ZYCOO
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CooVox Series User Manual(Admin)

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Safety Notice

Please read the following safety notices before installing or using this IP PBX. They are crucial for safe and reliable operation of the device. Failure to follow the instructions contained in this document may result in damage to your PBX and void the manufacturer's warranty.

1. Please use the external power supply which is included in the package. Other power supplies may cause damage to the device, affect the performance or induce noise.

2. Before using the external power supply in the package, please check your building power voltage. Connecting to Inaccurate power voltage may cause fire and damage.

3. Please do not damage the power cord. If the power cord or plug is impaired, do not use it. Connecting a damaged power cord may cause fire or electric shock.

4. Ensure the plug-socket combination is accessible even after the PBX is installed. In order to service the PBX it will need to be disconnected from the power source.

5. Do not drop, knock or shake the device. Rough handling can break internal circuit boards.

6. Do not install the device in places where there is direct sunlight. Also do not place the device on carpets or cushions. Doing so may cause the device to malfunction or cause a fire.

7. Avoid exposing the device to high temperature (above 40°C), low temperature (below -10°C) or high humidity. Doing so could cause damage and will void the manufacturer warranty.

8. Avoid letting the device come in contact with water or any liquid which would damage the device.

9. Do not attempt to open it. Non-expert handling to the device could cause damage and will immediately void the manufacturer warranty.

10. Consult your authorized dealer for assistance with any issues or questions you may have.

11. Do not use harsh chemicals, cleaning solvents, or strong detergents to clean the device.

12. Wipe it with soft cloth that has been slightly dampened in a mild soap and water solution.

13. If you suspect your device has been struck by lightning, do not touch the device, power plug or phone line. Call your authorized dealer for assistance to avoid the possibility of electric shock.

14. Ensure the PBX is installed in a well ventilated room to avoid overheating and damaging the device.

15. Before you work on any equipment, be aware of any hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents. If you are in a situation that could cause bodily injury.

Chapter 1　Brief Introduction

1.1 Brief Introduction of CooVox Series

The CooVox Series IP PBXs are designed to provide SMEs (small & medium enterprises) with all the standard and advanced features that are normally only available from large, expensive, legacy PBX manufacturers. Aimed at businesses with up to 100 extensions, the CooVox Series IP PBXs are based on SIP and OpenSource Asterisk 1.8, with whose innovative modular telephony design, that is easy to expand the PBX to meet the growing needs of your business.

CooVox Series IP PBXs come in four sizes: U20 / U50 / U60 / U100.

Each model will be introduced in detail below:

CooVox-U20 is configured with 2 analog ports:

	FXS	FXO
CooVox-U20	1	1
	0	2

CooVox-U50 consists of two main parts：U50 Host and Modules. There are 2 slots in the system and the modules can be utilized as in the diagram below:

U50 Slot / U50 Module	Slot 1	Slot 2
4FXS	✔	✔
4FXO	✔	✔
2FXOS	✔	✔
2GSM	✔	✔
4GSM	✔	✔
1PRI	✔	✖
4BRI	✔	✖

CooVox-U60 is configured with 24 analog ports:

	2FXS	2FXO	FXOS
CooVox-U60	✔	✔	✔

CooVox-U100 consists of two main parts：U100 Host and Modules. There are 2 slots in the

system and the modules can be utilized as in the diagram below:
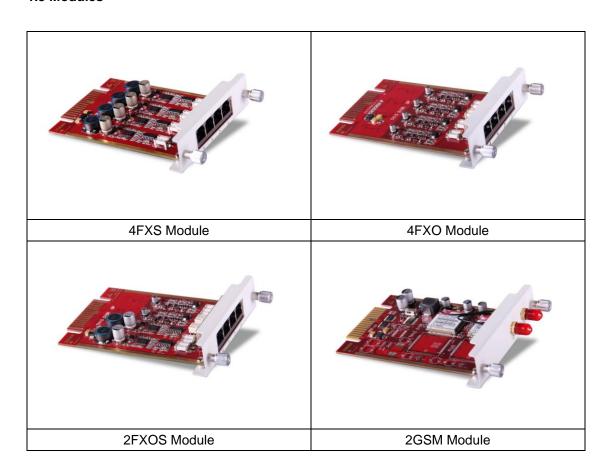
U100 Slot / U100 Module	Slot 1	Slot 2
4FXS	✔	✔
4FXO	✔	✔
2FXOS	✔	✔
2GSM	✔	✔
4GSM	✔	✔
1PRI	✔	✔
4BRI	✖	✔

1.2 Main Features

1. SIP/ IAX Extension Registration
2. Video Call
3. USB Mobile Hard Disk Record (Scalable)
4. IP Phone Provisioning (Grandstream /Yealink/Fanvil IP Phone)
5. Call Record /Ring Group Record/ Call Queue Record
6. Web-based Administration and configuration
7. Web-based Extension User Management
8. Voicemail
9. Caller ID
10. Call Parking
11. Call Forward
12. Call Transfer
13. Call Waiting
14. Call Center Queues
15. Black List
16. Phonebook
17. Flexible Dial Plan
18. Virtual Fax (fax to email, and email to fax)
19. DID
20. Dial by Name
21. Speed Dial
22. Do Not Disturb
23. Callback
24. Skype for SIP
25. Ring Group

26. Conference Bridge (Three Conferences)
27. Music On Hold
28. DISA (Direct Inward System Access) /Paging And Intercom
29. Call Detail Record
30. IP Phone Feature Code
31. BLF(Busy Lamp Field)
32. Static /DHCP /PPPoE Network Access
33. DHCP Server
34. System Backup
35. T.38 Pass-through
36. Audio Codec: G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX
37. Video Codec: H.261/ H.263 / H.263+ / H.264
38. VPN Server (L2TP / PPTP / OpenVPN, up to 10 connections for VPN clients)
39. VPN Client (L2TP / PPTP / OpenVPN / N2N)
40. SNMPv2
41. IPv4 / IPv6
42. DDNS(Dyndns.org /No-ip.com /zoneedit.com)

1.3 Modules

4FXS Module	4FXO Module
2FXOS Module	2GSM Module

4GSM Module	4BRI Module
1E1/T1 Module	32 EC Module

1.4 Hardware Interfaces

1.4.1 CooVox-U20

CooVox-U20 Front Panel

Indication	Function	Status	Explaination

CooVox-U20 Rear Panel

1 * Reset Button

1 * Power Interface (*DC 12V 2A*)

1 * Ethernet Interface (10/100Mbps)

2 * Analog Ports(FXO/FXS)

1 * UMTS Port

U20 LEDs Indication

PWR	Power Status	On	Power On
		Off	Power Off
SYS	System Status	Blink	System Works
		Off	System Fails
ETH	WAN or LAN Data Status	Blink	Data Transport
G	GSM or UMTS(3G) Status	Off	Module not running
		64ms On/800ms Off	Module doesn't find network
		64ms On/3000ms Off	Module finds network
1	FXO	Red	Channel Loading Success
		Blink	Channel Ringing
		Off	Channel Loading Failure
2	FXS	Green	Channel Loading Success
		Blink	Channel Ringing
		Off	Channel Loading Failure

1.4.2 CooVox-U50

CooVox-U50 Front Panel

CooVox-U50 Rear Panel

1 * Reset Button

1 * Power Interface (*DC 12V 2A*)

1 * Ethernet Interface (10/100Mbps)

1 * Console Interface

1 * USB Interface

Slot 1 for Analog/GSM/PRI/BRI Module Cards

Slot 2 for Analog/GSM Module Cards Only

Indication	Function	Status		Explaination
PWR	Power Status	On		Power On
		Off		Power Off
SYS	System Status	Blink		System Works
		Off		System Fails
ETH	Data Status	Blink		Data Transport
		Off		No Data Transport
USB	U-disk or UMTS(3G) Status	Off		Module not running
		On		Module Works
1-4(SLOT1/2)	SLOT 1/2 Status	FXS	Green	Channel Loading Success
			Blink	Channel Ringing
			Off	Channel Loading Failure
		FXO	Red	Channel Loading Success
			Blink	Channel Ringing
			Off	Channel Loading Failure
		GSM	Red	Channel Loading Success
			Blink	Channel Ringing
			Off	Channel Loading Failure
		E1/T1 (PRI/ R2) (Only for Slot 1) L1	Red	Module Loading Success
			Off	Module Loading Failure
		L2	Red	CPE signal
			Green	NET signal
			Off	No signal
		L3	Red	SS7 signal
			Green	MFCR2 signal
			Off	No signal
		L4	Red	Disconnected/ Alarm
			Green	Connected/ No Alarm
		BRI (Only for Slot 1)	Red	TE Mode
			Green	NT Mode
			Off	Module Loading Failure

1.4.3 CooVox-U60

CooVox-U60 Front Panel

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CooVox-U60 Rear Panel

1 * Power Interface

1 * Power Switch

2 * Ethernet Interfaces (10/100/1000Mbps)

1 * VGA Interface

2 * Audio Interfaces

2 * USB Interfaces

1 * Hardware Echo Cancellation Interfaces (onboard)

1 * UMTS Interface for 3G Data (onboard)

24 * Analog Ports (FXO/FXS)

U60 LED Indication

Indication	Function	Status		Explaination
PWR	Power Status	On		Power On
		Off		Power Off
SYS	System Status	Blink		System Works
		Off		System Fails
ETH	Data Status	Blink		Data Transport
		Off		No Data Transport
1-24 SLOTS	SLOT 1-24 Status	FXS	Green	Channel Loading Success
			Off	Channel Loading Failure
		FXO	Red	Channel Loading Success
			Off	Channel Loading Failure

1.4.4 CooVox-U100

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CooVox-U100 Front Panel

CooVox-U100 Rear Panel

1 * Reset Button

1 * Power Interface

1 * Power Switch

2 * Ethernet Interfaces (10/100 Mbps)

1 * VGA Interface

2 * USB Interfaces

2 * Audio Interfaces

SLOT 1 for any Module Cards (4FXO/ 4FXS/ 2FXOS/ 4GSM/ 2GSM/ 1PRI)

SLOT 2 for any Module Cards (4FXO/ 4FXS/ 2FXOS/ 4GSM/ 2GSM/ 1PRI/ 4BRI)

U100 LED Indication

Indication	Function	Status		Explaination
PWR	Power Status	On		Power On
		Off		Power Off
SYS	System Status	Blink		System Works
		Off		System Fails
ETH	Data Status	Blink		Data Transport
		Off		No Data Transport
1-4(SLOT1/2)	SLOT 1 /2 Status	FXS	Green	Channel Loading Success
			Blink	Channel Ringing
			Off	Channel Loading Failure
		FXO	Red	Channel Loading Success
			Blink	Channel Ringing
			Off	Channel Loading Failure
		GSM	Red	Channel Loading Success
			Blink	Channel Ringing
			Off	Channel Loading Failure
		E1/T1	L1 Red	Module Loading Success
			Off	Module Loading Failure
			L2 Red	CPE signal
			Green	NET signal
			Off	No signal
			L3 Red	SS7 signal
			Green	MFCR2 signal

			Off	No signal
		L4	Red	Disconnected/ Alarm
			Green	Connected/ No Alarm
		BRI	Red	TE Mode
		(Only for	Green	NT Mode
		Slot 2)	Off	Module Loading Failure

1.4.5 Model Comparison Table

Items		CooVox-U20	CooVox-U50	CooVox-U60	CooVox-U100
System Capacity	Concurrent Calls	10	20	80	80
	Extension Users	30	100	200	500
	Voicemail and Recording	21,000 mins (.gsm)	21,000 mins (.gsm)	200,000 mins (.gsm)	2,500,000 mins (.gsm)
		3000 mins (.wav)	3000 mins (.wav)	20,000 mins (.wav)	270,000 mins (.wav)
Hardware Capacity	SDRAM	128MB DDR2	256MB DDR2	1GB DDR3	2GB DDR3
	Memory (default)	4GB SD card	4GB SD card	32GB SSD	500GB HDD or 32GB SSD
Power Supply	Input	AC 100-240V	AC 100-240V	AC 100-240V	AC 100-240V
	Output	DC 12V/1A	DC 12V/2A	N/A	N/A

1.4.6 Environmental Requirements

1. Working Tempreture: 0 °C ~40 °C
2. Storage Tempreture: -20 °C ~ 55 °C
3. Humidity: 5~95% Non-Condensing

1.4.7 Packing List

CooVox Host	1 set
Power Supply	1 piece
Ethernet Cable	1 piece
Quick Installation Guide	1 piece
Warranty Card	1 piece

Notice:
1) ZYCOO Module cards will only function in CooVox IP PBX from ZYCOO;
2) Module cards for CooVox-U50/U100 will be packed separately but contained in the same package.

Chapter 2   Getting Started

(Take CooVox-U50 as example for the guide)

2.1 Before Configuration

What kind of IP Phones can be used with this device?

1.   FXS Interface: Analog Phone or fax machine

2.   SIP Extension: CooFone Series and ZP Series IP Phones provided by ZYCOO

           (D30/ D30P/ D60)

            Any standard SIP Phone based on SIP/ IAX2 protocol
            (eg: CISCO, Grandstream, Yealink, Polycom, Siemens, Snom,etc.)

2.2 Before Making a Call

2.2.1 Login IP PBX

Getting IP Address

There are three ways to set the IP address: Static, DHCP, PPPoE.

Default IP: 192.168.1.100:9999

Notice: you have to add port number 9999 after this IP address.

Defaults and Function Key

1.   Web Panel User name: admin
2.   Web Panel Password: admin
3.   *60   Enter Voicemail Box
4.   900/901/902   Default three conference room numbers
5.   #   Blind Transfer
6.   *2   Attended Transfer
7.   *   Disconnect Call

Administrator Login

After connecting the CooVox IP PBX to the local area network and setting your laptop to the 192.168.1.x subnet, launch the web browser and bring up the system login page by entering the following URL: http://192.168.1.100:9999. You will see the login interface as below:

Input username and password, press the "Login" button and you will see the   configuration interface below.

1.   Default username: admin and password: admin

 Notice

1.   Please use IE(7.0 or higher version), Chrome, Firefox web browser.

2.   If you do not see the interface above after inputting default IP and port number, please check whether your computer IP address is in the same segment with your IP PBX.

3.   For Security reasons, please modify the username and password after login successfully. You can modify these by selecting: 【System】--- 【Management】

4.   With the default setting, if there is no activity on the page for more than one minute, the system will timeout and automatically log out. To continue making configuration changes, you will need to login again.

