Synway IPPBX Series

UC200

User Manual

Version 1.0.0

Synway Information Engineering Co., Ltd

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Content

Copyright Declaration

Revision History

Version	Date	Comments
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Chapter 1 Product Introduction

Thank you for choosing the SynwayUC Series IPPBX products (hereinafter referred to as 'UC200').It is a powerful, reliable and cost-effective VoIP solution that Synway developed for Enterprise Unified Communications, Customer Service Center, Hotel Voice Communications, etc.

1.1 Characteristic Features

- UC200-15 supports up to 15 concurrent calls and up to 60 extensions.
- UC200-30 supports up to 30 concurrent calls and up to 120 extensions.
- UC200-45 supports up to 45 concurrent calls and up to 180 extensions.
- UC200-60 supports up to 60 concurrent calls and up to 240 extensions.
- Integrates 4 FXO ports and 2 FXS ports for communications without power.
- Built in with multi-level IVR systems, supporting teleconferencing, call queuing, CDR, sound monitoring, call broadcasting, etc.
- Supports HTTPS, TLS and SRTP to guarantee the safety of communications.
- Includes two100-megabyte network interface cards, supporting three modes: Dual, Bridge and Route.
- Has 8G internal storage with extendable external storage via TF and USB cards.

1.2 Typical Application

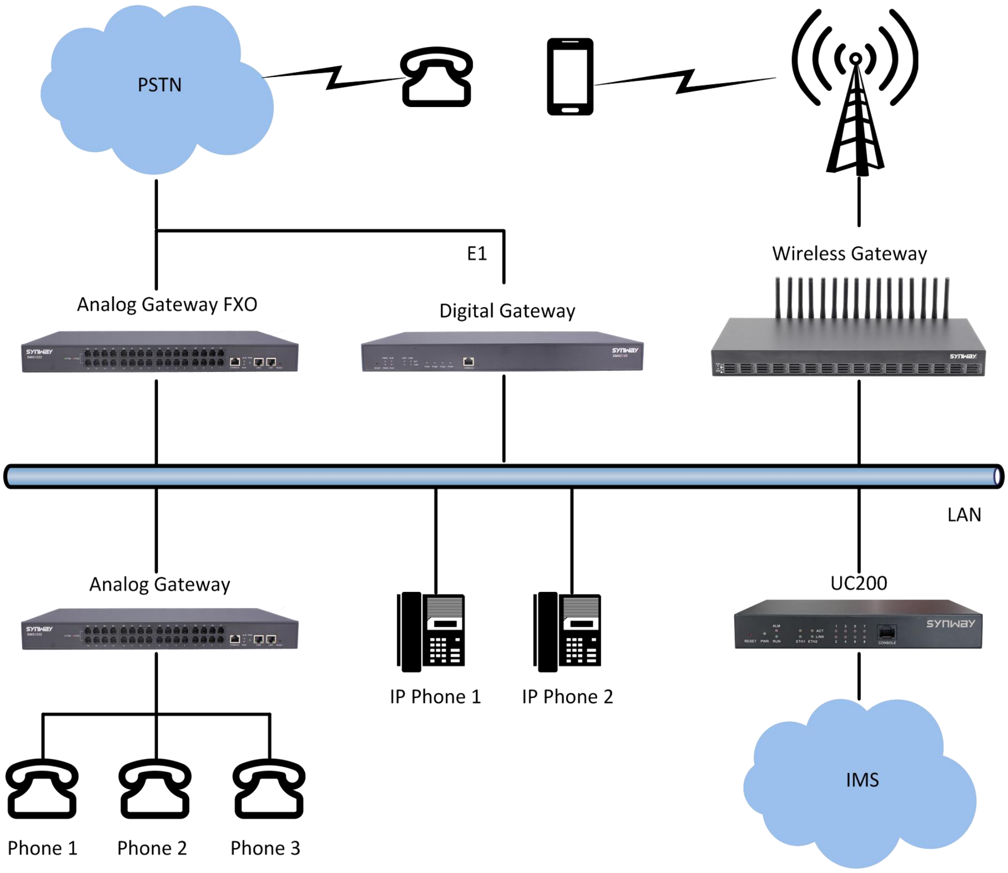

Figure 1- 1 Typical Application for IPPBX UC200

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: IVR,CDR,AlarmService,Broadcast,etc.

1.3 Feature List

Basic Features	Description
Extension	Allow users to make calls from extension to extension after registering SIP extensions to IPPBX.
Trunk	Allow extension users to make incoming and outgoing calls by SIP and FXO trunks with the help of inbound and outbound routes.
Inbound Routes	Enable forwarding calls from SIP or FXO trunks to internal extensions, IVR, conference, call center, DISA, callback systems, etc.

Outbound Routes	Enable making calls from extensions to external PSTN users.
CDR	Allow users to query and download detailed call records by condition on the webpage.
Call Recording	Record extensions, trunks, conferences, call centers; query, play and download the recording.
Call Forwarding	Extensions can be forwarded on different conditions such as‘Always’, ‘On Busy’, ‘No Answer’, or‘Not Registered’. Meanwhile, time condition settings are supported.
Call Waiting	This feature allows an FXS extension to receive another call while on the phone. It will make the feature of transfer on busy invalid.
Do Not Disturb	Reject all incoming calls to this extension.
Mobile Number	Multiple mobile numbers can be set for an extension to avoid missing any call to it.
Monitor	Support monitoring modes All, Listen, Whisper, Barge-in and monitoring authorities Disable, Enable All, Extensions to set for an extension.
Voicemail	Each extension supports an independent voicemail box as well as sending messages to a designated E-mail address.
Extension Security	Guarantee the security of extensions by password, ACL, UserAgent, etc.
Communication without Power	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.
IVR	Customize multi-level IVR.
Call Center Queue	Customize call center queues, providing multiple station ringing strategies to satisfy a variety of applications.
Conference	Support teleconferencing with more than 30 parties.
AutoCLIP	Redirect call to original extension.
Ring Groups	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).
Intercept Groups	Support interception of inside calls in a group and calls of specified extensions.
Call Broadcasts	Meet such requirements as broadcasting system.
Call Parking	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.
Blacklist	Numbers in the blacklist will be blocked to call in, or called, or both. It supports two modes: Exact Match and Regex Match.
DISA	Enable outside users using PBX service just like the system extensions to make calls.
Callback	Hang up the specified callers and let the PBX call them back.

