

## **XonTel Plus PBX**

## **User Manual**







## Preface

## Welcome

Thanks for choosing the **XonTel Plus PBX**. We hope you will make full use of this rich-feature PBX. Contact us if you need any technical support.

## About This Manual

This manual provides information about the introduction of the XonTel Plus PBX, and about how to install, configure or use the PBX. Please read this document carefully before install the PBX.

## **Intended Audience**

This manual is aimed primarily at the following people:

- Users
- Engineers who install, configure and maintain the PBX.

## Conventions

PBX or device mentioned in this document refers to the XonTel Plus PBX. Those words in blue are the contents that users need to pay attention to.



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## **1 Product Introduction**

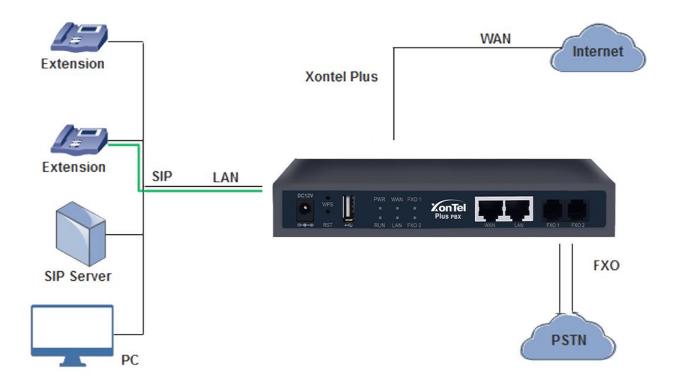
## 1.1 Overview

The XonTel Plus PBX is a multi-functional and all-in-one PBX, which integrates voice service which is VoIP and PSTN. It provides FXO interfaces, offering seamless connectivity to VoIP Network and PSTN.

XonTel Plus is ideally suitable for personal use. Meanwhile, it is perfect for small and micro enterprises, offering high-speed internet access and good voice service.

## **1.2 Application Scenario**

The application scenario of XonTel Plus PBX is shown as follows:





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## **1.3 Product Appearance**

#### Front View:



Back View:



## **1.4 Description of Indicators**

Indicator	Definition	Status	Description
PWR	Power Indicator	Off	There is no power supply or power supply is abnormal.
		On	The XonTel Plus device is powered on.
		Slow Flashing	The device is initialized successfully and is running normally
RUN	Running Indicator	On	The device is being initialized.
			The device is not running normally.
		Fast Flashing	The FXO port is connected with PSTN line and is in idle status
FXO	FXO In-use Indicator	Slow Flashing	The FXO port has yet to be connected with PSTN line, but is in normal status.
		On	The FXO port is currently occupied by a call.
		Off	The FXO port is faulty.
XX/A XY/I A XY	Network Connection	Off	Network does not work or network cable is not connected to the WAN/LAN port.
WAN/LAN	Indicator	Fast Flashing	Network is successfully connected.







## 1.5 Features & Functions

#### 1.5.1 **Key Features**

- FXO interface on a single PBX.
- Send/receive calls from PSTN/PLMN via FXO.
- Flexible dial plan and routing strategies based on time, number and source IP etc.
- IVR Customization.
- Support high-speed NAT forwarding.
- Serve as VPN client.
- Support voicemail and call recording.
- Built-in SIP server, support up to 60 SIP extensions and 15 concurrent calls.
- User-friendly web interface, multiple management ways.

#### **1.5.2 Physical Interfaces**

- FXO Ports: 2
- USB port: 1
- SD Slot: 1
- Network Port: 1 WAN Port & 1 LAN Ports (10/100 Base-T RJ45)

### 1.5.3 Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS, SDP, RTP/SRTP
- •Codecs: G.711a/µ law, G.723.1, G.729A/B, G722
- •Silence Suppression
- •Comfort Noise Generator (CNG)
- •Voice Activity Detection (VAD)
- •Echo Cancellation: G.168 with up to 128ms
- •Dynamic Jitter Buffer
- •Adjustable Gain Control
- •Automatic Gain Control (AGC)
- •Call Progress Tones: Dial Tone, Ring Back Tone, Busy Tone
- •FAX: T.38 and Pass-through
- •NAT Traversal: STUN/UPnP
- •DTMF: RFC2833/Signal/Inband









#### 1.5.4 **FXO**

- •FXO Connector: RJ11
- •Caller ID: FSK and DTMF
- Polarity Reversal
- •Answer Delay
- •Busy Tone Detection
- •No Current Detection

#### 1.5.5 Software Features

- •Ring Group
- •Routing Groups
- •Caller/Called Number Manipulation
- •Routing Based on Time Period
- •Routing Based on Caller/Called Number Prefix
- •Routing Based on Source Trunks
- •Dial Rules
- •Failover Routing
- •FXO Impedance Auto Match
- •IVR Customization
- •Auto Attendant Function
- •CDRs

### 1.5.6 Supplementary Services

- •Call Forwarding (Unconditional/Busy/No Reply)
- •Call Waiting and Call Holding
- •Call Transfer (Blind & Attended)
- •Call Queuing
- •Intra-group Pick-up
- •Auto-answer
- •Hotline
- •No Disturbing
- Voicemail
- •Three-way Conversation



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#### 1.5.7 Environmental

- •Power Supply: 12VDC, 2A
- •Power Consumption: 18W
- •Operating Temperature: 0  $^{\circ}$ C ~ 45  $^{\circ}$ C