2.   Network   WAN IP and MAC will be displayed

3.   Storage   Total storage and used storage will be displayed

4. Channels   Channel information will be displayed based on the modules installed

5. Device Info   Model No. And system version will be displayed

Commonly Used Buttons

On the home page, besides system info, there are other function buttons as below:

1. Logout   Logout the Web panel

2. Activate Changes   Activate the changes for your current configuration

System Menu

System Menu includes the following sub menu:
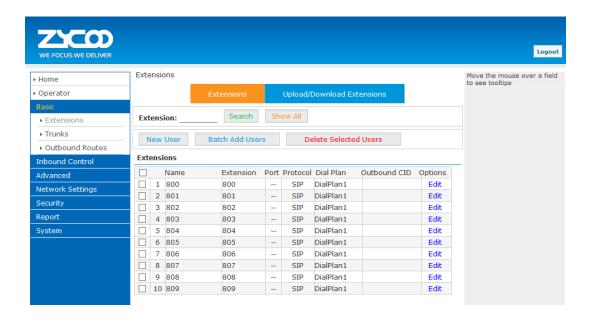
Home	Display device information
Operator	Extension / Trunk / Channel Status
Basic	Basic configuration on extension, trunks, etc.
Inbound Control	Configuration of Inbound Route, IVR and Black List, etc.
Advanced	Configuration of extension's default information, Conference Call, Call Transfer, Function Key, etc.
Network Settings	Configuration of Routing, Network, VPN, DHCP and other related network parameters
Security	Configuration of Firewall, SSH, FTP
Report	Record List, Call Logs and System Logs
System	Time Settings, Management, Back Up and Upgrade, etc.

2.2.2 Basic Configuration
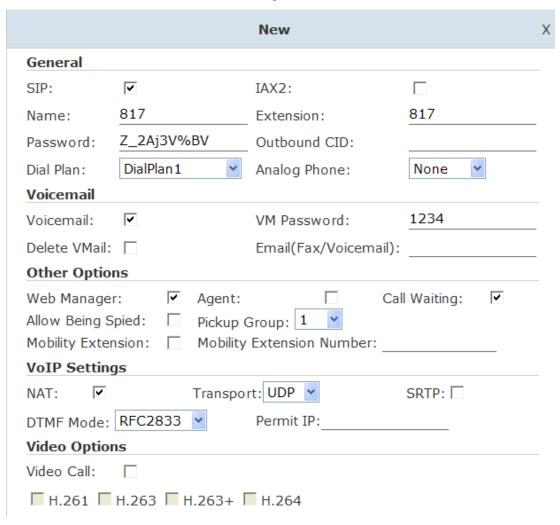
Extension Configuration

CooVox Supports SIP/ IAX2 and analog extensions as well as the ability to "Batch Add Users" by uploading extensions file.

Click【Basic】-> 【Extensions】to configure:

Click【New User】to see the extension configuration interface as below:

New		X

General

SIP:	☑	IAX2:	☐
Name:	817	Extension:	817
Password:	Z_2Aj3V%BV	Outbound CID:	
Dial Plan:	DialPlan1 ▾	Analog Phone:	None ▾

Voicemail

Voicemail:	☑	VM Password:	1234
Delete VMail:	☐	Email(Fax/Voicemail):	

Other Options

Web Manager:	☑	Agent:	☐	Call Waiting:	☑
Allow Being Spied:	☐	Pickup Group:	1 ▾		
Mobility Extension:	☐	Mobility Extension Number:			

VoIP Settings

NAT:	☑	Transport:	UDP ▾	SRTP: ☐
DTMF Mode:	RFC2833 ▾	Permit IP:		

Video Options

Video Call: ☐

☐ H.261 ☐ H.263 ☐ H.263+ ☐ H.264

Extension Settings:

Item	Explanation
SIP/IAX2	Choose extension protocol.
Name	Extension Name (English Character Only), e.g.: Tom.
Extension	Extension Number connected to the phone, e.g.: 888.
Password	Same password as voicemail. (4-16 digits, e.g.:123456)
Outbound CID	Override the caller ID when dialing out with a trunk.
Dial Plan	Please choose the Dial Plan which is defined in the menu "Outbound Routes".
Analog Phone	Please choose the relative FXS port for your analog phone.
Voicemail	Check this option to enable the voicemail account.
VM Password	Set password for Voicemail, for security reasons, do not use the extension number or any easy combination like "1234"
Delete VMail	Check this option to delete voicemail from the PBX after it's sent by email.
Email (FAX/Voicemail)	Extension user's email address to receive email messages with attached fax or voicemail (you need configure the fax to email/voicemail options), e.g.: Tom@gmail.com
Web Manager	Allow this user to login to the Extension Management Panel to manage extension options including voicemail, call recording, call transfer, etc when you select this option.
Agent	Check this option to set this extension user as agent.
Call Waiting	Enable call waiting
Allowing Being Spied	Check this option to allow this extension to be monitored (listened to or "spied").
NAT	Check this option if extension user or the phone is located outside the NAT(Network Address Translation) available gateway.
Pickup Group	Select the Pickup Group which the extension user belongs to.
Mobility Extension	After check this option, you must set mobility extension number. User can make calls to the IP PBX server with this mobility number, and have all rights of this extension, e.g.: Outbound Call, Internal Call, Listen to the voicemail.
Transport	Select the Transport Protocol: UDP, TCP, TLS
SRTP	Enable SRTP (Secure Real-time Transport Protocol)
DTMF Mode	Default DTMF is rfc2833. It can be changed if necessary..
Video Call	Check to enable video calling for this extension. And select the video codecs you need to use.
Permit IP	Set device ip address or subnet permitted to register this extension with the IP PBX, e.g.:192.168.1.77 or 192.168.10.0/255.255.255.0. Devices with other IP addresses are not allowed to register this extension with the IP PBX.
Audio Codecs	Select what audio codecs you need to use.

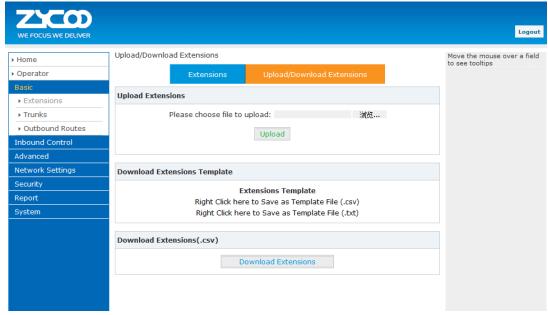⚠ **Notice:**

1. There are 30 default extensions which number started with "8"*; you can add or delete extension by your requirement.

2. Maximum extensions: 100

Upload/Download Extensions

Click【Upload/Download Extensions】to batch add extensions as below:

Download the extension template from the【Download Extensions Template】,

open the template using an editor or application like Microsoft Excel and carefully add extension information based on the template format and save.

Select the extension file to upload from【Upload Extensions】

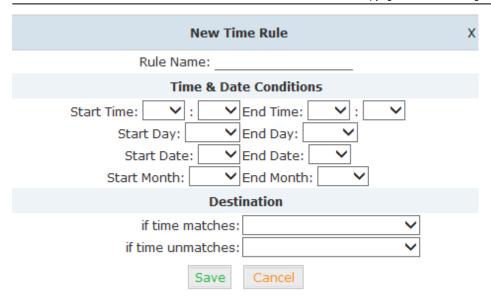Download current extensions information from【Download Extensions(.csv)】

2.2.3 Time Based Rules

Create a Time Rule. For example, BusinessHours.

Select the starting and ending time, starting and ending days of the week, specific start and end dates and/or start and ending month of the year.

When an inbound call is processed, if the current time of the PBX is within these parameters, then the "if time matches" destination will be used for the call.   If the current time of the PBX is outside these parameters, then the "if time does not match" destination will be used for the call.

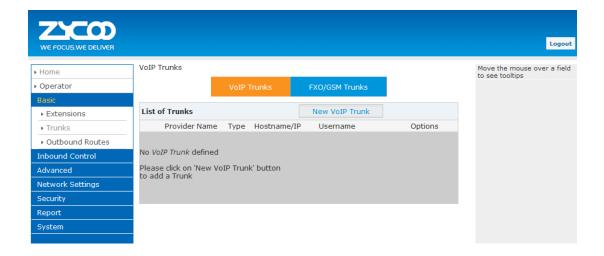Please set from this page:【Time Based Rule】---【New Time Rule】:

New Time Rule:

Item	Explanation
Rule Name	Define the name for this Time Rule.
Time&Date Conditions	Set parameters for Time/Day/ Date/ Month.
Destination	Select destination if time matches or does not match the conditions set. For example for BusinessHours, "if time matches", select operator extension during BusinessHours. If outside business hours, select "if time does not match" destination of Operator voicemail

2.3 Outbound Call

2.3.1 Trunks

If you want to set up outbound route connected to PSTN (Public Switch Telephone Network) or VoIP provider, please configure on this page:【Basic】-> 【Trunks】

CooVox supports two kinds of trunks for your choice: VoIP or SIP Trunk and FXO/GSM/PRI/BRI Trunk.

How to add each trunk:

VoIP Trunks

Click【VoIP Trunk】-> 【New VoIP Trunk】:
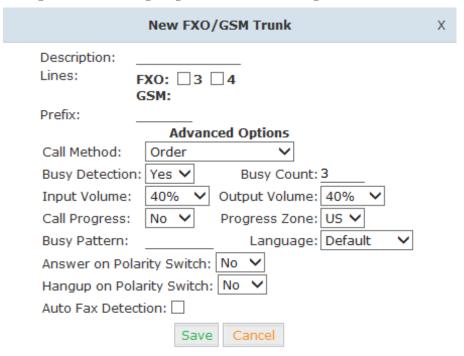
VoIP Trunks Reference:

Item	Explanation
Description	Description of SIP trunk.
Protocol	Select protocol for outbound route, SIP or IAX2.
Host	Set host address (provided by VoIP Provider).
Maximum Channels	Set maximum channels for simultaneous call. (Only for outbound call; "0" = no limitation).
Prefix	The prefix will be added in front of your dialed number automatically when the trunk is in use.
Caller ID	This Caller ID will be displayed when user make outbound call. Note: This function must be supported by local provider.
Without Authentification	If your trunk is static IP based and does not require a registration string when connecting the CooVox IP PBX, check this option.
Username	Username provided by VoIP Provider.
Password	Password provided by VoIP Provider.
Advanced Options	Advanced options for this trunk, e.g.: codecs, dialplan, etc.

The outbound trunk will be in the list of VoIP Trunk when the trunk is added successfully.

FXO/GSM Trunk

Click【FXO/GSM Trunk】-> 【New FXO/GSM Trunk】:

FXO/GSM Trunk Reference:

Item	Explanation
Description	Description for this trunk.
Lines	Check one or more channels (FXO or GSM) to be included in this trunk group
Prefix	The prefix will be added to the dialed number automatically when this trunk is in use.
Advanced Options	Advanced Options for this trunk, e.g.: Call Method, Busy Detection, etc.
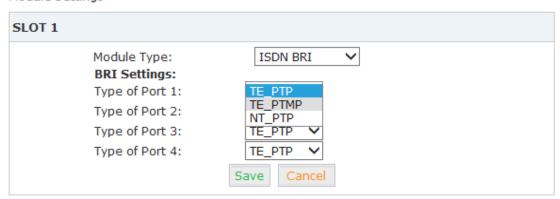
Select one or more of the available channels to be used for this trunk group.

Note: each channel can only be included in one trunk group. If no channels appear then all available channels are already defined.

BRI / PRI Trunk
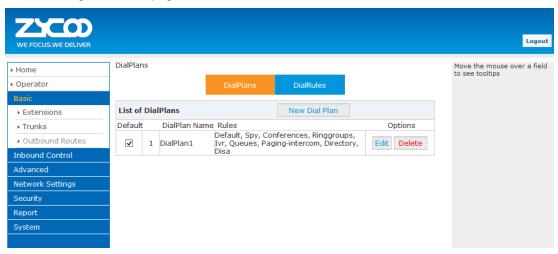
Please set up BRI/PRI trunk similarly to the FXO/GSM trunk settings above.

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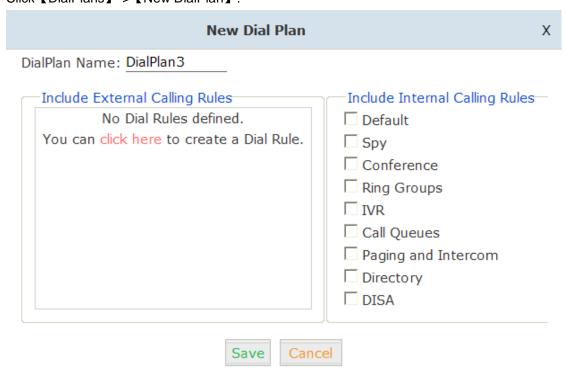
2.3.2 Outbound Routes

Outbound Routes are used to define which trunk groups are used by a specific extension when placing outbound calls. If you don't allow an extension user to place external calls, please ignore this part.

Please configure on this page: 【Basic】-> 【Outbound Routes】

You can configure the basic match pattern of outbound routes and create different dial plan on this page. Create as many different dial plans as you need to determine how you need extensions to be allowed to make calls. For example, create "InternalDialPlan" to include all Internal Calling Rules but do not select any outbound dial rules. Select "InternalDialPlan" for all extension users that do not need the ability to make external calls.

Click【DialPlans】-> 【New DialPlan】:

You can create one or more DialRules for DialPlans from this page:

New DialRule	X

Rule Name: _____

PIN Set: ☐

Call Duration Limit: _____ seconds

Time Rule: ☑

Start Time: 09 ⌄ : 00 ⌄ End Time: 17 ⌄ : 30 ⌄

Start Day: Mon ⌄ End Day: Fri ⌄

Place this call through:

test(FXO/GSM)
voip(SIP)

»»

→

←

««

Available Trunks   **Selected Trunks**

Custom Pattern: _____

Z Any digit from 1 to 9
N Any digit from 2 to 9
X Any digit from 0 to 9
. Any number of additional digits

Delete ____ digits prefix from the front and auto-add digit _____ before dialing

Save   Cancel

Reference:
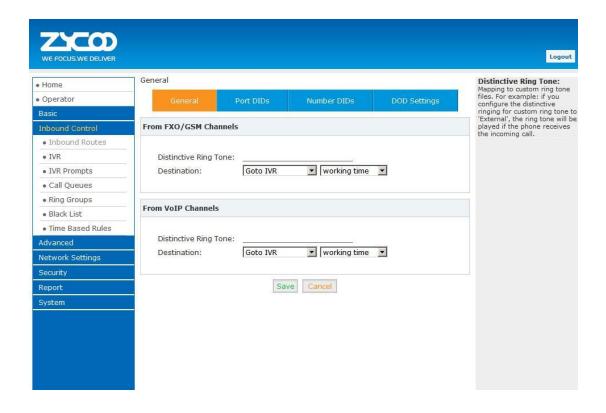
Item	Explanation
Rule Name	Define the name for the dial rule.
Pin Set	Input this Pin when you use this dial rule.
Call Duration Limit	Set the duration limit for a call, beyond which the call will be auto hung up
Time Rule	Set the time interval for this DialRule, beyond which the call based on this DialRule won't work
Place this call through	Select one of the trunk groups that have been set up to use for this dial rule
Custom Pattern	N any digit from 2 to 9
	Z any digit from 1 to 9

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	X	any digit from 0 to 9
	.	One or more digits
Delete[ ]digits prefix	How many digits will be deleted from what the user dialed to what is actually sent over the trunk. For example, user dialed 94166445775 and you selected to delete 1 digit, then 4166445775 is sent out the trunk.
Auto-add digit[ ]	If add digit "9", when dial 12345, 912345 will be sent.