Speed Dial	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.
Time Condition	This feature is supported for inbound routes, call forwarding, mobile extensions, etc.
PIN Code	This feature is supported for outbound routes, DISA, conference, voicemail, etc.
Signaling & Protocol	**Description**
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261
Voice	CODEC: G.711A, G.711U, G.729, G.722, AMR, G.726, GSM DTMF Mode: RFC2833, RFC4733, SIP INFO, INBAND
Network	**Description**
Network Protocol	Supported protocol: TCP/UDP, TLS, SSH, HTTPS, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.
Static IP	IP address modification support.
DHCP	IP address dynamic allocation support.
DNS	Domain Name Service support.
Security	**Description**
ACL	This feature is supported for extension registration and WEB access, etc.
Auto Defense	Allow users to customize dynamic firewall strategies to guarantee the security of system and network.
TLS&SRTP	Guarantee the security of signaling and voice communications.
Maintain & Upgrade	**Description**
WEB Configuration	Support of configurations through the WEB user interface.
Language	Chinese, English.
Software Upgrade	Support of user interface, IPPBX service, kernel and firmware upgrades based on WEB.
Tracking Test	Support of Ping and Tracert tests based on WEB.
SysLog Type	ERROR, WARNING, NOTICE, INFO, DEBUG, CONSOLE

1.4 Hardware Description

UC200 adopts an external 12V power supply.

The table below gives a detailed introduction to the interfaces, buttons and LEDs on UC200.

Interface	Description
LAN	Amount: 2
	Type: RJ-45

	Bandwidth: 10/100Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
FXS/FXO	Amount:6
	Type: RJ-11
	Maximum Transmission Distance: 1500m
	Charge Mode: Negative Anti-billing Supported
Console Port	Amount: 1
	Type: MiniUSB
	Baud Rate: 115200 bps
	Connector: MiniUSB
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
Power	External power input: 12V, ≥3A, positive inside and negative outside.
Button	**Description**
Reset	Restore the IPPBX to factory settings.
LED	**Description**
PWR	Indicates the power state. It lights up and keeps on after UC200 starts up with the power cord well connected.
SYS	Keeps on after the startup of Linux and regularly flashes after the bootup of the PBX service.
SD	Keeps on since the SD card is well connected and flashes in data reading and writing.
WAN	Keeps on since the network is well connected and flashes in data transmission.
LAN	Keeps on since the network is well connected and flashes in data transmission.
Channel Indicator	FXS and FXO channels are respectively marked by green and red LED after power on. 1. When the channel is idle, the LED Lights up; 2. When the channel is off-hook, the LED flashes slowly; 3. When the channel is ringing, the LED flashes fast.

For other hardware parameters, refer to Appendix A Technical Specifications.

Chapter 2 Quick Guide

This chapter is intended to help you grasp the basic operations of the UC series IPPBX products in the shortest time.

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

- UC200 *1
- External 12V Power Adapter *1
- Warranty Card *1
- Installation Manual *1

Step 2: Connect the network cable.

Connect the LAN port of UC200 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 UC200.

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 UC200.

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 UC series 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 holesare never jammed.
- During runtime, if the SYS indicator doesn't flashesregularlyandyou 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 3.1 System Login

Make sure the LANs of PC and IPPBX are in the same network segment. Enter the default IP address of PBX 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 Runtime Status

It includes two parts: System Status and PBX Status.

3.2.1 System Status

3.2.1.1 System Info

Item	Description
System Time	Current system time of IPPBX
Up Time	Running time of IPPBX since startup
Product	UC200, supporting registration of 200 SIP extensions by default which can be extended to 500.
Serial Number	Unique identifier of the device
uboot	Version information of the current uboot
kernel	Version information of the kernel
version	Version information of the current software

3.2.1.2 Network

3.2.1.2.1 LAN

Item	Description
TYPE	Static IP, DHCP or PPPoE
MAC	MAC address of LAN
IP Address	IP address of LAN
Gateway	Gateway address which displays only when LAN is the default network interface
Subnet Mask	Information about subnet mask
Preferred DNS Server	Information aboutpreferred DNS server
Alternate DNS Server	Information about alternate DNS server

3.2.1.2.2 WAN

Item	Description
TYPE	Static IP, DHCP or PPPoE
MAC	MAC address of WAN
IP Address	IP address of WAN
Subnet Mask	Information about subnet mask
Preferred DNS Server	Information aboutpreferred DNS server
Alternate DNS Server	Information about alternate DNS server

3.2.1.3 Performance

Item	Description
CPU	Real-time display of current CPU utilization
MEMORY	Real-time display of current memory utilization
LAN	Real-time display of current rate of LAN
WAN	Real-time display of current rate of WAN

3.2.1.4 Storage Usage

Item	Description
Flash	Display of total and used storage of the built-in flash card as well as the utilization
TF	Display of total and used storage of the outer TF card as well as the utilization
USB	Display of total and used storage of the outer USB card as well as the utilization

3.2.2 PBX Status

3.2.2.1 Extension

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

3.2.2.2 Trunk

Item	Description
Trunk Name	User-defined name of the trunk
Type	Trunk type, FXO or SIP
Trunk Status	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 Regiter trunk, display of

	status: fail to register/registered.
Domainname/IP/Port	For a SIP extension, display of domain name/IP addressof the registered IP/Soft phone; for an FXO extension, display of physical port number.

3.3 CDR

3.3.1Call Detail Records

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

Basic	Description
Time Range	Query CDR according to the start and end times.
Source	Usually it is the calling party number.
Destination	Number of the call destination
Direction	Three options available: Inbound, Outbound and Local
Status	Include such options as Answered, Missed, Voicemail, Cancelled, Failed, etc.
Talk Duration	Query CDR according to the time length of the call.
Advanced	**Description**
Hangup Cause	Query CDR according to the reason why the call ends.
MOS Score	Query CDR according to Mean Opinion Score (MOS) which is a measure of voice quality.
CID Name	Query CDR according to the name of caller identification (CID).
Caller Destination	Query CDR according to the original destination of the caller.

