitoring Tool</b>	Designate an FXO or FXS port for recording, as well as designate an FXO port to pick up or hang up the call.
<b>IP Ping</b>	Test connection of the destination via IP ping.
<b>Trace Route</b>	Test the network route and path as well as the response time.
<b>Call Testing</b>	Do simple call tests for UC.

### 3.9.11. Event Log

Item	Description
<b>Event Query</b>	All the logs that are reported by the trigger event will be recorded in the event log. You can query them by 'Event Type', 'Event Name', and 'Time'.
<b>Event Display</b>	The log list shows such detail information as time, event type, event name, and log content.

## 3.10. Call Detail Records

### 3.10.1. Call Detail Records

See below for all kinds of query conditions of call records.

Basic	Description
<b>Time Range</b>	Query CDR according to the start and end times.
<b>Call From</b>	Usually, it is the calling party number.
<b>Call To</b>	Search the CDR information according to the final destination number.
<b>Direction</b>	Three options available: Inbound, Outbound and Local
<b>Call Status</b>	Include such options as Answered, Missed, Voicemail, Cancelled, Failed, etc.
<b>Talk Duration</b>	Query CDR according to the time length of the call.
Advanced	Description
<b>Hangup Cause</b>	Query CDR according to the reason why the call ends.
<b>MOS Score</b>	Query CDR according to Mean Opinion Score (MOS) which is a measure of voice quality.
<b>CID Name</b>	Query CDR according to the name of caller identification (CID).
<b>OriginalDestination</b>	Query CDR according to the original destination of the caller.
<b>Trunk Name</b>	Query CDR according to the used trunk name.
<b>Outbound CallerId Number</b>	Query CDR according to the calling party number of the outgoing call.
<b>Last Destination</b>	Search the CDR information according to the latest destination.
<b>DID Number</b>	Search the CDR information according to the DID number.
<b>Wait Time in Queue</b>	Search the CDR information according to the wait time in queue.
<b>Agent Ring Time</b>	Search the CDR information according to the agent ring time.
<b>Agent Talk Time</b>	Search the CDR information according to the agent talk time.



## 3.10.2. Extension Summary

See below for all kinds of query conditions of call records.

Basic	Description
<b>Quick Select</b>	Count extension calls based on an approximate time. The default setting is today.
Custom Choose	Description
<b>Time Range</b>	Count extension calls according to the start and end times.

## 3.10.3. Conference Recording Session

See below for all kinds of query conditions of conference records.

Basic	Description
<b>Start Time/End Time</b>	Query conference calls according to the start and end times.
<b>Room Name</b>	Name of the conference room.
<b>Conference Center Number</b>	Number of the conference center.

## 3.11. Value Added Features

### 3.11.1. Authorization

#### 3.11.1.1. Authorization Info

Item	Description
<b>Serial Number</b>	Device serial number
<b>Max Sessions</b>	Concurrent number. The default value for FCM5404 is 30.
<b>Max extensions</b>	The number of extensions. The default value for FCM5404 is 150.
<b>Features</b>	The features supported by this device,

#### 3.11.1.2. Upload Authorization File

Item	Description
<b>Upload</b>	Manually upload the authorization file to the IPPBX and you can view the latest authorized information in 'Authorization Information'.

#### 3.11.1.3. Clear Authorization

Item	Description
<b>Clear</b>	Enter your password to do the clearance.





### 3.11.12. Auto Provision

Item	Description
<b>Auto Provision</b>	This feature is to generate a configuration file of the IP phone. The phone obtains the configuration file address by sending a paging packet, and downloads the configuration file automatically.
<b>Manufacturer</b>	Select the IP phone manufacturer that generates the phone book.
<b>Name</b>	When the type is Local Phone Book, it cannot be modified; when the type is Remote Phone Book, the phone book can be obtained by this name.
<b>Type</b>	Local Phone Book: IP phone is saved locally on the phone; Remote Phone Book: IP phone can be accessed by URL.
<b>Please Select a File</b>	Upload the phone book to be imported. Different manufacturers need to download different templates.
<b>Template</b>	Select a pre-configured template to automatically fill in such information as features, preferences, etc.
<b>Model</b>	IP phone model.
<b>Account</b>	Information required for the IP phone to register with the SIP server.
<b>Line Key Setting</b>	Set the shortcut keys of the IP phone.
<b>Features</b>	<p>Call Waiting: When the phone is in a call and another extension calls it, it will enter the waiting state if this feature enabled, or it will hang up the call if this feature disabled.</p> <p>Auto Answer: The IP phone automatically answers after ringing.</p> <p>Voicemail: When enabled, press the voicemail button on the phone to automatically send *97.</p> <p>Local Phone Book: When enabled, the configured phone book is automatically downloaded to the IP phone's address book.</p> <p>Key As Send: To determine if the IP phone directly calls out when pressing the # key.</p> <p>Remote Phone Book: You can directly view the phone book on the server by using this feature.</p>
<b>CODEC</b>	Set the codec supported by the IP phone.
<b>Config</b>	<p>Enable Auto Provision: It can provide automatic configuration function for IP telephone. By default it is enabled.</p> <p>Interface: Select the interface to provide automatic configuration for IP phones. By default it is LAN.</p> <p>Auto Edit: When it is enabled, the address and port of Auto Provision will be automatically modified according to the network and SIP settings.</p>
<b>Preference</b>	<p>Language: Set the display language of the IP phone.</p> <p>Admin Password: Set the login password for the IP phone.</p> <p>NTP Server: Set the NTP server address of the IP phone.</p> <p>Time Display Format: Set the time format for the IP phone.</p> <p>Date Display Format: Set the date format for the IP phone.</p> <p>Time Zone: Set the time zone for the IP phone.</p>





**FIBERME Communications LLC**

1749 Old Meadow Rd., Suite 401  
McLean, VA 22102, USA



[www.fiberme.com](http://www.fiberme.com)  
[sales@fiberme.com](mailto:sales@fiberme.com)