2.4 Inbound Call

2.4.1 Inbound Routes

Click【Inbound Control】->【Inbound Routes】

General

Distinctive Ring Tone：mapping the custom ring tone file, e.g.: Set distinctive ring tone as "External", the phone will play this ring tone when receiving the call.
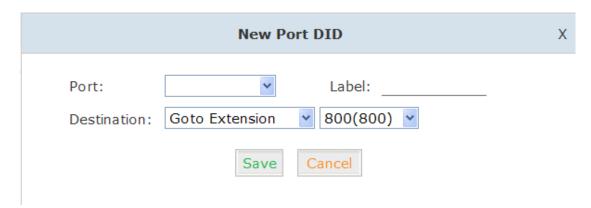
Note: The phone must support such feature as well.

Select all calls coming in on a specific port (FXO/GSM/VOIP) and select which destination (Extension User, IVR, Queue, Conference Bridge, IVR, etc) should answer those calls. Setting the label will assign this label to be displayed.

Port DIDs

To have incoming calls from a PSTN trunk port (FXO/GSM trunk) answered by a specific extension user, call queue, conference bridge, or IVR, please configure here:

Click【Port DIDs】->【New Port DIDs】:

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1.   Port   Select the trunk group port
2.   Label   Set a label for this port. Incoming calls from this port will display
                     the specified label.
3.   Destination   Incoming calls will be answered by the specified destination
                     (extension user, call queue, conference bridge, or IVR)

Number DIDs

If you want to select the destination of inbound calls on PRI/BRI or VoIP Trunks based on the incoming DNIS (dialed number or DID). You can specify the DID and destination (user extension, queue, conference bridge, or IVR:

Click 【 Number DID】->【New Number DID】:

4.   DID Number   Set DID Number
5.   Destination   Select the extension for access directly(Extension User/
                       Call Queue/ conference/ IVR)

DOD Settings

To configure outbound calls from user extensions to answer with specified destinations (user extension, queue, conference bridge, IVR) , please click 【DOD Settings】->【New DOD】

6.   DOD Number   Set the DOD (direct outbound dial) number, and use it to match
                          the Caller ID.
7.   Destination   Outbound calls will access directly to this destination
                          (user extension, queue, conference bridge, or IVR)
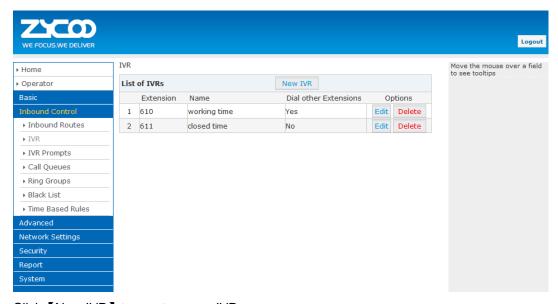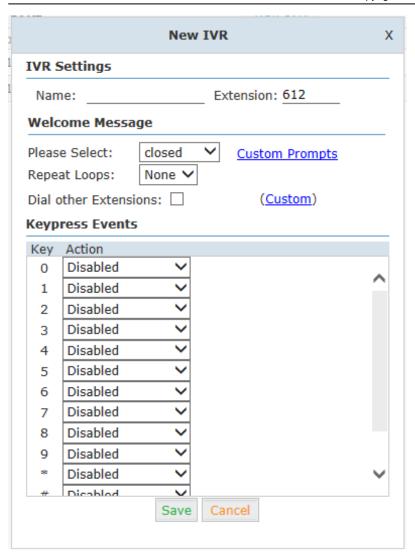
2.4.2 IVR

IVR (Interactive Voice Response) or Automated Attendant will allow callers to select from a specific set of options by pressing the selected digit on their telephone dial pad.
Click【Inbound Control】->【IVR】:

Click【New IVR】to create a new IVR:

New IVR   X

IVR Settings

Name: _____   Extension: 612 _____

Welcome Message

Please Select: | closed ∨ | Custom Prompts

Repeat Loops: | None ∨ |

Dial other Extensions: ☐   (Custom)

Keypress Events

Key	Action
0	Disabled ∨
1	Disabled ∨
2	Disabled ∨
3	Disabled ∨
4	Disabled ∨
5	Disabled ∨
6	Disabled ∨
7	Disabled ∨
8	Disabled ∨
9	Disabled ∨
*	Disabled ∨
#	Disabled ∨

Save   Cancel
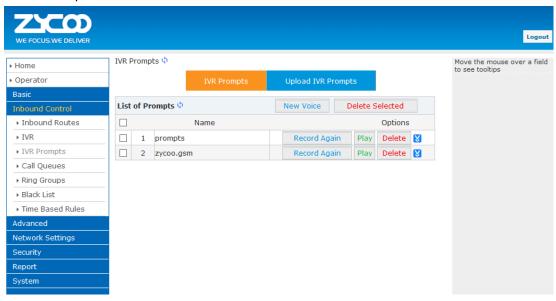
Item	Explanation
Name	Enter a descriptive name for the IVR
Extension	Enter a unique extension or IVR number. This number is used to access the IVR from an internal extension
Custom	Click "Custom" to choose a DialPlan for IVR
Please Select	Select the IVR prompt that will provide the caller with instructions on what options are available. To configure the prompt in this page:【IVR Prompt】
Repeat Loops	Loop times to repeat playing the IVR prompt if the caller does not select an option
Dial Other Extension	Allow user to dial other extensions besides of the listed options
Keypress Event	Select the available options beside the designated digit

2.4.3 IVR Prompts
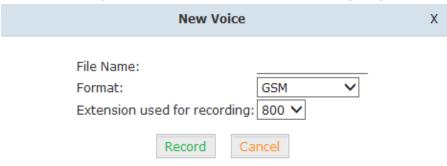
IVR prompts can be recorded by using any extension registered to the PBX or they can be uploaded from the "Upload IVR Prompt" section below.

IVR Prompts

【IVR Prompts】
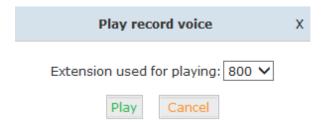
Click【IVR Prompts】----【New Voice】to create new IVR prompt:

1.   File Name   Define a name for this voice file.
2.   Format   Select the voice format,GSM/WAV(16bit) supported only.
3.   Extension used for recording: Select the extension which is used for recording the
                                       IVR prompt.

Click【Record】, the extension will ring, and the prompt can be recorded after picking up the phone.
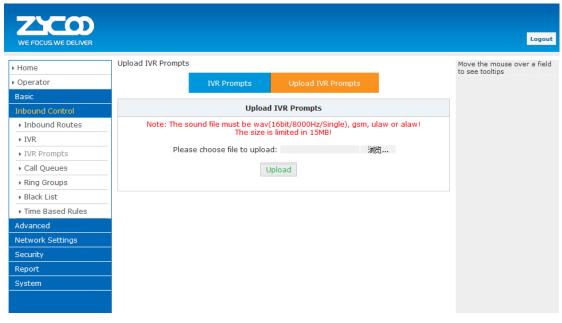
To hear the existing recording, please click【Play】:

Select the extension, click【Play】, the selected extension will ring, and you will hear the recorded prompt after picking up the phone.

Upload IVR prompt

【Upload IVR prompt】

 Notice:
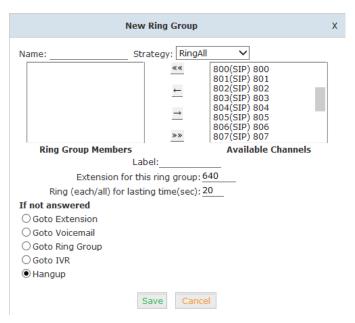
CooVox supports custom audio file with wav,gsm,ulaw,alaw format. Recordings must be smaller than 15MB.

2.4.4 Ring Groups

A Ring Group (sometimes called a Hunt Group) is a way to ring a collection of extensions by dialing a single extension number.   The methodology used to ring that collection of extensions is called the ring strategy.   Once the timeout (number of seconds) is reached, the call will then be directed to the "if not answered" or failover destination.

To configure a Ring Group   Click【Inbound Control】->【Ring Groups】->【New Ring Group】:

1. Name — Define a name for the Ring Group

2. Strategy — Select "Ring All" or "Ring in order"

3. Ring Group Members — Select the Ring Group Member from "the Available Channels", click [←] to add.
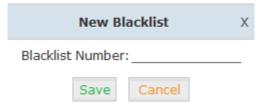
4. If not answered — You can choose to forward the call to extension, voicemail ring group, IVR or hang up if not answered.

2.4.5 Blacklist
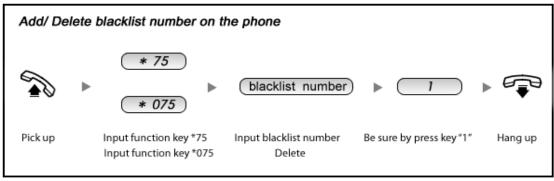
The Blacklist feature allows the blocking of specific phone numbers by Callerid.

Click【Inbound Control】->【Blacklist】->【New Blacklist】

Input the caller ID in the space provided.   Once configured, future calls from this caller ID will be blocked.

To maintain this list of blocked numbers, see the instructions in the following diagram:

Reference:
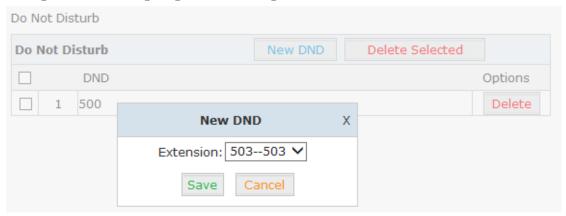
Item	Explanation
*75	When the registered extension user inputs *75 + blacklist number, this number will be added in the list of Blacklist Number.
*075	When the registered extension user inputs *075+blacklist number, this number will be deleted in the list of Blacklist Number.

2.4.6 Do Not Disturb
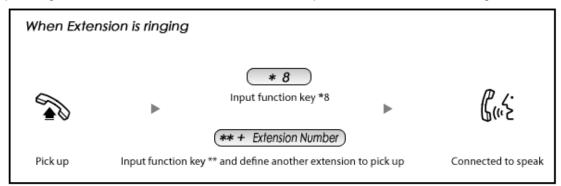
The administrator can config DND for extensions on this page:

Click【Inbound Control】->【Do Not Disturb】:

2.4.7 Call Pickup

This feature allows users to answer a call that is ringing on another users extension by pressing the selected feature code on their own phone as shown in the diagram below.

Reference:
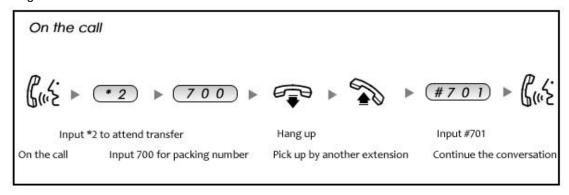
Item	Explanation
*8	Input function key *8 to pick up the registered extension which is in the ring at random. This can be defined in 【Feature Codes】
**	Input function key ** and define another extension to pick up. This can be defined in 【Feature Codes】.

2.5 During a Call

2.5.1 Call Parking

This feature allows a call to be placed on hold (system will play the parked number, e.g. 701) and then retrieved from any other extension by entering the parked number.   After answering the call, to park the call press *2 700 on the telephone dialpad (to transfer the call to the parking lot 700).   This will park the call and the system will play the parking space (e.g. 701). To retrieve the call from the parking lot, anyone can pick up any registered extension and dial the parking space number (e.g. 701) and will be connected with the parked caller. Refer to the diagram below:

Reference:
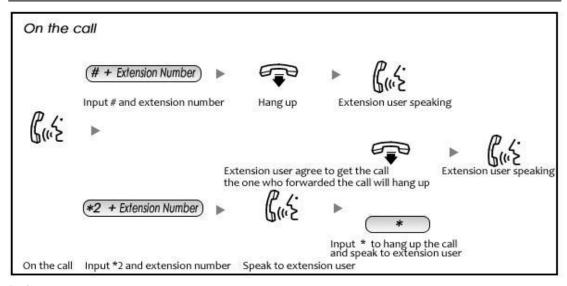
Item	Explanation
Extension to Dial for Parking Calls:	Default Number: 700, Define in【Feature Codes】
What Parking space or Extension to park calls on	Default Number : 701 - 720. Define in【Feature Codes】
How many seconds a call can be parked for	Default is 45 seconds. Define in 【Feature Codes】.

2.5.2 Call Transfer

This feature allows an incoming call that is answered on one extension to be sent to another user's extension.   Refer to the diagram as below:

Reference:
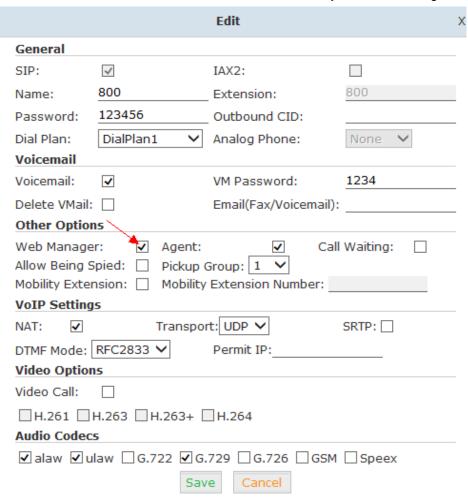
Item	Explanation
Blind Transfer	Default is #. Define in 【Feature Codes】
Attended Transfer	Default is *2. Define in【Feature Codes】
Complete Attended Transfer	Default is *, it can be used when you use *2. Define in 【Feature Code】
Timeout for answer on attended transfer	Default is 15 seconds. Define in【Feature Codes】

2.6 User Extension Settings

2.6.1 Follow Me Settings

This feature allows a call to an extension to be automatically forward to one or more internal extensions or external phone numbers.   To allow the user to configure these settings, first the user must be allowed access to the User Web Portal.   To do this, select the "Web Manager" box under "Other Options".