3.4 PBX

3.4.1 Extensions

3.4.1.1 Basic

General	Description
Type	Extension type, SIP or FXS
Extension	Extension number consists of all digits.
Password	It is generated randomly during the creation of a SIP extension and can be modified by users.
Enabled	Set whether to enable the extension or not.
Max Registrations	Maximum amount of registrations of this SIP extension, with the default value of 3.
Effective Caller ID Number	The callerID number for this extension to call outbound, i.e. the UserName field.
UserInfo	**Description**
Name	The callerID number for this extension to call outbound, i.e. the DisplayName field.
User Password	The password for this extension user to log into the system. Username is Name, while the default password is 'Pass' plus the extension number.
Voicemail Mail To	The email address to send voicemail to
Mobile Number	Fill in the mobile phone number of this extension user.

Prompt Language	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.

3.4.1.2 Features

Voicemail	Description
Voicemail Enabled	Once this feature is enabled, the call to this extension will enter the voicemail if failed. By default, the setting is True.
Voicemail Password	The password to enter the extension voicemail which is a randomly generated value by default and can be modified by users.
Voicemail Keep Local	Set whether to save the voicemail at IPPBX after it is sent with a specified email. By default, the setting is True.
Voicemail File	Set the way to send the voicemail. Two options are available: Download Link and Audio File Attachment, and the latter is the default setting.
Monitor	**Description**
Allow being monitored	Set if this extension can be monitored or not. *Disable: Not allow to be monitored, as default. *Enable All: Allow all extensions to monitor.*Extensions: Select extensions to monitor. .
Monitor Mode	Set the mode in which this extension monitors other ones. The default setting is None None: You will not be allowed to monitor calls; All: All the following 3 modes will be available for use; Listen:You can only listen into the call, but cannot talk (default feature code:*90) Whisper: You can talk to the extension you are monitoring without being heard by the other parties (default feature code: *91) Barge-in: You can talk to both parties (default feature code: *92)
Call Forwarding	**Description**
Always	Always redirect calls to the designated destination within the period set by the following time condition select box. The default setting is Disabled.
On Busy	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.
No Answer	Redirect calls to the designated destination if not answered within the period set by the following time condition select box. The default setting is Disabled.
Not Registered	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.
Follow Me	**Description**
Follow Me	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.
Do Not Disturb	**Description**

Do Not Disturb	When DND is enabled for an extension, it will reject all incoming calls.. The default setting is Disabled.

3.4.1.3 Advanced

RTP Settings	Description
Enable SRTP	When this feature is enabled, the RTP stream is encrypted, sharing the same certification with TLS. The default setting is False.
SIP Bypass Media	Set whether to send the media stream point to point or in transparent proxy mode.
RTP Codec String	Set RTP Codecs
Register Settings	**Description**
AuthACL	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.
Online Detection	Send the OPTIONS message to this extension to check if it is registered and reachable. The default setting is False.
SIP Force Expires	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.
SIP Expires Max Deviation	Reply to new REGISTER messages with time difference. This item should work with ***SIP Force Expires***. For example, if ***SIP Force Expires*** 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.
UserAgentAuthentication	It is null by default, which means not to verify the UserAgent field in the Register message. If it is not set to null, a SIP extension can register successfully only when the UserAgent field in the Register message conform with the character string of this configuration item.
Call Settings	**Description**
Call Timeout	Sets the maximum ringing duration in seconds for every call of this extension. The default value is 30s.
Max Call Duration	Set the maximum call duration in seconds for every call of this extension,Acall will be terminated once it exceeds the time. The default value is 6000s.
Outbound Restriction	When this feature is set to True, this extension cannot call out except for emergency numbers. The default settings is False.
Extension Trunk	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.
FXS Settings	**Description**
Min Flash Detection	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.
Max Flash Detection	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.
RX Volume	Set the volume in the direction from the analog phone to the FXS port. The value range is -7~7.
TX Volume	Set the volume in the direction from the FXS port to the analog phone. The value range is -7~7.
Echo Cancellation Level	The default value is 64ms
Enable Cut DTMF	Set the length of the in-band DTMF voice to cut. Do not set it too large lest normal voice signals be cut.
Enable DTMFPassthrough	Enable the DTMF passthrough during the conversation.
Call Waiting	Enable the Call Waiting feature for this extension.
Wait dial tone timeout	The maximum time of an FXO call to wait for the dial tone from the line since it is picked up.

3.4.2 Trunks

3.4.2.1 Basic

Item	Description
Trunk Type	Trunk type, SIP or FXO.
Trunk Name	User-defined, consisting of letters and digits.
Enabled	Enable or disable the trunk.
Transport	Three options available: UDP, TCP, TLS. TLS goes valid only if it is enabled in ***SIP Settings***.
Register	Set whether to register the SIP trunk, which is determined by the trunk provider. The default setting is False.
Profile	Two options available: WAN (default), LAN.
Trunk IP/Domain	IP address or domain name of the SIP trunk plus port number.
Username	Username of the registered SIP trunk
Auth Username	Used for SIP authentication. In most cases, it is the same with the username.
Password	The registration password of the SIP trunk.
Expire Seconds	The default value is 300 seconds.
RegFail Retry	The default value is 30 seconds.
Keep Inbound CallerID	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.
Enable Proxy	Support of proxy mode for trunks like IMS.
Outbound CallerId Name	CallerID name of this trunk displayed in an outbound call, having a higher priority than similar settings in ***Extensions***.
Outbound CallerID Number	CallerID number of this trunk displayed in an outbound call, having a higher priority than similar settings in ***Extensions***.

3.4.2.2 CODEC

Item	Description
Codec Preferences	Set the RTP codec for SIP trunk outbound calls. G711A，G711U，G729，GSM，G722，G723，iLBC，SPEEX，G726 are supported at present.If none is selected, all Codecs in SDP will be used by default; otherwise,, only the selected ones will beassigned.