Click【Basic】-> 【Extension】-> 【Edit】the extension you want to configure.
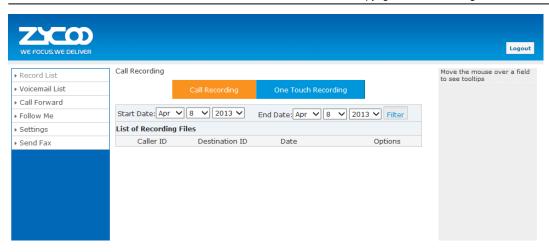
Check【Web Manager】 and【Save】

Then login the Extension Web Panel:

2.6.2 Call Recording
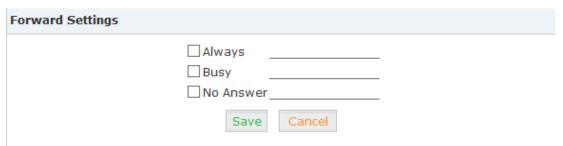
This feature allows users to access calls they have recorded.   To configure this setting, please see the diagram below.

2.6.3 Call Forward

This feature allows calls to an extension to be automatically forwarded to a specific internal extensions or external phone number. To configure this setting, please see below:

Click【Call Forward】:

Reference
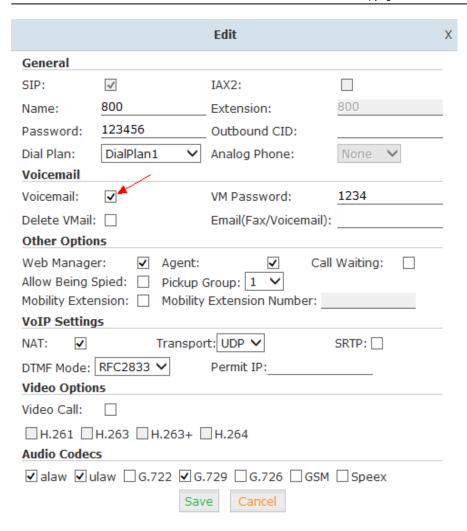
Item		Explanation
Status	Always	All incoming calls will be forwarded.
	Busy	Forward when extension is busy.
	No Answer	Forward when no answer from extension.

2.6.4 Voicemail
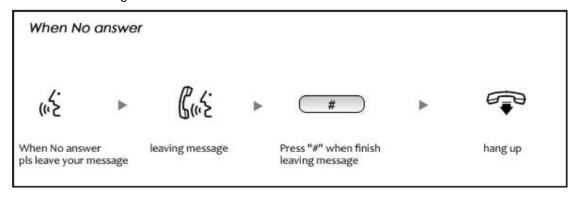
Calls that are not answered have the option to be sent to a voicemail account so the caller can leave a recorded message. Optionally, these recorded messages may be sent to a user's email account.

Click【Basic】-> 【Extension】-> 【Edit】the extension you want to configure.
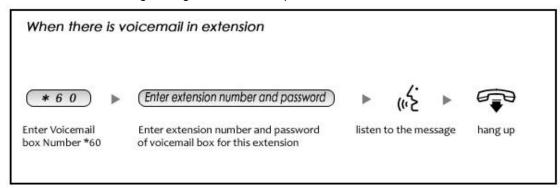
Please enable【Voicemail】before configuration, and configure【VM Password】and【Email】.
If incoming calls are not answered, when the default ring time is over, the system will play:
"please leave your message and press the "#"key ". Then voicemail will be sent to the
specified mailbox by email.

To Leave a Message

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To Listen to the message using the users desk phone

 Notice:

1.   Proper Email address is necessary to receive voicemail via email.

2.   You must configure the SMTP and Email template. For detail settings, please see the detail configuration guide【Voicemail】in Chapter 3.
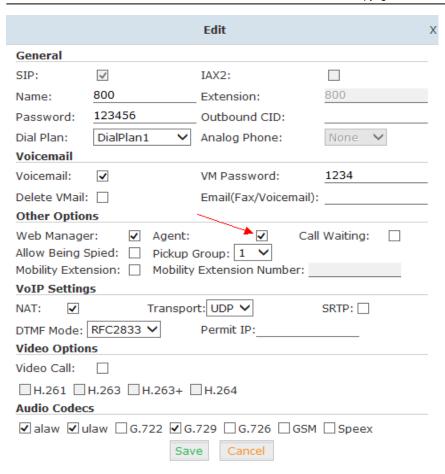
2.7 Call Center (Call Queues)

Create Agent
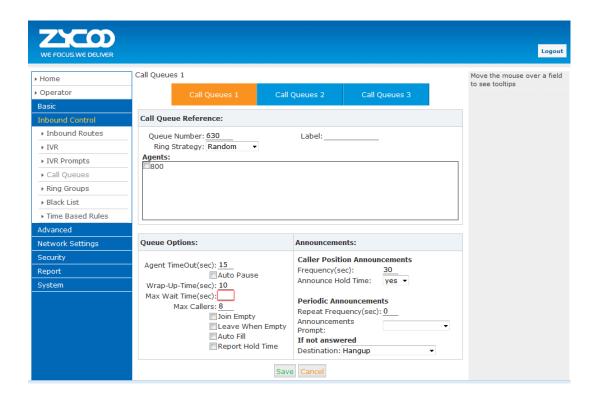
To allow a user to be considered an agent in a Call Center queue, please check the "Agent" option for that specific user extension.

Click【Basic】->【Extension】->【Edit】the extension you want to configure:

Step1: Check【Agent】and【Save】
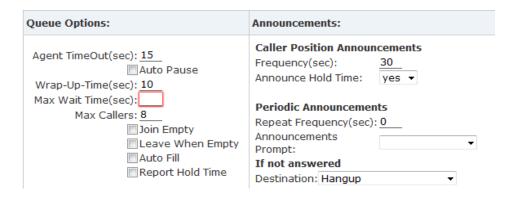
Step2: Click【Inbound Control】->【Call Queues】

Reference

Item	Explanation
Queue Number	Define an extension number to identify the queue.
Label	Define the label for the queue.
Ring Strategy	RingAll--Ring all available agents until one answers(default)
	RoundRobin – Starting with the first agent, ring the extension of each agent in turn until the call is answered.
	LeastRecent – ring the extension of the Agent who has least recently received a call
	FewestCalls – ring the extension of the Agent who has taken the fewest number of calls.
	Random – ring the extension of a random Agent.
	RRmemory -- RoundRobin with Memory, like RoundRobin above, except instead of the next call starting with the first agent, the system remembers which extension was was called last and begins the round robin with the next agent .
Agent	Check each agent that is to be a member of this specific Call Center Queue.

Queue Options:

Agent TimeOut(sec): 15
    ☐ Auto Pause
Wrap-Up-Time(sec): 10
Max Wait Time(sec):
        Max Callers: 8
    ☐ Join Empty
    ☐ Leave When Empty
    ☐ Auto Fill
    ☐ Report Hold Time

Announcements:

Caller Position Announcements
Frequency(sec):   30
Announce Hold Time: yes ▼

Periodic Announcements
Repeat Frequency(sec): 0
Announcements
Prompt: ▼
If not answered
Destination: Hangup   ▼

Reference:

Item	Explanation
Agent TimeOut(sec)	Specify the number of seconds to rin an agent's extension before sending the call to the next Agent (based on Ring Strategy).
Auto Pause	If an Agent's extension rings and the Agent fails to answer the call, automatically pause that agent so the stop receiving calls from the queue.
Wrap-Up-Time(sec)	This is the amount of time in seconds that an agent has to complete work on a call after the call is disconnected. (Default is 0, which means no wrap-up time.)
Max Wait Time(sec)	Calls that have been waiting in the queue for this number of seconds will be sent to the ""If not answered" destination.
Max Callers	Max number of the callers who are allowed to wait in the queue. (Default is 0, which means no limitation.). With this number of callers in the queue already, subsequent callers will be sent to the ""If not answered" destination.
Join Empty	Allow callers to enter the Queue when no Agents are available. If this option is not defined, callers will not be able to enter Queues with no available agents - callers will be sent to the "If not answered" destination.
Leave When Empty	If this option is selected and calls are still in the queue when the last agent logs out, the remaining callers in the Queue will be transferred to "If not answered" destination. This option cannot be used with Join Empty simultaneously.
Auto Fill	Callers will be distributed to Agent automatically.
Report Hold Time	Report the hold time of the next caller for Agent when the Agent is answering the call.
Frequency(sec)	Repeat frequency to announce the hold time for callers in the Queue.("0" means no announcement).
Announce Hold Time	Announce the hold time. Announce (yes), do not announce (no) or announce once (once), it will not be announced when the hold time is less than 1 minute.
Repeat Frequency(sec)	Interval time to play the voice menu for callers.("0" mean not to play).
Announcement Prompt	Select a prompt as the Announcements Prompt from the IVR Prompts.

2.8 Conference Bridge

A conference bridge is a virtual meeting room that allows multiple callers to hear and speak to each other. The conference bridge can be protected with a password so only callers with the password can access the conference. The software supports up to three conference rooms. To configure a conference bridge, go to 【Advanced】->【Conference】:
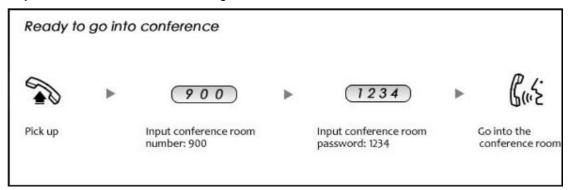
Conference(Default)

Conference(Default)	Conference 2	Conference 3

Conference Number

Room Extension: 900

Conference Password

Guest Password: 1234
Administrator Password: 2345

Conference Options

Conference DialPlan [DialPlan1 ▼]
☑ Play hold music for first caller
☑ Enable caller menu
☐ Announce callers
☐ Record conference
☐ Quiet Mode
☐ Leader Wait

Save   Cancel

Reference:
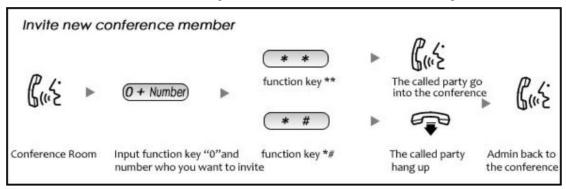
Item	Explanation
Conference Number	The number that internal callers use to access the conference room, the default number is "900".
Conference Password	Password for users to access the conference, e.g.:"1234".
Administrator Password	Password for administrator to access the conference.
Conference DialPlan	Use this dialplan to invite other participants.
Play hold music for the first participant	Check this option to play the hold music for the first participant in the conference until another participant enters in this conference.
Enable caller menu	Check this option to allow the participant to access the Conference Bridge menu by pressing "*" on the dialpad.
Announce callers	Check this option to announce to all Bridge participants that new participant is joining the conference.
Record conference	Recorded conference format is WAV.
Quiet Mode	If check this option, all the participants in the conference can hear only, but it is not allowed to speak.
Leader Wait	Wait until the conference leader(administrator) entering the conference before starting the conference.

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To join a conference, refer to the diagram as below:

While in a conference, the administrator can invite new guest (extension user or external number) into the conference. (Default password for admin is 2345)

As an administrator, to invite a new guest to the conference, refer to the diagram as below:
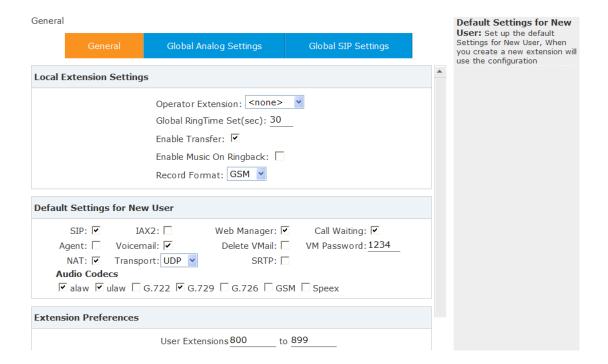
Chapter 3   Advanced

3.1 Options

General

Default settings for local extension and new extension.

Click【Advanced】-> 【Options】-> 【General】:

Reference
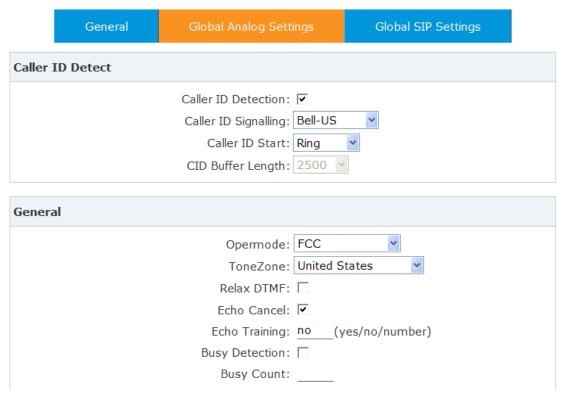
Item	Explanation
Operator Extension	Set extension number for Operator.
Global RingTime Set	Set RingTime for every extension.
Enable Transfer	Check to enable Transfer.
Enable Music On Ringback	Check to enable Music On Ringback.
Record Format	Set the format for recording files. (GSM/WAV only)
Defaut Setting for New User	Check to enable the default settings.
Extension Preferences	Set the rule for extensions.

Global Analog Settings

Click【Advance】-> 【Options】-> 【Global Analog Settings】:

Global Analog Settings

| General | Global Analog Settings | Global SIP Settings |

Caller ID Detect

Caller ID Detection: ☑

Caller ID Signalling: Bell-US

Caller ID Start: Ring

CID Buffer Length: 2500

General

Opermode: FCC

ToneZone: United States

Relax DTMF: ☐

Echo Cancel: ☑

Echo Training: no ___ (yes/no/number)

Busy Detection: ☐

Busy Count: _____

Reference:

Item	Explanation
Caller ID Detection	Enable/Disable Caller ID Detection
Caller ID Signaling	Select the mode of Caller ID Signaling.
Caller ID Start	Ring--Caller ID start before ring.
	Polarity--Caller ID start when polarity reversal starts.
CID Buffer Length	Default CID Buffer Length
Opermode	Set the Opermode for FXO/GSM Ports.
ToneZone	Select the ToneZone in your country.
Relax DTMF	Enable/Disable Relax DTMF inspection.
Echo Cancel	Enable/Disable Echo Cancel
Echo Training	Set Echo Training (default unit: ms)
Busy Detection	Enable/Disable Busy Detection.
Busy Count	Count the Busy Detection. It will be active when enable Busy Detection.

Global SIP Settings

【Global SIP Settings】is appropriate for advanced administrators. Please contact our technical support department before modifying anything in this section.

3.2 Voicemail

Click【Advanced】-> 【Voicemail】-> 【General】:

General
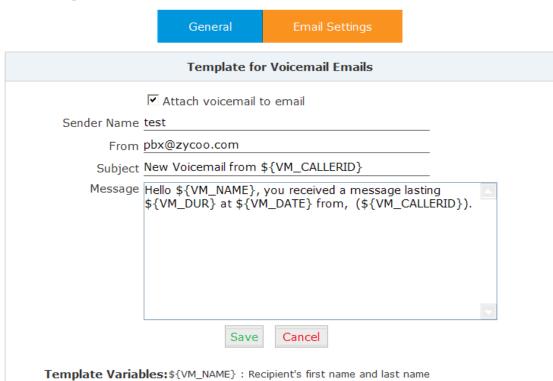
General	Email Settings

VoiceMail Reference

Max Greeting Time(sec):   30

Dial "0" for Operator: ☑

Voice Message Options

Message Format: WAV (16-bit)

Maximum Messages: 100

Max Message Time(min):   2

Min Message Time(sec):   5

Playback Options

☑ Say Message CallerID

☑ Say Message Duration

☐ Play Envelope

☐ Allow Users to Review

Reference

Item	Explanation
MaxGreeting Time(sec)	Maximum recording length for voicemail greetings
Dial "0" for Operator	Select this option to allow callers to press Dial "0" to transfer out of voicemail to the Operator.
Message Format	Save the voice message as this format, WAV(16-bit) or Raw GSM.
Maximum Messages	Maximum voicemail messages to be allowed to leave.
Max Message Time(min)	Maximum Time for each message to be allowed to leave.
Min Message Time(sec)	MinimumTime for each message. The message will be deleted automatically if the time is less than the min message time.
Say Message CallerID	Play the Caller ID of the caller before playing the voice message.
Say Message Duration	Play the message duration before playing the voice message.
Play Envelope	Play the date, time and caller ID for the voicemail message.
Allow Users to Review	Check this option to allow users to review the voice message.

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Click【Advance】-> 【Voicemail】-> 【Email Settings】

Email Settings

| General | Email Settings |

Template for Voicemail Emails

☑ Attach voicemail to email

Sender Name test

From pbx@zycoo.com

Subject New Voicemail from ${VM_CALLERID}

Message Hello ${VM_NAME}, you received a message lasting ${VM_DUR} at ${VM_DATE} from, (${VM_CALLERID}).

Save   Cancel

Template Variables: ${VM_NAME} : Recipient's first name and last name
${VM_DUR} : The duration of the voicemail message
${VM_MAILBOX} : The recipient's extension
${VM_CALLERID} : The Caller ID of the person who left the message

Reference:

Item	Explanation
Attach voicemail to Email	The voicemail will be sent as attachment to the user's Email.
Sender Name	The sender's name will be displayed when you receive the Email.
From	Mailbox to send email
Subject	Subject of the Email.
Message	Input the Email template.

3.3 SMTP Settings

To allow email messages to be sent to users with attached voicemail and faxmail messages, the SMTP settings need to be configured.
Click【Advance】-> 【SMTP Settings】:

SMTP Settings
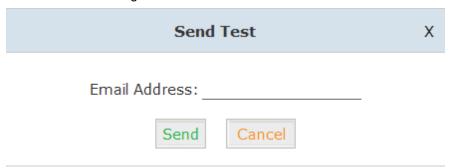
SMTP Settings:

SMTP Server: _____

Port: 25 _____

SSL/TLS: ☐

☑ Enable SMTP Authentication

Username: _____

Password: _____

Send Test

Save   Cancel

Reference

Item	Explanation
SMTP Server	You must set SMTP Server address or domain connected to the CooVox IP PBX, which is used for sending the voice message to Email.
Port	Port number for SMTP server. Default is 25, and it will be changed to 465 when you enable SSL/TLS.
SSL/TSL	Enable SSL/TLS.
Enable SMTP Authentication	If your SMTP server needs authentication, please enable this option, and configure the following.
Username	Input username of your Email.
Password	Input password of your Email.
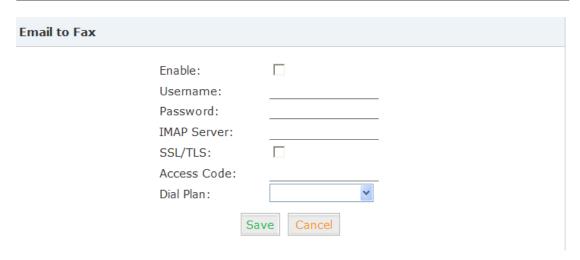
Click【Send Test】after configuration, the following diagram will be displayed to ask you to input the Email for receiving.

Send Test   X

Email Address: _____

Send   Cancel

Specify the email address and click【Send】-to send the test email. Verify that email was successfully sent or not. If no email was received, please modify the SMTP settings and retry.

3.4 Email to Fax

Users can send fax by Email. Please configure as below.
Click【Advanced】-> 【Email to Fax】

Email to Fax

Enable: ☐
Username: _____
Password: _____
IMAP Server: _____
SSL/TLS: ☐
Access Code: _____
Dial Plan: [▼]

[Save] [Cancel]

Check "Enable", input username, password and IMAP Server(server format: imap.XX.com), select the DialPlan, then "Save" and "Activate".

Practical Case:
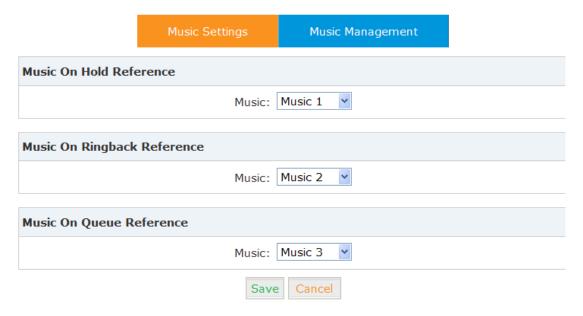
To Send a fax to telephone number 85337096: In DialPlan 1, there is prefix "9" before the telephone number; you need input the 【Access Code】: 985337096 and make this the subject when sending Email. Then the fax will be sent by Email as attachment.

If you need dial the extension when sending fax, e.g.: fax number: 85337096 ext.800, you need use the 【Access Code】: 985337096-800 as subject.

3.5 Music Settings
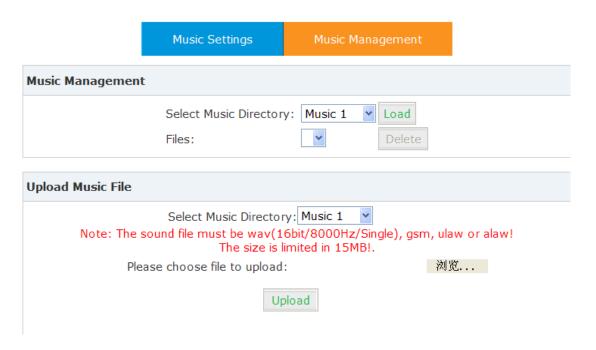
Management of Music on Hold, Music on Ringback, Music on Queue.
【Music Settings】:

| Music Settings | Music Management |

Music On Hold Reference

Music: [Music 1 ▼]

Music On Ringback Reference

Music: [Music 2 ▼]

Music On Queue Reference

Music: [Music 3 ▼]

[Save] [Cancel]

Select the different music file for different Music.
【Music Management】
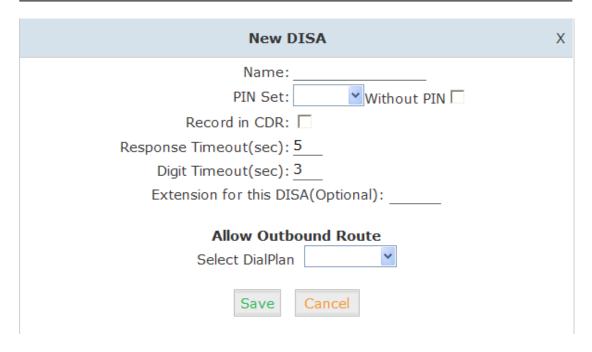
Item	Explanation
Select Music Directory	Select which Music Directory you wish to load.
File	Display music name under the music file, you can delete it.
Select Music Directory	Select the file where you want to save your uploaded music.
Please choose file to upload	Select the music you want to upload. Note: music file must be WAV(16bit/8000Hz/Single), GSM, ulaw or alaw, and less than 15MB.

3.6 DISA

This feature allows an authorized user to call into the PBX and then place an outbound call using another trunk. For example, an employee working out of the office who needs to make an international call using trunks connected to the PBX. By calling the DISA number, after PIN authentication, the caller hears dial tone and can dial the call.

Please configure as below.
Click【Advance】-> 【DISA】--【New DISA】

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Reference
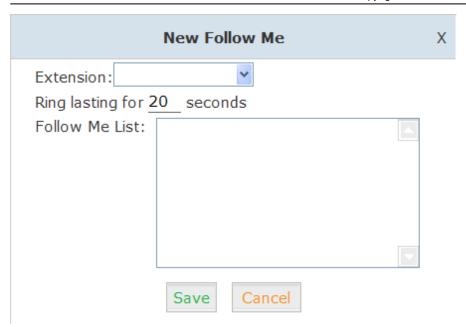
Item	Explanation
Name	Define a name for DISA.
PIN Set	User will be prompted to input this number when PIN Authentication is needed.
Record in CDR	Check to record.
Response Timeout(sec)	The maximum time for waiting before hanging up if the dialed number is incomplete or invalid. Default is 10 seconds
Digit Timeout(sec)	The maximum interval time between digits when typing extension number. Default is 5 seconds.
Extension for this DISA(Optional)	If you want to access DISA by dialing an extension, you can define an extension number for this DISA.
Select DialPlan	Select the DialPlan for this DISA.

3.7 Follow Me

This feature allows callers to automatically be forwarded to one or more internal extensions and/or one or more external phone numbers when the call is not answered at the primary extension.

Please configure as below:
Click【Advanced】-> 【Follow Me】-> 【New Follow Me】:

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New Follow Me   X

Extension: [▼]

Ring lasting for 20   seconds

Follow Me List: [   ]

[Save]   [Cancel]

Select an extension, set the ring duration, and add the numbers in the Follow Me List; 【Save】
and 【Activate】.

List Format: Extension Number, Ring Duration

E.g.:   806,30

    808,20

806 rings, after 30 seconds, the call is going to 808
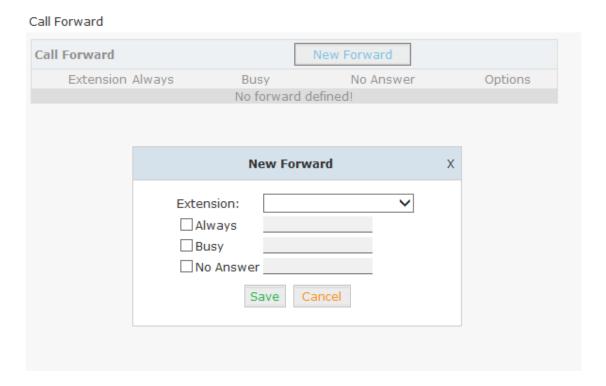
【Follow Me Options】

Follow Me Options

☐ Playback the incoming status message prior to starting the follow-me step(sec).

☐ Record the caller's name so it can be announced to the callee on each step.

☐ Playback the unreachable status message if we've run out of all steps or the callee was set
not to be reachable.

[Save]

3.8 Call Forward

The administrator can config the Call Forward on this page:
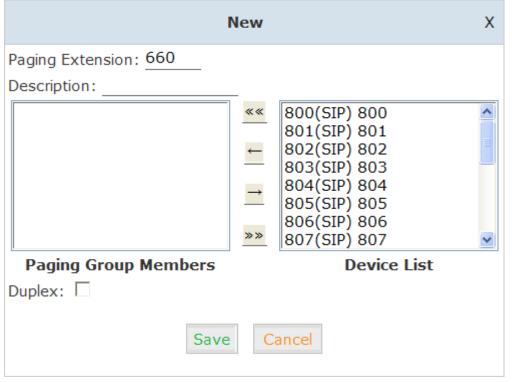
Click 【Advanced】-> 【Call Forward】:

Call Forward

Call Forward		New Forward		
Extension Always	Busy	No Answer		Options
		No forward defined!		

New Forward   X

Extension: [dropdown]
☐ Always _____
☐ Busy _____
☐ No Answer _____

Save Cancel

3.9 Paging and Intercom

This feature allows setting up a Paging group so when the Paging extension is dialed, the listed extensions allow the caller to speak through the speaker phone. The extensions in the Paging group must use phones that support this feature. If the Duplex option is selected, and the listed extensions use phones that support Duplex, then all the phones in the paging group will be able to have two-way conversations.

Click【Advanced】-> 【Paging and Intercom】-> 【New Paging Group】:

New   X

Paging Extension: 660
Description: _____

	800(SIP) 800
««	801(SIP) 801
←	802(SIP) 802
	803(SIP) 803
→	804(SIP) 804
	805(SIP) 805
»»	806(SIP) 806
	807(SIP) 807

Paging Group Members   **Device List**

Duplex: ☐

Save Cancel

Reference:
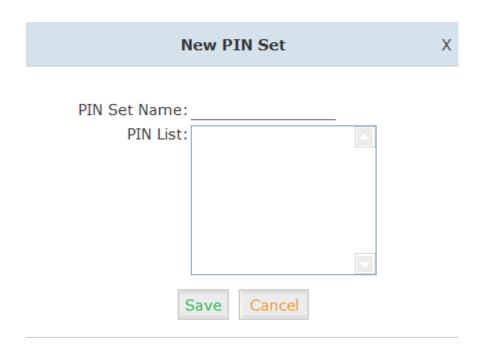
Item	Explanation
Paging Extension	Define an extension for this Paging Group.
Description	Define a name for this Paging Group.
Paging Group Members	Selected devices in this Paging Group.
Device List	Select device(s) here to Paging Group.
Duplex	Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it look like an "instant conference".

3.10 PIN Sets

This feature allows an administrator to specify a list of PIN codes in a PIN Set.   An Outbound Route can be specified that a valid PIN code from a selected PIN Set must be used in order to have access to a give Outbound route (e.g. for long distance or international calling).

Please configure as below.
Click【Advanced】-> 【PIN Sets】-> 【New PIN Set】:

1.   PIN Set Name   Define the name for this PIN Set.
2.   PIN List   Define PIN codes in this list.

3.11 Call Recording

This feature allows an administrator to enable Call Recording to record incoming and/or

outgoing calls related to the specified extension.

Please configure as below:
Click【Advanced】->【Call Recording】->【New Call Recording】:

Reference:

Item	Explanation
Extension	Define an extension for recording.
Call Recording Time	Set the time to record.
Inbound Record	Check to record inbound calls.
Outbound Record	Check to record outbound calls.

3.12 Speed Dial

This feature allows setting up system wide speed dial numbers that translate a feature code (*99) plus a two-digit code (00-99) into an external phone number.

Please configure as below.
Click【Advanced】->【Speed Dial】->【New Speed Dial】:

New Speed Dial   X

Notice:Don't forget to add the outbound dial prefix if you would like
to dial an outside number
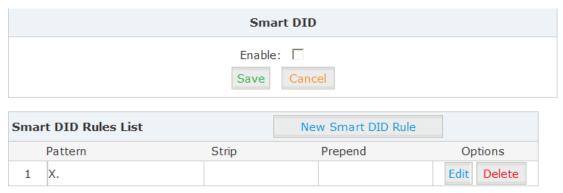
Source Number: _____

Destination Number: _____

Save   Cancel

E.g.: prefix is *99 , speed number is 00, destination telephone number is 85337096.
When dial *9900, the call is going to 85337096 automatically.