3.4.2.3 Advance

VoIP Settings	Description
Send CID Type	* NONE: Put the CID information only in the From field; * Remote-Party-ID:Add the Remote Party-ID field with the CID information; * P-Asserted-Identify: Add the P-Asserted-Identify field with the CID information
OPTIONS Interval	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.
Send Privacy ID	When this item is set to True, the header field Privacy:id will be added to the INVITE message.
From User	Use the value of this item to override the UserName field in the From header field while sending the INVITE message.
From Domain	Use the value of this item to override the Domain field in the From header field while sending the INVITE message.
DNIS	**Description**
DNIS	Dial Number Identification Service allowsusers to set the display name of a SIP trunk incoming call according to the called number so that the called extension of IPPBX can specify which trunk the call comes in and which consultation the call is about just from the name displayed.
DNIS Name	The name of the caller ID displayed for the incoming call through this SIP trunk.
DNIS Number	The number of the callee ID of the incoming call through this SIP trunk according to which users determine the value of DNIS Name.
FXO	**Description**
TX Volume	Set the volume in the direction from the FXO port to the analog phone. The value range is -7~7.
RX Volume	Set the volume in the direction from the analog phone to the FXO port. The value range is -7~7.
Hangup Detection	**Description**
Hangup Detection Method	Two methods available: Busy Tone and Polarity.
Busy Count	Specify how many busy tones to wait for before hangup.
Busy Freq	Set the frequency of busy tones detected.
Answer Detection	**Description**
Answer Detection Method	Set whether to use the Polarity method to detect if the remote end picks up the call and answers. None: 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.
DID Number	Set the DID number for the incoming call through this FXO port.
Other Settings	**Description**
Limit Max Calls	This defines the maximum number of concurrent calls allowed in this trunk, 0 means unlimited
Caller ID Detection	Set whether to detect the Caller ID.
Polarity Delay	Set the minimum time interval for the answer polarity detection and the hangup polarity detection. The default value is 600ms.
CallerID After Polarity	When this item is set to True, the Caller ID will be detected after polarity reversal.
Ring Detect Timeout	The default value is 5000ms.
Echo Cancellation Level	The default value is 64ms.
Tone Scheme	Two options available: USA and China.

3.4.2.4 DOD

Item	Description
DOD	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.
DOD Name	The caller ID name of an outbound call.
DOD Number	The caller ID number of an outbound call.

3.4.2.5 Adapt Caller ID

Item	Description
Adapt Caller ID	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.
Match Mode	Use regular expression to match. ^ means starting the match; $ means ending the match; \d indicates a random number; .indicates a random character; \d+ indicates any digit number consisting of more than one byte; .* indicates any number consisting of digits or characters. For example, ^00\d+ indicates the match of all digit numbers starting with 00; .*99$ indicates the match of any character or digit number ending with 99.
Strip	Remove the prefix of an incoming call number.
Prepend	Add the prefix content after removing the prefix.

3.4.3 Inbound Routes

Item	Description
Name	User-defined name of this inbound route.
Enabled	Set whether to enable this route.
DID Pattern	Use regular expression to match. ^ means starting the match; $ means ending the

	match; \d indicates a random number; .indicates a random character; \d+ indicates any digit number consisting of more than one byte; .* indicates any number consisting of digits or characters. For example, ^00\d+ indicates the match of all digit numbers starting with 00; .*99$ indicates the match of any character or digit number ending with 99.
Caller ID Pattern	Same as the item ***DID Pattern***.
Destination	Multiple options available, such as Extensions, IVR Menus, Ring Groups, Conference Rooms, Call Center, etc.
Enable Fax Detection	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.
Fax Destination	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.
Enable Time Condition	The feature is disabled by default. Once enabled, it is required to set a destination corresponding to this time condition.
Distinctive Ring Tone	Send the INVITE message with the Alert-Info header field to the called extension to let it select different ring tone files based on the Alert-Info header field.
Order	Used to adjust the priority of multiple inbound routes.
Member Trunks	Select the trunks that can use this route.

3.4.4 Outbound Routes

Item	Description
Name	User-defined name of this outbound route.
Enabled	Set whether to enable this route.
Dial Patterns	Use regular expression to match. \d indicates a random number; .indicates a random character; \d+ indicates any digit number consisting of more than one byte; .* indicates any number consisting of digits or characters. For example, 00\d+ indicates the match of all digit numbers starting with 00; .*99 indicates the match of any character or digit number ending with 99.
Strip	Remove several digits from the prefix.
Prepend	Add several digits to the prefix.
Member Extensions	Add member extensions for controlling the outbound call authority. Only those extensions selected have the authority to use this route.
Member Trunks	Select the trunks that can use this route.
Password	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 proceeded 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 proceededonly if the entered PIN is correct.
Order	Used to adjust the priority of multiple outbound routes.Smaller Number means higher Priority.
Time Condition	Set which time period to use this route. Uncheck any option by default, which

	means no time limits on outbound calls.

3.4.5 Outbound Restrictions

Item	Description
Name	Name of this user-defined outbound restriction.
Time Limit	Set a time limit for calls. The default value is 5 minutes.
Number of Calls Limit	Set how many calls are allowed in the limited time. For example, if ***Time Limit*** 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 6th call, it will be locked.
Auto Cancel Restriction	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.
Enabled	Set whether to enable this outbound restriction rule.
Member Extensions	Select the extensions that use this restriction rule.

3.4.6 AutoCLIP

Item	Description
AutoCLIP	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.
View AutoCLIP List	A list of extension outbound calls.
Delete Used Records	If enabled, when an AutoCLIP record is matched, it will be automatically deleted afterwards.
Record Keep Time	Set how long each record will be kept in the AutoCLIP list. The default value is 8 hours.
Only Keep Missed Call Records	If enabled, the system will only keep records of outbound calls that are not answered by the called party in the AutoCLIP list.
Match Outgoing Trunk	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.
Record PSTN Trunk	If enabled, calls that go out through PSTN will be recorded to the AutoCLIP list.
Digits Match	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.