3.13 Smart DID

Smart DID: After extension user makes an outbound call, the call is ringing back to CooVox IP
PBX, and directed to the extension who made the last call. Please configure as below.

Click【Advanced】->【Smart DID】:

Smart DID

Enable: ☐
Save   Cancel

Smart DID Rules List		New Smart DID Rule		
Pattern	Strip	Prepend		Options
1 X.				Edit Delete

Check "Enable" and "Save" to make this function activate.

Click【New Smart DID Rule】to display the following diagram:

New Smart DID Rule   X

Pattern: _____

Strip: ____digits before dialing

Prepend: ____before dialing

Save   Cancel

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Input the pattern and define how many digits need to be stripped or prepend, then click "Save"--"Activate".

3.14 Callback

This feature allows an external caller to place an inbound call to the CooVpx IP PBX.   The inbound call will be disconnected and subsequently the PBX will place an outbound call back to this number and forwarded to defined destination after the call is connected.

Please configure as below.

Click【Advanced】-> 【Callback】:

Callback Number Settings

Enable: ☐
Strip: _____ digits before dialing
Prepend: _____ before dialing
DialPlan: _____ ▼
Save   Cancel

List of Callback Number	New Callback Number	
Callback Number	Destination	Options
No Callback Number defined!		

Enable this function; select DialPlan, and define the callback rule (strip digits or prepend prefix).

Click【New Callback Number】to add callback number.

New Callback Number   X

Callback Number: _____
Destination: Goto Extension ▼ 800(800) ▼
Save   Cancel
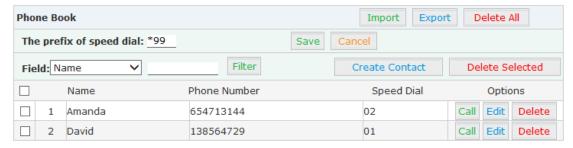
Input callback number and define the destination.

3.15 Phone Book

When incoming call Caller ID matches the number in the phone book, the name of matched number will be displayed. Please configure as below.

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Click【Advanced】->【Phone Book】:

Reference:

Item	Explanation
Import & Export	Import & Export a list, make sure it's UTF-8 coded if it's not in English
Delete All	Delete all the contacts from the phone book
The prefix of speed dial	Set the prefix of speed dial
Filter	Search contacts, by name, phone number or speed dial
Create Contact	Create a contact
Delete Contact	Delete a selected contact
Call	Click to call the number directly

Click【Create Contact】to see the following diagram:

3.16 Feature Codes

Click【Advanced】->【Feature Codes】to see the following diagram, and you can define the code for each feature.

Feature Codes Management

Call Parking

Extension to Dial for Parking Calls: 700

Extension Range to Park Calls: 701-720

Call Parking Time(sec): 45

Parking Hints: ☐

Pickup Call

Pickup Extension: *8

Pickup Specified Extension: **

Transfer

Blind Transfer: #

Attended Transfer: *2

Disconnect Call: *

Timeout for answer on attended transfer(sec): 15

One Touch Recording

One Touch Recording: *1

Call Forward

Enable Forward All Calls: *71

Disable Forward All Calls: *071

Enable Forward on Busy: *72

Disable Forward on Busy: *072

Enable Forward on No Answer: *73

Disable Forward on No Answer: *073

Reference:

Item	Explanation
Extension to Dial for Parking Calls	Define an extension for parking calls.
Extension Range to Park Calls	Define the extension range for parking calls. (e.g.: 701-720)
Call Parking Time(sec)	Define the time for parking calls. CooVox IP PBX will return the call to the extension after this time limit has expired.
Pickup Extension	This feature code will pick up a call given that the callers extension and the ringing extension are in the same pickup group and call group.
Pickup Specified Extension	This feature code allows a caller to Pickup a call ringing on the specified extension. Default: Dial**+extension number to pickup the specified extension.
Blind Transfer	To Allow unattended or blind transfer while on a call based on the following steps: 1. While on a call with caller "A", the user dials the blind transfer key sequence (in this case "#"). The system places the original call with "A" on hold, says "Transfer" then gives a dial tone. 2. dial the transferee extension or phone number you wish to transfer the call to "B" and hangup the phone. 3. The original caller "A" is transferred immediately to the transferee "B" and "B" sees the callerid of "A".
Attended Transfer	To Allow attended or supervised transfer while on a call based on the following steps: 1. While on a call with caller "A", the user dials the supervised transfer key sequence (in this case "*2"). The system places the original call with "A" on

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	hold, says "Transfer" then gives a dial tone.
	2. dial the transferee extension or phone number you wish to transfer the call to "B" and wait for "B" to answer the phone and talk to "B" to introduce the call.
	1. If "B" does not wish to take the call, "B" can hang up the call and you are returned to your call with "A".
	2. If "B" wishes to accept the call, you hang up the phone and caller "A" is transferred to the transferee "B".
	3. If the call goes to voicemail or you wish to abort the transfer, simply press the "disconnect call" key sequence (in this case "*") and the transfer will be aborted and you will be back on the call with the original caller "A".
Disconnect Call	Disconnect the current transfer call (for Attended transfer).
Timeout for answer on attended transfer（sec）	Set the timeout value
One Touch Recording	Configure the function key for One Touch Recording
Call Forward	Enable/Disable Call Forward and the settings of function keys for different forward modes.
Do Not Disturb	Enable/Disable "Do Not Disturb"
Spy	Configure the function keys for spy modes.
Blacklist	Add/Delete blacklist number.
Voicemail	Configure the function keys for entering voicemail and check extension voicemail.
Invite Participant	In conference, the administrator can invite people into the conference by dialing "0". After pressing "0", you will get dialtone, and you can dial to invite people. After the call is connected, please press ** to direct the people into the conference, or *# to hang up the current call and return to the conference.
Create Conference	During the call, you can dial *0 to forward to the conference with the callee.
Return to conference with participant	In conference, the administrator can dial "0" to invite people into the conference. After pressing "0", you will get dialtone, and you can dial to invite the participant; when the call is connected, dial "**" to return to the conference with invited participant.
Return to conference without participant	In conference, the administrator can dial "0" to invite people into the conference. After pressing "0", you will get dialtone, and you can dial to invite the participant. When the call is connected, you can dial "*#" to hang up and return the conference yourself.
Pause Queue Member Extension	Pause the agent, and the agent cannot receive the call.
Unpause Queue Member Extension	Unpause the agent, and the agent can receive the call.
Others	Function key for Intercom/ Paging/ Directory
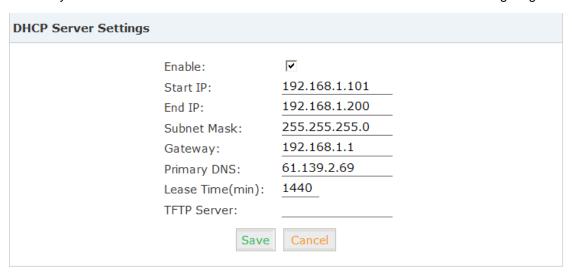
3.17 IP Phone Provisioning

When many IP Phones are needed, please record the MAC, extension number, and username of each phone according to the format (please take reference of the auto provision script file model for details) , then import the format file, once the phone is connected to the local
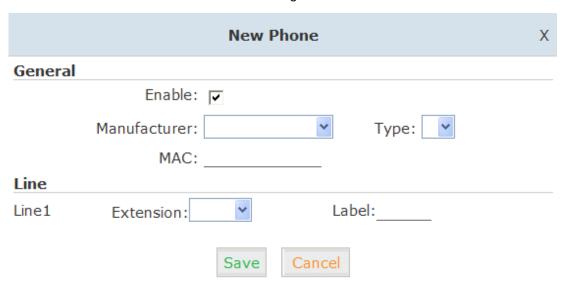
network, it will get the extension number and password automatically. There are two operation methods to fulfill this function, please see details as below.

Enable DHCP service

Click【System】-> 【Network Advanced】-> 【Enable】DHCP Server in the following diagram:

DHCP Server Settings

Enable:	☑
Start IP:	192.168.1.101
End IP:	192.168.1.200
Subnet Mask:	255.255.255.0
Gateway:	192.168.1.1
Primary DNS:	61.139.2.69
Lease Time(min):	1440
TFTP Server:	

Save   Cancel

Then Click【Advanced】-> 【Phone Provisioning】-> 【New Phone】:

New Phone   X

General

Enable: ☑

Manufacturer: [   ] ▼   Type: [ ] ▼

MAC: _____

Line

Line1   Extension: [   ] ▼   Label: _____

Save   Cancel

Enable Phone Provisioning in【Basic】, select the IP Phone manufacture, input MAC of the phone, and select the extension for provisioning.

⚠ **Notice**

CooVox IP PBX supports IP Phones from ZYCOO / Yealink / Grandstream now.

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Chapter 4   Network Settings

4.1 Network

You can configure the WAN Port, and define the Virtual Interface.

Click【Network Settings】-> 【Network】-> 【IPv4 Settings】:

Reference

Item	Explanation
IP Assign	Static/ DHCP/PPPoE supported.
Virtual Interface	Define the virtual interface for WAN Port.

Click【Network Settings】-> 【Network】-> 【IPv6 Settings】

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IPv6 Reference:

Item	Explanation
Enable	Enable IPv6, define the IPv6 address, gateway, and DNS.

Click【Network Settings】->【Network】->【VLAN Settings】:

VLAN Reference:

Item	Explanation
Enable	Enable VLAN, define the VLAN address and VLAN ID.

4.2 Static Routing

Click【Network Settings】->【Static Routing】:

Item	Explanation
Destination	Set destination network for static routing.
Subnet Mask	Set subnet mask of the destination network.
Gateway	Define the gateway accessing the destination network.

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Click【Network Settings】->【Static Routing】->【Routing Table】, the current routing information will be displayed as below:

Routing Table

Routing Table:
```
Kernel IP routing table
Destination     Gateway          Genmask          Flags Metric Ref    Use Iface
0.0.0.0         192.168.1.1      0.0.0.0          UG    0      0        0 eth0
192.168.1.0     0.0.0.0          255.255.255.0    U     0      0        0 eth0
```

4.3 VPN Server

CooVox IP PBX supports three kinds of VPN servers: L2TP/PPTP/OpenVPN.
Click【Network Settings】->【VPN Server】：

Reference:

Item	Explanation
VPN Server Mode	Three kinds of VPN servers L2TP/PPTP/OpenVPN supported（Only one mode can be enabled simultaneously)
Enable	Enable/Disable VPN Server

When the mode is L2TP or PPTP VPN server, click【Network Settings】->【VPN Server】->【VPN Users Management】：

	VPN Server		VPN Users Management	
List of VPN Users			New VPN User	
	Username		Availability	Options
1	test1		yes	Edit Delete

This page is used for management of VPN username and password.

When the mode is OpenVPN server, click【Network Settings】->【VPN Server】->【OpenVPN Certificate Download】:

	VPN Server		OpenVPN Certificate Download	
List of OpenVPN Certificate		New Certificate		Delete Selected
☐	Certificate Name			Options
☐	1	Client1.tar		Download Delete

This page is used for management of OpenVPN certificate file.

4.4 VPN Client

CooVox IP PBX supports four kinds of VPN Clients: L2TP /PPTP /OpenVPN /N2N
Click【Network Settings】->【VPN Client】:

VPN Client

○ L2TP ● PPTP ○ OpenVPN ○ N2N

Enable: ☑
Enable 40/128-bit encryption for MPPE: ☐
Server Address: 192.168.100.100
Username: admin
Password: •••••

Save Cancel
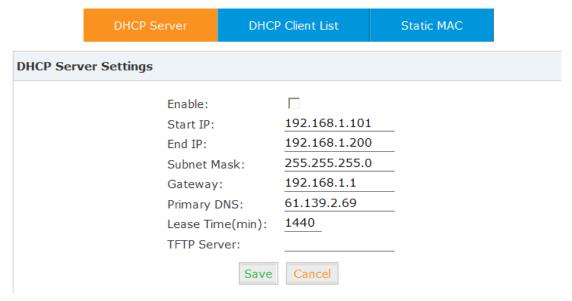
```
Status:pptp client Connect: ppp1 <--> /dev/pts/2
       pptp client sh: can't execute '/sbin/ip': No such file or directory
       pptp client sh: can't execute '/sbin/ip': No such file or directory
```

Reference:

Item	Explanation
VPN Client	Four kinds of VPN Clients supported: L2TP/PPTP/OpenVPN/N2N（Only one mode can be enabled simultaneously）
Enable	Enable/Disable VPN Client

4.5 DHCP Server

Click【Network Settings】->【DHCP Server】:

DHCP Server	DHCP Client List	Static MAC

DHCP Server Settings

Enable:	☐
Start IP:	192.168.1.101
End IP:	192.168.1.200
Subnet Mask:	255.255.255.0
Gateway:	192.168.1.1
Primary DNS:	61.139.2.69
Lease Time(min):	1440
TFTP Server:	

Save   Cancel
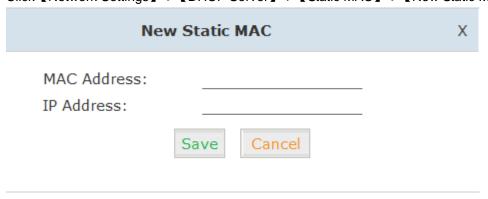
Click【Network Settings】->【DHCP Server】->【DHCP Client List】:

DHCP Server	DHCP Client List	Static MAC

DHCP Client List:

```
Mac Address         IP Address        Host Name              Expires in
6c:3e:6d:e0:f2:00   192.168.1.101     iPhone                 expired
00:03:58:45:87:9a   192.168.1.102                            expired
0c:74:c2:47:71:6d   192.168.1.103     hnteki-iPhone          expired
20:c9:d0:85:3b:fb   192.168.1.104                            expired
08:ed:b9:e7:c5:7f   192.168.1.105     DPVYE1J0WCAAC7I        expired
78:e4:00:8e:c3:99   192.168.1.106     LBSZLACHCIC            22:10:25
68:a3:c4:ef:5d:8b   192.168.1.107     HBWang                 1 days 00:00:00
0c:72:2c:5a:39:41   192.168.1.108     MW150R                 00:00:57
```

This page is used to display DHCP Client address and related information.

When DHCP Server distributes address, the Client's MAC address is associated with the IP address, and then the device will get the same IP address every time.

Click【Network Settings】->【DHCP Server】->【Static MAC】->【New Static MAC】:

New Static MAC   X

MAC Address:	
IP Address:	

Save   Cancel

4.6 DDNS Settings

After setting DDNS (Dynamic Domain Network Server), CooVox IP PBX settings will be visited remotely. Click【Network Settings】-> 【DDNS Settings】:

DDNS Settings

Enable: ☐
DDNS Server: dyndns.org
Username: _____
Password: _____
Domain: _____

Save   Cancel

Status:Disabled

CooVox supports DDNS provided by Dyndns.org / No-ip.com / zoneedit.com.