3.4.7 Call Features

3.4.7.1 IVR

Basic	Description
Name	User-defined IVR name.
IVR Number	The extension number that can be routed to this IVR.

Greet Long	It is played as the first prompt for entering the IVR menu.
Greet Short	It is played when the user doesn't enter any key or enters a wrong key.
Response Timeout	The time waiting for a digit input after prompt. The default value is 5000ms.
Inter-Digit Timeout	The maximum time between your entering of two adjacent DTMF digits. The default value is 3000ms.
Max Timeouts	Maximum number of timeouts before exit. The default value is 3.
Max Failures	Maximum number of retries before exit. The default value is 3.
Digit Length	Maximum number of digits allowed for the caller ID.
Direct Extension	Set whether the user can dial directly to extensions after hearing the IVR prompt.
Direct Outbound	Set whether the user can dial directly out after hearing the IVR prompt.
Advanced	**Description**
Invalid Sound	The prompt played in case of invalid keypress.
Exit Sound	The prompt played upon exiting the IVR menu.
Exit Action	The destination selected to enter after exiting the IVR menu.
Caller ID Name Prefix	The prefix of the caller ID name sent upon the call passing from IVR to an internal extension.
Ring Back	The ring back tone the caller will hear upon the call passing from IVR to an internal extension.
Key Press Event	**Description**
Keypress Event	Set the destinations respectively according to 0~9, *, # and invalid keys.

3.4.7.2 Conference Room

Item	**Description**
Room Name	User-defined name of a conference room.
Conference Center Number	The number dialed to reach this conference room.
Greeting	The greeting played upon joining this conference room.
Schedule	Set the start and end time for this conference room.
No Pin	Set whether a password is needed for entering this conference room. The default setting is True.
Max Members	The maximum number of members allowed in this conference room.
Wait for Moderator	If set, the participants could not hear each other until the moderator joins the conference.
Announce	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.
Mute Participant	If set to True, the participants expect for the moderator are not allowed to speak in this conference room. The default setting is False.
Allow Participant to Invite	If set to True, all participants are allowed to invite other users to enter this conference room. The default setting is True.
Moderator Member	Specify the moderator extension for this conference.

3.4.7.3 Call Center Queues

3.4.7.3.1 Basic

Item	Description
Queue Name	User-defined name of a call center queue.
Queue Number	The number dialed to reach this call center queue.
Agent Password	Set the password for dynamic agents to enter this call center queue.
Ring Strategy	Ring All: All available agents ring. Longest Idle Agent: 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.
Agent Call Timeout	The maximum time for each agent to ring. The default value is 15 seconds.
Max Wait Time	The maximum time a caller can wait in a queue before being pulled out, calculated by second. 0 means no time limit.
Timeout Action	Select the destination to enter when the call in the queue doesn't be answered in the maximum waiting time.
Agent Answer Announce	Announcement played upon the agent answers the call. The default setting is null.
Agent Retry Time	The interval time between the failed and new calls of an agent. The default value is 30 seconds.
Wrap Up Time	The interval time between the answer of an incoming call and the allocation of a new one.
Max Queue Length	Set how many callers are allowed to line in the queue.
Caller ID Name Prefix	The prefix of a caller ID name sent when the queue allocates a call to the agent.
Alert Info	Set the content of the Alert-Info field.

3.4.7.3.2 Caller Experience Settings

Item	Description
Music on Hold	Select the music on hold to play when the caller enters this queue.
Join When No Agent	If enabled, callers can join a queue that has no agents.
Max Wait Time with No Agent	The maximum waiting time for a caller in the queue that has no agents.
Join Announce	Announcement played to callers upon joining the queue.
Caller Position Announcements	**Description**
Announce Position	Announce the current position of the caller in the queue.
Announce Hold Time	Announce how long the caller shall wait in the queue.

Call Duration	The average call length estimated by users based on actual situations, used to calculate the waiting time for the caller.
Announce Frequency	Set how often to announce the queue position and the hold time.
Periodic Announcements	**Description**
Announce Sound	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'.
Announce Frequency	How often the system prompt is played.
Events	**Description**
Option Digits	The keys that might be pressed after the caller hears the system prompt.
DTMF Action	The destination the call will be transferred to after the caller's keypress.

3.4.7.4 Intercept Groups

Item	Description
Name	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.

3.4.7.5 Ring Groups

Item	Description
Name	User-defined name of a ring group.
Ring Group Number	The number dialed to reach this ring group.
Ring Strategy	Three options available: Simultaneous, Sequence, Random. *Simultaneous: All extensions ring at the same time. *Sequence: Ring one by one. Timeout by Second *Random: Random select extensions, none-repetitive.
Timeout Destination	Select the destination to enter when agents in this ring group are all not answered.
Ring Timeout(s)	The timeout time to ring next extension, and also the timeout time to enter Timeout Destination if all extensions are unavailable.
Alert Info	Set the content of the Alert-Info field.
Ring Back Scheme	The ringback tone sent to the caller.
CID Name Prefix	The prefix of a caller ID name sent to the extension.
Extension Answer Confirm	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.

3.4.7.6 BlackList

Item	Description
Name	User-defined name of a blacklist.
Match Mode	Set the mode to match the caller number coming in through the trunk with the blacklist, two options available: Exact Match and Regex Match.
BlackListNumber	An exact number in the blacklist.

Regular Expression	Fill in following the rule of Regular Expression.

3.4.7.7 PIN Numbers

Item	Description
Name	User-defined name of a PIN number.
PIN List	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.
Record in CDR	Set whether to record the PIN number in CDR.

3.4.7.8 Speed Dial

Item	Description
Name	User-defined name of a speed dial, which must be unique.
Speed Dial Number	Number of a speed dial.
Destination	Destination number that the speed dial number corresponds to.

3.4.7.9 Call Broadcasts

Item	Description
Name	User-defined name of a call broadcast, which shall be unique.
Number	Number of a call broadcast. The default value range is 6300~6399.
Type	Two options available: Unilateralism and Bidirectional.
CallerID Name Prefix	The prefix of a caller ID name of the call started by the call broadcast.

3.4.7.10 DISA

Item	Description
Name	User-defined name of a DISA, which must be unique.
Response Timeout	The maximum time waiting for the caller to press digits after prompt.
Digit Timeout	The maximum time permitted between two digits in dialing an extension number.
PIN Type	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.

3.4.7.11 Call Back

Item	Description
Name	User-defined name of a callback, which must be unique.
Delay	The delay time to call back after rejecting an incoming call.
Strip	Set how many digits will be stripped from the call number before the callback is placed.
Prepend	Set the digits to prefix the callback number before the callback is placed.
Destination	The destination which the callback will direct the call to.
Through	*Auto *From Come in *Select SIP Trunk

3.4.7.12 Emergencies

Item	Description
Emergency Number	The emergency number users fill in by actual requirements, such as 110, 911.
Trunk	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.
Announce	When an emergency number is dialed, the system will make a notification call to the selected extension with a prompt. Multiple extensions are allowed.

3.4.8 Time Condition

Item	Description
Time Condition	It can be set for such features as outbound routes, inbound routes, call forwarding, follow me.
Name	User-defined name of a time condition.
Type	Three options available: WorkTime, Holiday, Custom.
WorkTime	Multiple times allowed to set, including month, day of month, day of week, hour, minute.
Holiday	Multiple times allowed to set, including month, day of month, day of week.
Custom	Multiple times allowed to set, including month, day of month, week of month, day of week, hour, minute, as well as exclude holiday.

3.4.9 Feature Code

Digits Timeout	Description
Feature Code Digits Timeout	The maximum time waiting for the next feature code digit. The default value is 3 seconds.
Recording	
One Touch Record	The feature code that is used to start or stop call recording. The default code is *2.
Voicemail	
Voicemail Main Menu	The feature code that is used to access the global menu for voicemail. The default code is *98.
Voicemail for Extension	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**
Blind Transfer	Extension A presses this feature code in a call and dials Extension B after hearing the dial tone to transfer the call successfully.
Attended Transfer	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.

Attended Transfer Timeout	The timeout to transfer a call. The call will be transferred back after the set time. The default value is 15 seconds.
Intercept	**Description**
Group Intercept	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.
Extension Intercept	By dialing this feature code plus an extension number, users can answer incoming calls to this extension. The default code is **.
Intercom	**Description**
Intercom	By dialing this feature code plus an extension number, users can start an intercom call to this extension. The default code is *88.
Call Parking	**Description**
Call Parking	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. The default feature code is *5.
Park Extension	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.
Park Extension Start/Park Extension End	The range of extensions where the call can be parked at. The default setting is 5901~5999.
Park Timeout	The maximum time for an extension allowed to park. The default value is 90 seconds.
Call Forwarding	**Description**
Enable Forward All Calls	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.
Disable Forward All Calls	Dial this feature code to disable forwarding of all calls. The default feature code is *720.
Toggle Forward All Calls	Dial this feature code to toggle forwarding of all calls. The default feature code is *73.
Enable Forward When Busy	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 *74.
Disable Forward When Busy	Dial this feature code to disable call forwarding when busy. The default feature code is *740.
Enable Forward No Answer	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 *75.
Disable Forward No Answer	Dial this feature code to disable call forwarding when no answer. The default feature code is *750.
DND	**Description**

Enable Do Not Disturb	Dial this feature code to put the extension into the DND state. The default feature code is *78.
Disable Do Not Disturb	Dial this feature code to take the extension out of the DND state. The default feature code is *780.
Toggle Do Not Disturb	Dial this feature code to toggle the DND state. The default feature code is *77.
Call Monitor	**Description**
Listen	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 *90.
Whisper	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 *91.
Barge-in	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 *92.
Agent	**Description**
Agent Status	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.
Agent Status ID	By dialing this feature code plus a queue number, the extension can follow the prompt to query the agent status. The default feature code is *23.
BlackList	**Description**
Blacklist Add	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.
Blacklist Remove	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.

3.4.10 Voice Prompts

3.4.10.1 Voice Prompts

Item	Description
Music On Hold	The musiccatalog to play when a call is being held. The default setting is defaultcatalog.
Play Call Forwarding Prompt	If enabled, the system will play a prompt before transferring a call. The default setting is Disabled.
Music On Hold	Set what to play when a call is being held during call forwarding. The default setting is default.

3.4.10.2 System Prompt

Item	Description
Upload System Prompts	The supported compression format is zip. Please make sure of the integrity of voice packages to guarantee the normal use.

Prompts List	Display all the voice packages in IPPBX and allow you to select one as the system prompt.

3.4.10.3 Music On Hold

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

3.4.10.4 Custom Prompt

Item	Description
Upload	The file to be uploaded should be: 8000Hz sampling rate, 16bit, single channel, wav format.
Record	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 wantto record.

3.4.11 Voicemail

Message Options	Description
Max Messages per Folder	The maximum number of messages to store in a single folder of voicemail. The default value is 100.
Max Message Time	The maximum length of a single piece of message. The default value is 300 seconds.
Min Message Time	The minimum length of a single piece of message. The default value is 3 seconds.
Press 5 to leave a message	If this option is checked, you will hear the prompt: The phone you dial is unavailable now. Please press 5 to leave your message; if it is unchecked, you will hear the prompt: The phone you dial is unavailable now.
Operator Breakout from Voicemail	If this option is checked, you will hear an extra prompt: Press 0 for operator.
Greeting Options	**Description**
Busy Prompt	Select the greeting that will be played when the extension is busy.
Unavailable Prompt	Select the greeting that will be played when the extension is unavailable.
Playback Options	**Description**
Announce Message Caller ID	If this option is checked, the extension number of the caller who left the message will be announced before the content of this message.
Announce Message Duration	If this option is checked, the duration of the message will be announced before the content of this message.
Announce Message Arrival Time	If this option is checked, the arrival time of the message will be announced before the content of this message.