4.7 SNMPv2 Settings

SNMP(Simple Network Management Protocol): Used for remote management.
Click【Network Settings】-> 【SNMPv2 Settings】:

SNMPv2 Settings

Read Only

Enable: ☐
RO Community: public
RO Network: _____ / ____

Read and Write

Enable: ☐
RW Community: private
RW Network: _____ / ____

Save   Cancel

Reference
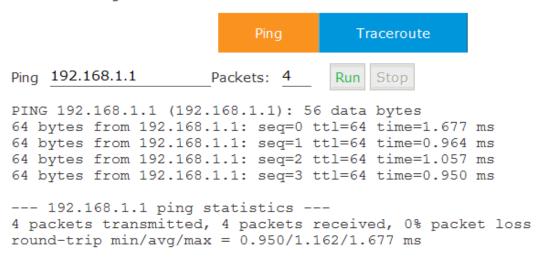
Item	Explanation
Enable	Enable "Read Only" of SNMP
RO Community	Define the name of RO Community of SNMP
RO Network	Define network of RO

4.8 Trouble Shooting

You can ping other network device through CooVox IP PBX and track network routing by command "Traceroute" .

Click【Network Settings】-> 【TroubleShooting】:

Troubleshooting

Ping	Traceroute

Ping 192.168.1.1   Packets: 4   Run Stop

```
PING 192.168.1.1 (192.168.1.1): 56 data bytes
64 bytes from 192.168.1.1: seq=0 ttl=64 time=1.677 ms
64 bytes from 192.168.1.1: seq=1 ttl=64 time=0.964 ms
64 bytes from 192.168.1.1: seq=2 ttl=64 time=1.057 ms
64 bytes from 192.168.1.1: seq=3 ttl=64 time=0.950 ms

--- 192.168.1.1 ping statistics ---
4 packets transmitted, 4 packets received, 0% packet loss
round-trip min/avg/max = 0.950/1.162/1.677 ms
```

Chapter 5   Security

5.1 Firewall

Click【Security】->【Firewall】

| Command: iptables _____ | Run |

Result:

IP Tables List:

```
Chain INPUT (policy ACCEPT)
target     prot opt source            destination

Chain FORWARD (policy ACCEPT)
target     prot opt source            destination

Chain OUTPUT (policy ACCEPT)
target     prot opt source            destination
```

Iptables Command:

Check iptables list   iptables -L -n

Clear iptables list   iptables -F

Deny an IP（192.168.0.3）   iptables -A INPUT -s 192.168.0.3 -j DROP

Deny every IP to access 80 port   iptables -A INPUT -p tcp --dport 80 -j DROP

Deny IP（192.168.0.3）to access 80 port   iptables -A INPUT -s 192.168.0.3 -p tcp
                                             --dport 80-j DROP

5.2 Service

【Service】: settings of SSH/FTP and HTTP Port.
Click【Security】->【Service】:

Service Settings

Enable SSH: ☑ Port:22
Enable FTP: ☑ Port:21
HTTP Port: 80

Save   Cancel

Enable SSH to login background management system through SSH.
Enable FTP to allow uploading files to system through FTP.

5.3 SIP Allowed Address

Define an allowed address, from which every SIP request will never be filtered or refused.

Click【Security】-> 【SIP Allowed Address】:

SIP Allowed Address

List of SIP Allowed IP Address	Add Allowed IP	
Allowed IP		Options
No *SIP Allowed Address* defined!		

Add Allowed IP   X

Allowed IP: 192.168.15.0

Subnet Mask: 255.255.255.0

Save   Cancel

Chapter 6   Report

6.1 Register Status
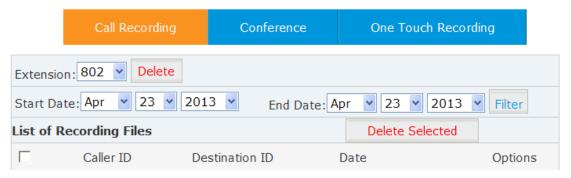
Check status of all kinds of users & trunks.

Register Status ⟳

SIP Users Status	IAX2 Users Status	SIP Trunks Status	IAX2 Trunks Status
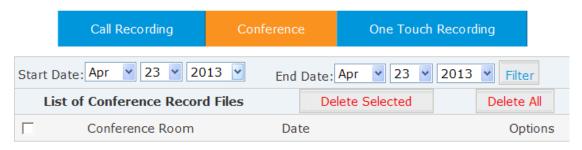
6.2 Record List

Check recordings of specified extension or conference here, or delete the recording file.

【Record List】:

Call Recording	Conference	One Touch Recording

Extension: 802   Delete

Start Date: Apr 23 2013   End Date: Apr 23 2013   Filter

List of Recording Files   Delete Selected

☐	Caller ID	Destination ID	Date	Options

【Conference】:

Call Recording	Conference	One Touch Recording

Start Date: Apr 23 2013   End Date: Apr 23 2013   Filter

List of Conference Record Files   Delete Selected   Delete All

☐	Conference Room	Date	Options

【One Touch Recording】

Call Recording	Conference	One Touch Recording

Extension: Delete

Start Date: Apr 23 2013   End Date: Apr 23 2013   Filter

List of Recording Files   Delete Selected

☐	Caller ID	Destination ID	Date	Options

6.3 Call Logs

Check call logs by caller ID or callee ID.

Click【Report】->【Call Logs】:

⚠️ **Notice**
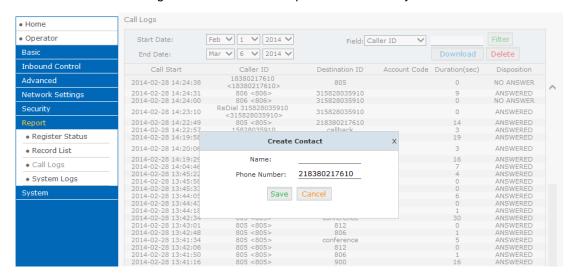
Duration in the call logs is not real charged duration. If you need billing, PSTN must support polarity reversal function, and meanwhile, you must configure relevance parameters of polarity reversal in trunk configuration for the CooVox IP PBX.

The number in the call logs can be added in the phone book directly:

6.4 System Logs

Click【Report】->【System Logs】, you can download/ delete the system logs.

System Logs

Enable System Log: ☐ Enable PBX Log: ☐

Enable PBX Debug Log: ☐ Enable Access Log: ☐

Save Cancel

List of Logs ↻   Download Selected   Delete Selected

☐		Name	Type		Options	
☐	1	login201303.log	Login Log		Delete	Download
☐	2	login201304.log	Login Log		Delete	Download
☐	3	pbx20130311.log	PBX Log		Delete	Download
☐	4	pbx20130313.log	PBX Log		Delete	Download
☐	5	pbx20130315.log	PBX Log		Delete	Download
☐	6	pbx20130319.log	PBX Log		Delete	Download
☐	7	pbx20130320.log	PBX Log		Delete	Download

Chapter 7   System

7.1 Hot Standby (For U100 only)

The function will working between the two Coovox-U100 devices. When the primary server faults, the slave server will replace it.

Hot Standby

Hot Standby Settings

Enable:	☐
Hot Standby Mode:	[▼]
Local Hostname:	_____
Remote Hostname:	_____
Local IP:	_____
Local Heart line:	7790
Local Port:	7788
Remote IP:	_____
Remote Heart line:	7789
Remote Port:	7788
Virtual IP:	_____
SYNC Network Rate:	100Mbps ▼
Status Fresh Time(sec):	5
Remote Link Timeout(sec):	15
Administator Email:	_____
Administator Phone Number:	_____

Save   Cancel
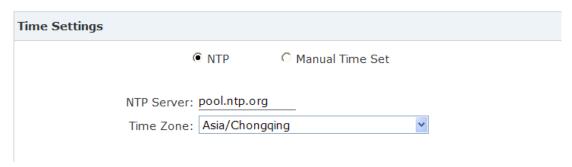
Item	Explanation
Enable	Enable 'Hot Standby' function.
Hot Standby Mode	Set the local server hot standby mode.
Local Hostname	Set the local server host name.
Remote Hostname	Set the remote server host name
Local IP	Set the local server IP address.
Local Heart Line Port	Set the local server heart line port
Local Port	Set the local server port (default: 7788)
Remote IP	Set the remote server IP address
Remote Heart Line Port	Set the remote server heart line port
Remote Port	Set the remote server port(default: 7788)
Virtual IP	Set the virtual IP address. The primary server and slave server must use same virtual IP address
SYNC Network Rate	Select the server network rate.
Status Fresh Time	Set the status fresh time(sec)
Remote Link Timeout	Set the remote link timeout(sec)

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Administrator Email	Set the administrator email, if the primary server faults will send email to administrator.
Administrator Phone Number	Set the administrator phone number, if the primary server faults will call administrator

7.2 Time Settings
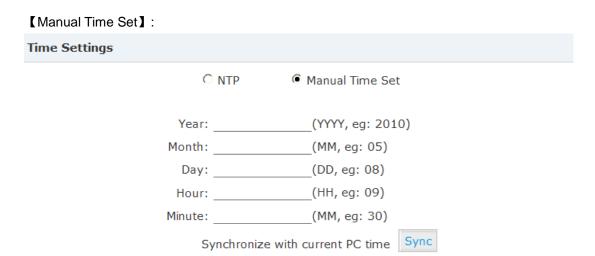
Time settings for CooVox system. The system supports either NTP or Manual Time Set.

【NTP】:

Time Settings

◉ NTP   ○ Manual Time Set

NTP Server: pool.ntp.org

Time Zone: Asia/Chongqing ▾

Reference:

Item	Explanation
NTP Server	Define the NTP Server. You can input the IP address or domain of this server, whether it's local or remote. Default server is pool.ntp.org. Be aware that the CooVox IP PBX needs to be able to connect to an NTP server to properly function.
Time Zone	Select your time zone so that the system will set time based on the time zone.

【Manual Time Set】:

Time Settings

○ NTP   ◉ Manual Time Set

Year: _____(YYYY, eg: 2010)

Month: _____(MM, eg: 05)

Day: _____(DD, eg: 08)

Hour: _____(HH, eg: 09)

Minute: _____(MM, eg: 30)

Synchronize with current PC time   Sync

After entering Year/ Month/ Day/ Hour/ Minute, then save and activate.

Or, you can click【Sync】to synchronize with current PC time.

7.3 Module Settings (Support for U50/U100)

When use the module except FXO/FXS/GSM. You need to set the module parameters with the page.

Click【System】-> 【Module Settings】:

Module Settings

SLOT 1	
Module Type:	FXS/FXO/GSM ∨
	Save Cancel

Module Type: Select the module type

● FXS/FXO/GSM module   Default type. You don't need set anything for
                                those modules.

● E1/T1 module

Module Settings

SLOT 1	
Module Type:	E1/T1 ∨
E1/T1 Settings:	
Mode:	E1 ∨
Signalling:	CPE ∨
Framing:	CCS ∨
Coding:	HDB3 ∨
CRC4:	☑
	Save Cancel

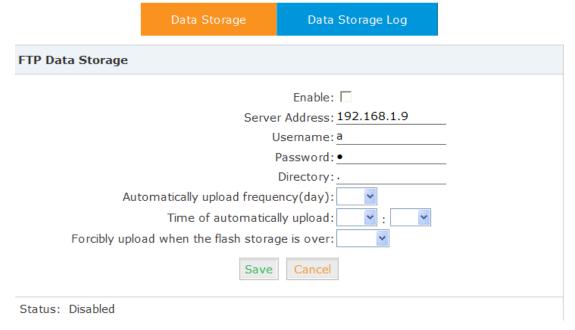| Item	Explanation
Mode	Set E1 or T1 mode for the module.
Signaling	Set the module signaling.
Framing	One of 'd4' or 'esf' for T1; 'cas' or 'ccs' for E1.
Coding	One of 'ami' or 'b8zs' for T1; 'ami' or 'hdb3' for E1.
CRC4	Enable CRC4 Verification.

● ISDN BRI module

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Module Settings

SLOT 1

Module Type:	ISDN BRI
BRI Settings:	
Type of Port 1:	NT
Type of Port 2:	NT
Type of Port 3:	TE
Type of Port 4:	TE
	Save Cancel

Set the BRI NT or TE mode for each port.

7.4 Data Storage

When you need mass storage of recording files, voicemails, call logs, etc, you can upload these files to FTP server through FTP Data Storage based on the specified time frequency Click【System】-> 【Data Storage】:

Data Storage	Data Storage Log

FTP Data Storage

Enable:	☐
Server Address:	192.168.1.9
Username:	a
Password:	●
Directory:	.
Automatically upload frequency(day):	
Time of automatically upload:	:
Forcibly upload when the flash storage is over:	
	Save Cancel

Status: Disabled

Reference

Item	Explanation
Enable	Enable FTP Data Storage.
Server Address	Set FTP server address (IP address or domain).
Username	Username for login FTP.
Password	Password for login FTP.
Directory	Define a directory used for storage on FTP server.
Automatically upload frequency (day)	Define frequency by days to upload the data.
Time of automatically upload	Define the time to upload the data.
Forcibly upload when the flash	Forcibly upload data when flash storage is over the percentage value.

storage is over	

Check from【Data Storage Log】:

Click【Refresh】to refresh data storage log.
Click【clear】to clear data storage log.

7.5 Management

【Management】is used for modify password of CooVox system, and the settings of system voice.
Click 【System】->【Management】:

7.6 Backup

Click【System】->【Backup】

Backup	Upload Backup File

List of Backups   Take a Backup

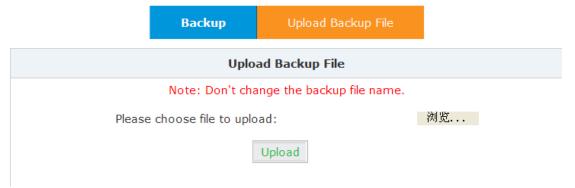|	Name	Date	Options	
1	backup_2013jan09_135847	Jan 09, 2013	Restore Delete	⏬
2	backup_2013jan09_135854	Jan 09, 2013	Restore Delete	⏬
3	backup_2013mar13_155906	Mar 13, 2013	Restore Delete	⏬
4	backup_2013mar28_174911	Mar 28, 2013	Restore Delete	⏬
5	backup_2013mar28_174938	Mar 28, 2013	Restore Delete	⏬

Reference:

Item	Explanation
Take a Backup	Take a backup of the current system configuration.
Restore	Restore system to the specified backup configuration.
Delete	Delete specified backup file.

Click the download button " ⏬ " to download the specified backup file and manage locally.