3.4.12 Records

Item	Description
Internal Call Being Recorded Prompt	The prompt that will be played to both the caller and the callee before the recording of internal calls.
Outbound/Inbound Calls Being Recorded Prompt	The prompt that will be played to both the caller and the callee before the recording of outbound/inbound calls.
Record Trunks	Select trunks on which the calls will be recorded.
Record Extensions	Select extensions on which the calls will be recorded.
Record Conferences	Select conference rooms in which the calls will be recorded.
Record Callcenters	Select call center queues in which the calls will be recorded.

3.4.13 Preference

Item	Description
Max Duration	The maximum time length permitted for a call. The default value is 6000 seconds.
Attended Transfer Caller ID	The Caller ID that will be displayed on the recipient's phone. There options available: Auto, Transferor, Transferee (default). Example: 500 calling 501, 501 transfers this call to502. * Auto: When 501 iscalling 502, the screen of the 502 extension will show 501 as thecallerid. When 500 is talking to 502, it shows 500. * Transferor: Show 501 all time. * Transferee: Show 500 all time.
Extension Preferences	**Description**
User Extensions	The number range of user extensions. By default it is 1000~5899.
Ring Group Extensions	The number range of user extensions in a ring group. By default it is 6200~6299.
Paging Group Extensions	The number range of user extensions in a paging group. By default it is 6300~6399.
Conference Extensions	The number range of user extensions in a conference room. By default it is 6400~6499.
IVR Extensions	The number range of IVR extensions. By default it is 6500~6599.
Queue Extensions	The number range of user extensions in a call center queue. By default it is 6700~6799.

3.4.14 SIPSettings

Item	Description
Enable Session Timer	Enable the timer for a SIP session which should be refreshed in a designated time. The default setting is enabled.
Session Timeout	Set the maximum refresh interval for the session timer. The default value is 1800 seconds.
UserAgent	The content of the User-Agent field which is defined by users.
Trunk Profile Setting	**Description**

Enable External_LAN	By checking this option, you can create SIP trunks on the LAN port.
Enable External_WAN	By checking this option, you can create SIP trunks on the WAN port. It appears only when the network mode is set to Double or Route.
SIP IP	The IP address to be monitored by using the SIP protocol. By default it is the IP address of this network port.
SIP Port	The port to be monitored by using the SIP protocol. By default it is 5080.
Public SIP IP	The SIP IP used for NAT traversal when the PBX stays in the LAN.
Public RTP IP	The RTP IP used for NAT traversal when the PBX stays in the LAN.
Enable TLS	If this option is checked, the SIP trunk will support UDP, TCP, TLS at the same time.
TLS Only	If this option is checked, the calls on this SIP trunk will only support TLS.
TLS SIP Port	The default value is 5081.
TLS Version	The TLS version used by the SIP trunk. The default value is SSLV23.
Extension Profile Setting	**Description**
Enable Internal_LAN	By checking this option, you can create SIP extensions on the LAN port.
Enable Internal_WAN	By checking this option, you can create SIP extensions on the WAN port. It appears only when the network mode is set to Double or Route.
SIP IP	The IP address to be monitored by using the SIP protocol. By default it is the IP address of this network port.
SIP Port	The port to be monitored by using the SIP protocol. By default it is 5080.
Public SIP IP	The SIP IP used for NAT traversal when the PBX stays in the LAN.
Public RTP IP	The RTP IP used for NAT traversal when the PBX stays in the LAN.
Enable TLS	If this option is checked, the SIP extension will support UDP, TCP, TLS at the same time.
TLS Only	TLS If this option is checked, the calls on this SIP extension will only support TLS.
TLS SIP Port	The default value is 5061
TLS Version	The TLS version used by the SIP extension. The default value is SSLV23.
TLS Certificate	Upload a TLS certificate used by this SIP extension.

3.4.15 Regular Expression

REGEX Matching Rule

Character	Description
“0”~”9”	Digits 0~9.
“^”	‘^’ means the starting of match. For example, ^13 indicates to match any number starting with 13.
“$	‘$’ means the ending of match. For example, 56$ indicates to match any number ending with 56.
“\d”	‘\d’ represents any digit number. \d{4} indicates to match any number of 4 digits. \d+ indicates to match any digit number consisting of more than one byte;
“.”	‘.’ indicates to match a random character which can be letters, *, #, digits 0~9.
“*”	‘*’ means to replicate the previous character. For example, \d* indicates to match digit numbers of any length; .* indicates to match characters of any length.
“[ ]”	‘[ ]’ is used to define the range for a number. Values within it only can be digits ‘0~9’, punctuations ‘-‘ and‘,’. For example, [1-3,6,8] indicates any one of the numbers 1, 2, 3, 6, 8.
“-”	’-’ is used only in ‘[ ]’ between two numbers to indicate any number between these two numbers.
“,”	’,’ is used to separate numbers or number rangesin‘[ ]’, representing alternatives. For example, [1,3,5]indicates any one of the numbers 1, 3, 5; [1-3,6-9]indicates any one of the numbers1, 2, 3, 6, 7, 8, 9.

The default value for IPPBX is usually \d*. The table below lists some matching rules commonly used.

Matching Rule	Description
\d*	Digit number of any length
01[3,5,8]\d{9}	Any 12-digit number starting with 013, 015 or 018
010[6-8]\d{7}	Any 11-digit number starting with 0106, 0107 or 0108
\d*110	Digit number of any length ending with 110
120	Full-match number 120
.*	Character of any length

3.5 System

3.5.1 Network Settings

3.5.1.1 Basic Settings

Item	Description
Hostname	The default value is UC200.
Mode	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.
Default Interface	When the IPPBX is in the Dual network mode, users should make an interface selection from LAN and WAN.
IPv4	**Description**
Network Mode	Three options available: IP, DHCP, PPPoE, which are the same as the PC in settings.
IPv6	**Description**
Network Mode	Two options available: Static IP, DHCP.

3.5.1.2 Static Routes

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

3.5.2 Security Strategy

Static Defense	Description
Enable Firewall	It is checked by default.
Enable Ping	If it is unchecked, the ping will be forbidden.
Add	The way to add a static security strategy is the same as adding a firewall rule for Linux.
Auto Defense	Description
Add	Add an auto defense rule for a device.

3.5.3 Date Time Settings

Item	Description
Current System Time	Display the current system date and time of the PBX.
Time Zone	The default setting is GMT+8:00 (Beijing).
Set up Manually	Set the date and time manually.
Synchronized with NTP Server	Fill in the address or domain name of a NTP server and the PBX will synchronize with it in time automatically.

3.5.4 Storage

3.5.4.1 Preference

3.5.4.1.1 Storage Locations

Item	Description
Voicemail	A location to store your voicemail. It is Local Flash by default. If you plug TF or USB storage cards to the PBX, there will be more options: TF or USB.
Recordings	A location to store your recordings. It is Local Flash by default. If you plug TF or USB storage cards to the PBX, there will be more options: TF or USB.
OTR	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, there will be more options: TF or USB.
Logs	A location to store your logs. It is Local Flash by default. If you plug TF or USB storage cards to the PBX, there will be more options: TF or USB.

3.5.4.1.2 Storage Devices

Item	Description
Local	Display the total storage, available size, usage of the local flash card, providing a reference for storage setting.
TF	Display the total storage, available size, usage of the external TF card, providing a reference for storage setting.
USB	Display the total storage, available size, usage of the external USB card, providing a reference for storage setting.

3.5.4.2 Auto Cleanup

3.5.4.2.1 CDR Auto Cleanup

Item	Description
Max Number of CDR	Set the maximum number of CDR that should be retained. The default value is 5000. If the threshold is reached, the oldest CDR will be deleted.
CDR Preservation Duration	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.

3.5.4.2.2 Voicemail and One Touch Recording Auto Cleanup

Item	Description
Max Number of Files	Set the maximum number of voicemail and one touch recording files that should be retained respectively for each extension. The default value is 50. If the threshold is reached, the oldest files will be deleted.
Files Preservation Duration	Set the maximum number of minutes when voicemail and one touch recording files should be retained respectively for each extension. The default value is 0 which means no limitation. If the threshold is reached, the oldest files will be deleted.

3.5.4.2.3 Recordings Auto Cleanup

Item	Description

Max Usage of Device	Set the maximum storage percentage of recording files for the device. The default value is 80%. If the threshold is reached, the oldest files will be deleted.
Rec Preservation Duration	Set the maximum number of days when recording files should be retained. The default value is 0 which means no limitation. If the threshold is reached, the oldest files will be deleted.

3.5.5 User Permission

Item	Description
Admin	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.
Public	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.
User	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.

3.5.6 Email Settings

Item	Description
Username	The email account which is used to send emails, in the format of god@qq.com.
Password	The login password of the Email account used to send emails.
Display Name	The display name for the email being sent.
Send Mail Server	Only the SMTP server is supported now whose format is smtp.qq.com.
Port	The port of the SMTP server.
Enable SSL/TLS	Depend on if the mail server requires or not.
Test Mail	After settings are done, click Test Mail to check if the settings are correct. A test email will be send to the mailbox.

3.6 Maintenance

3.6.1 Upgrade

Item	Description
Manual Upgrade	Use the upgrading file to upgrade the PBX version manually.

3.6.2 Reboot

Item	Description
Service Reboot	Reboot the IPPBX service program.
System Reboot	Reboot the IPPBX system.

Auto Reboot	Set auto reboot in a day, a week or a month.

3.6.3 Backup and Restore

Item	Description
Backup	Back up all PBX configurations to files.
Restore	In restoring configurations, you can choose not to restore the network settings and your user permission.

3.6.4 Reset

Item	Description
Reset	Restore to the factory settings. You can choose not to restore the network settings.

3.6.5 PBX LOG

Item	Description
Log Level	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.
Log List	The system will generate a log file every day which can be downloaded and deleted.

3.6.6 Log Viewer

Item	Description
Filter	Main WEB operations will all be recorded to operation logs which can be queried by Username, IP Address, Start and End Date.
Display	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.

3.6.7 Trouble Shooting

Item	Description
Ethernet Capture Tool	Set filter conditions for network capture, such as SIP only, both SIP and RTP, etc.
Port Monitoring Tool	Designate an FXO or FXS port for recording.
IP Ping	Test connection of the destination via IP ping.
Trace Route	Test the network route and path as well as the response time.

Appendix A Technical Specifications

Dimensions

186×30×108mm³

Weight

0.83 kg

Environment

Operating temperature: 0℃—45℃

Storage temperature: -20℃—85℃

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

Amount: 2 (10/100 BASE-TX (RJ-45))

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

FXS Port

Amount: 2

Type: RJ11

Maximum transmission distance: 1500m

Impedance

Telephone line impedance: Compliant with the national standard impedance for three-component network

Console Port

Amount: 1 (MiniUSB)

Baud rate: 115200bps

Connector: MiniUSB Connector

Data bits: 8 bits

Stop bit: 1 bit

Parity unsupported

Flow control unsupported

Note: Follow the above settings to configure the serial port; or it may work abnormally.

Power Requirements

Input power:

12V DC not less than 3A

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

Audio Encoding & Decoding

G.711A	*64 kbps*
G.711U	*64 kbps*
G.729A/B	*8 kbps*

Sampling Rate

8kHz

Appendix B Troubleshooting

Q1. What to do if I forget the IP address of UC200?

There are two ways to get the IP address:

1) Long press the Reset button on the gateway to restore to factory settings. The default IP address is 192.168.1.101 (WAN) or 192.168.0.101 (LAN).

2) Dial the corresponding function key through an FXS port to query the IP address. See 3.4.9 Function Key for more details.

Q2. Which RTP codecs are supported by UC200?

At present, the supported RTP codecs are: G.711A, G.711U, G.729, G.722, G.723, G.726, GSM and AMR.

Q3. How to configure the features Communication without Power for UC200?

The feature **Communication without Power** is implemented in hardware. Once the power to the device is cut off, the station which is linked with the FXS port of UC200 and the trunk which is linked with the FXO port will connect to each other directly and keep the good communications between phones and networks. The FXS and FXO ports are one-to-one correspondence.

Appendix C Technical/sales Support

Thank you for choosing Synway. Please contact us should you have any inquiry regarding our products. We shall do our best to help you.

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