Click【Upload Backup File】to upload the backup file here.

Upload Backup File

Backup	Upload Backup File

Upload Backup File

Note: Don't change the backup file name.
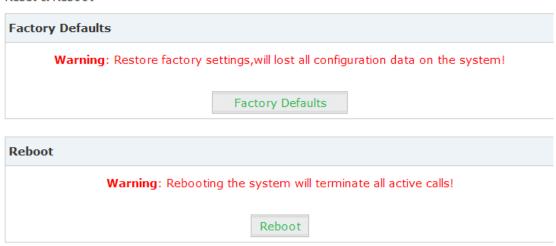
Please choose file to upload: 浏览...

Upload

Click【browse】to select the local backup file, and click【Upload】to upload the backup file to system.

7.7 Reset & Reboot

If you need reset the system to factory defaults or reset, please click【System】->【Reset & Reboot】:   Restoring factory settings will make configuration data in the system lost.

Reset & Reboot

Factory Defaults

Warning: Restore factory settings,will lost all configuration data on the system!

Factory Defaults

Reboot

Warning: Rebooting the system will terminate all active calls!

Reboot

Click【Factory Defaults】to reset the system to factory defaults.

Click【Reboot】to reboot the system.

7.8 Upgrade

7.8.1 WEB Upgrade

Click【System】-> 【Upgrade】-> 【WEB Upgrade】:

Upgrade System Package

⦿ WEB Upgrade   ◯ TFTP Upgrade

Restore Default Set: ☐

Please choose file to upload: 浏览...

Upload

Click【Browse】to select the firmware file, then click【Upload】to upload the selected firmware to system and finish the upgrading automatically.

If check【Restore Default Set】, the system will clear all the configuration and reset to factory default.

7.8.2 TFTP Upgrade

Click【System】-> 【Upgrade】-> 【TFTP Upgrade】:

Upgrade System Package

○ WEB Upgrade   ● TFTP Upgrade

Restore Default Set: ☐
Enter The Package Name: uImage-md5.u50 _____
TFTP Server IP address: _____
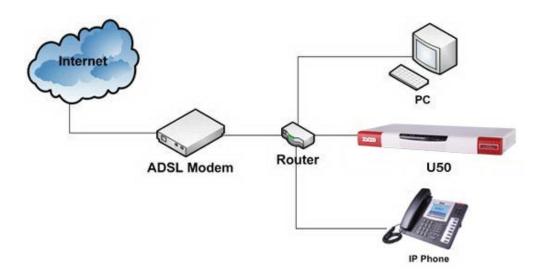
Start

Reference:

Item	Explanation
Restore Default Set	System will restore to factory defaults after checking this option.
Enter The Package Name	Enter the package name for upgrading.
TFTP Server IP address	Enter your TFTP server IP address.

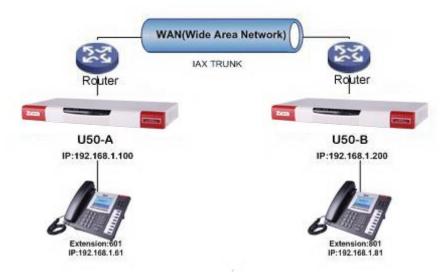## Chapter 8   Operating Instructions

(Take CooVox-U50 as example)

8.1 How to connect CooVox-U50 in the Network

If your office accesses the public network through router, you can put the CooVox IP PBX behind the router. You should connect the WAN port of the IP PBX to the LAN port of the router.

8.2 How to combine two sets CooVox IP PBX in the same network

We start to combine two IP PBXs in the same network and then try to expand to different network. Combine two IP PBXs in the same LAN from the structure as below:

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Register U50-B IP to a trunk of U50-A, and register U50-A IP to a trunk of U50-B, without authentication for each registration.

Configuration Rule:

1. IP Phone registers on CooVox-U50-A with extension number 601.
2. Another IP Phone registers on Coovox-U50-B with extension number 801.
3. CooVox-U50-A WAN IP: 192.168.1.100.
4. CooVox-U50-B WAN IP: 192.168.1.200.
5. Extension format of CooVox-U50-A: 6XX.
6. Extension format of CooVox-U50-B: 8XX.
7. All extensions on U50-A can call extensions on U50-B by 8XX format.
8. All extensions on U50-B can call extensions on U50-A by 6XX format.

Step1: Register U50-B IP to a trunk of U50-A

CooVox-U50-A: Click【Basic】-> 【Trunks】-> 【New VoIP Trunk】:

Step2: Register U50-A IP to a trunk of U50-B as the same way of step 1.

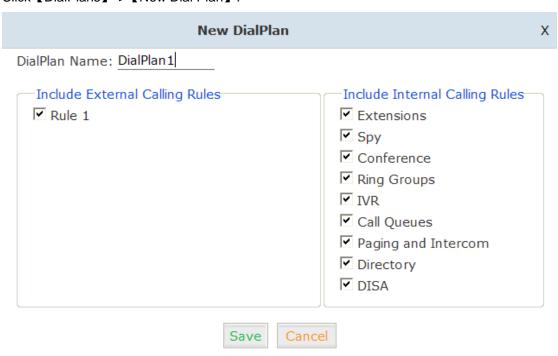**Step 3:** Create DialRule on U50-A, and add the DailRule to the DialPlan

Click【Outbound Routes】-> 【DialRules】-> 【New Dial Rule】:

New DialRule   X

Rule Name: rule 1

PIN Set: ☐

Place this call through:

Available Trunks		Selected Trunks
	»»	U50-A(SIP)
	→	
	←	
	««	

Custom Pattern: _____

Z Any digit from 1 to 9
N Any digit from 2 to 9
X Any digit from 0 to 9
. Any number of additional digits

Delete ___ digits prefix from the front and auto-add digit _____ before dialing

Save   Cancel

Select the created line 192.168.1.200 to 【Selected Trunks】, custom pattern is XXX, save and activate.

Click【DialPlans】-> 【New Dial Plan】:

New DialPlan   X

DialPlan Name: DialPlan1

Include External Calling Rules
☑ Rule 1

Include Internal Calling Rules
☑ Extensions
☑ Spy
☑ Conference
☑ Ring Groups
☑ IVR
☑ Call Queues
☑ Paging and Intercom
☑ Directory
☑ DISA

Save   Cancel

Check the created calling rule, save and activate.

Step4: Create dialrule on CooVox-U50-B, add the created dialrule to the dialplan as the same way of Step 3.
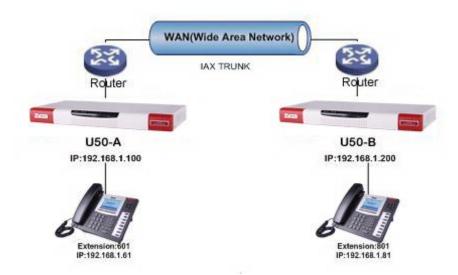
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Step 5: Activate the current configuration and test:

1. Register IP Phone to U50-A as extension 601.

2. Register another IP Phone to U50-B as extension 801.

3. Make a call from 801 to 601, 601 rings and the call is connected.

4. Make a call from 601 to 801, 801 rings, and the call is connected.

8.3 How to connect two sets CooVox IP PBXs in different network?

E.g.: two sets CooVox-U50 in the internet.

Normally, the two sets CooVox-U50 are located in different place; but they are in the internet, and have public IP address.

Note: Enable NAT on Router.

For external line configuration, you must use public IP address.

Take the following instructions as example:

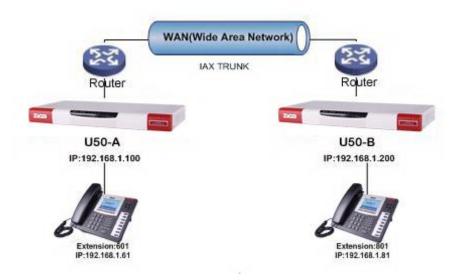Register U50-B IP to a trunk of U50-A with authentication.

Configuration Rule:

1. IP Phone registers on U50-A as extension 601.

2. Another IP Phone registers on U50-B as extension 801.

3. U50-A IP:192.168.1.100.

4. U50-B IP:192.168.1.200.

5. Extension format of U50-A: 6XX.

6. Extension format of U50-B: 8XX

7. Create an extension 888 with password 123456 on U50-B.

8. All extensions on U50-A can call extensions on U50-B with format 8XX.

9. All extensions on U50-B can call extensions on U50-A with format 6XX.

For detail steps, please take chapter 8.2 as reference.
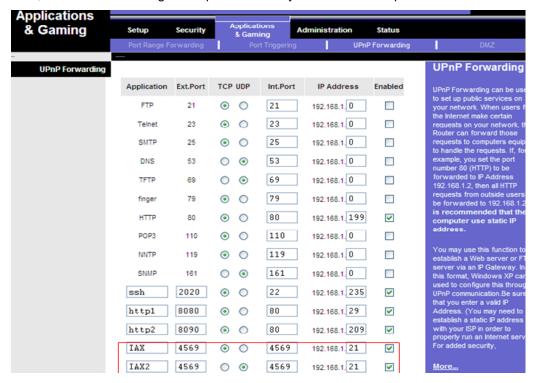
Two sets U50 behind router

Sometimes U50 doesn't have public IP, and you have to configure port mapping for your router.

Step1: Configure the mapping rule of U50-A on the router.

U50-B is connected behind the router, registers on U50-A through internet, you need configure the port mapping of IAX2 port(4569) on the router. Then, all data received from WAN port of router(192.168.1.100:4569) will be sent to U50-A

Now, take the web management panel of Linksys router as example.

Step2: U50 Configuration

Configure the trunk and dialplan on U50-B, register U50-B IP to U50-A, configuration is same as above, but you have to replace the public IP with internal IP:192.168.1.21.

Step3: Configure port mapping rule of U50-B on the router

Configure port mapping of U50-B on the router as the same way of step1..

Step4: Connect two sets U50 and make the call

Create extension 601 on U50-A, extension 801 on U50-B, and create the correct outbound rule.

 Notice

Public IP must be provided by network provider. It could be dynamic IP address, and easy to change; you can resolve this problem by using DDNS.

8.4 How to resolve the problem "one-way" audio problems

If U50 is behind router, to resolve the problem, please set up IP address as below:

Click【Advanced】->【Option】->【Global SIP Settings】:

NAT Support

External IP: _____

External Host: _____

External Refresh(sec): _____

Local Network Address: _____

1. External IP   External IP or domain to replace the device IP
2. External Host   External domain to replace the device IP
3. External Refresh(sec)   Refresh time, default is 10 seconds.
4. Local Network Address   IP address and subnet mask needed to be converted .
                             E.g.: 192.168.1.100/255.255.255.0

8.5 How to use Skype on CooVox-U50

8.5.1 Visit the Top-up Page

Visit the top-up page: http://skype.tom.com/products/en/skypeout.html

Select subscription, payment method and enter the Skype account to top up credit.
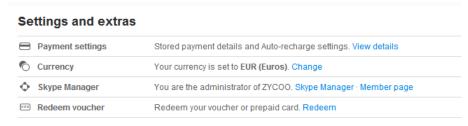
 Notice

First top up for business account must be more than €50.

8.5.2 Manage Skype Account

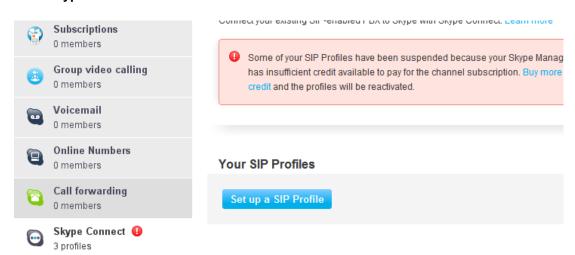After login, you will find **Skype Manager**, and click it.
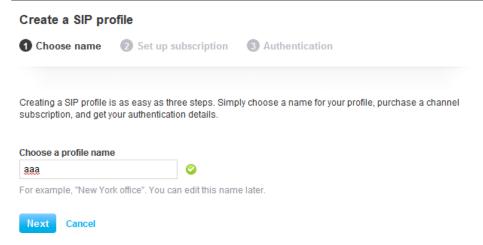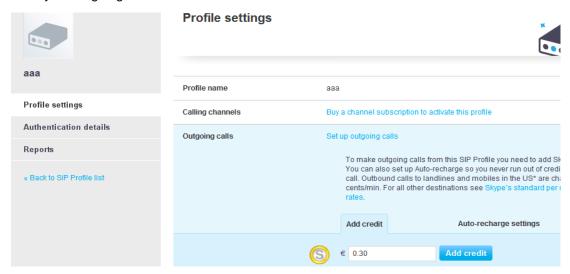

8.5.3 Create a SIP File
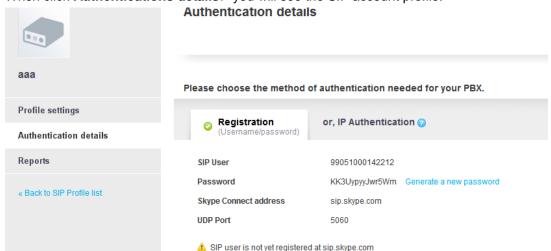
Click **Features:**

Click **Skype connect:**

Click **Set up a SIP Profile:**

Create a SIP profile

1 Choose name   **2** Set up subscription   **3** Authentication

Creating a SIP profile is as easy as three steps. Simply choose a name for your profile, purchase a channel subscription, and get your authentication details.

Choose a profile name

aaa ✓

For example, "New York office". You can edit this name later.

Next   Cancel

Create a SIP account, and each account has a channel, you need pay €5 for each channel as monthly rent. Then input the registration profile in the VoIP trunk of U50 and distribute the money to Outgoing calls.

Profile settings

aaa

Profile settings

Authentication details

Reports

« Back to SIP Profile list

Profile name	aaa
Calling channels	Buy a channel subscription to activate this profile
Outgoing calls	Set up outgoing calls

To make outgoing calls from this SIP Profile you need to add Sk You can also set up Auto-recharge so you never run out of credi call. Outbound calls to landlines and mobiles in the US* are cha cents/min. For all other destinations see Skype's standard per i rates.

Add credit   Auto-recharge settings

(S)   € 0.30   **Add credit**

When click **Authentications details**，you will see the SIP account profile:

Authentication details

aaa

Profile settings

Authentication details

Reports

« Back to SIP Profile list

Please choose the method of authentication needed for your PBX.

✓ **Registration**
(Username/password)   or, IP Authentication ?

SIP User	99051000142212
Password	KK3UypyyJwr5Wm Generate a new password
Skype Connect address	sip.skype.com
UDP Port	5060

⚠ SIP user is not yet registered at sip.skype.com

Select the created line 192.168.1.200 to 【Selected Trunks】, custom pattern is XXX, save and activate.

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For any questions or problems during installation and use,
please feel free to contact our technical support via
email: support@zycoo.com
or phone : 0086 28 85337096.