Storage Temperature: -20 °C~80 °C

- •Humidity: 10%-90% (Non-Condensing)
- •Dimensions:  $260 \times 180 \times 35 mm$  (W/D/H)
- •Weight: 1.0kg

#### 1.5.8 Maintenance

- •Web GUI for Configuration
- •Telnet Management
- •Configuration Restore & Backup
- •Multiple Languages
- •Firmware Upgrade: support HTTP/HTTPS/TFTP/FTP
- •Auto Provision
- •CDR Query and Export
- •Syslog Query and Export
- •Network Tools: Ping, Traceroute and Nslookup
- •Flow Statistics: TCP, UDP, RTP
- Network Capture



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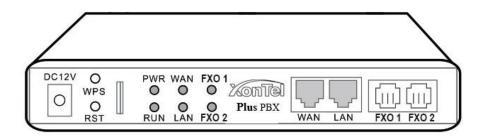
## **2** Quick Installation

## 2.1 Installation Attentions

To avoid unexpected accident or device damage, please read the following instructions before you install the XonTel Plus PBX.

- The adapter of the PBX accepts DC input voltage of 12V 2A. Please ensure stable and safe power supply;
- To reduce the interference to telephone calls, please separate power cables from telephone lines;
- To guarantee stable running of the PBX, please make sure that there is enough network bandwidth;
- For better heat dissipation, please place the PBX on a flat surface and do not pile up with other devices;

## 2.2 Installation Steps



- Connect the power adapter to the power jack;
- Connect PSTN lines to the FXO ports;
- Connect network cable to the LAN port(s) and WAN port (please refer to 2.3 Network Connection);

## 2.3 Network Connection

XonTel Plus works in two network modes: route mode and bridge mode. When it is under the route mode, the IP address of WAN port must be different from the IP address of LAN port. But when it is under the bridge mode, the IP address of WAN port and that of LAN port are the same.



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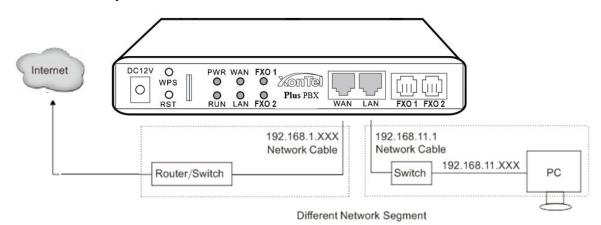
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#### 2.3.1 Network Connection Diagram under Route Mode

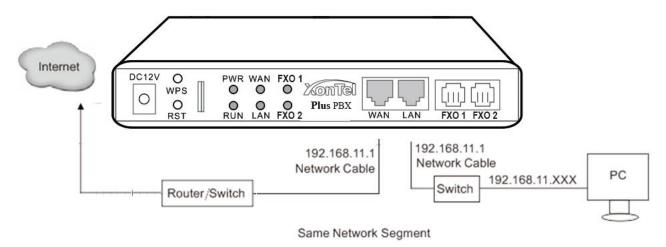
Under the route mode, the default IP address of WAN port is a DHCP IP address, while the default IP address of the LAN port is a static IP address, namely **192.168.11.1**.



Note: The IP address of LAN port of the PBX and the IP address of PC must be at the same network segment, while that of WAN port is at a different network segment.

### 2.3.2 Network Connection Diagram under Bridge Mode

Under the Bridge mode, the IP address of WAN port is the same with that of LAN port.



Note: The IP address of PC and that of WAN port of the XonTel Plus PBX are at the same network segment.



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## 2.4 Connect PBX to Network

#### 2.4.1 Connect PBX to Network via Network Port

Please connect the XonTel Plus PBX to network according to the network diagrams in Section 2.3 Network Connection. Connect a PSTN line to the FXO port. Use a mobile phone to dial the number of the FXO port, and then dial \*158# to query the IP address of LAN port after hearing IVR. Modify the IP address of PC to make it at the same network segment of LAN port of the PBX.

You are also allowed to log in the PBX by using the WAN port, but you need to enable the port first.

### 2.4.2 Preparations for Login

Modify the IP address of the PC to make it at the same network segment with the XonTel Plus PBX, since the default IP address of LAN port of the PBX is **192.168.11.1**.

Check the connectivity between the PC and the XonTel Plus. Click **Start**  $\rightarrow$  **Run** of PC and enter **cmd** to execute '**ping 192.168.11.1**' to check whether the IP address of LAN port runs normally.

### 2.4.3 Login to Web Interface

Open a web browser and enter the IP address of LAN port (**the default IP is 192.168.11.1**). Then the login GUI will be displayed.

You also can enter the IP address of WAN port, but it's required to modify the IP address of PC to make it at the same network segment with WAN port.

It is suggested that you should modify the username and password for security consideration.

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Login

By default, the username is **admin**, while the password is **xontel**. After entering username and password, click **Login** to enter into the web interface.

Under some circumstances, login of the Web will be limited:

- For three consecutive login failures, you need to slide to validate your user account;
- Failing to log in the Web for ten times consecutively, the IP address of the XonTel Plus device will be put into the blacklist, and you need to reset a new IP address for the device;
- Successful login or device restart will wipe out login failure records.







## **3 Basic Operation**

## 3.1 Methods to Number Dialing

• Dial the called number and press #.

## 3.2 Call Holding

If a calling party places a call to a called party which is otherwise engaged, and the called party has the call holding feature enabled, the called party is able to switch to the new incoming call while keeping the current call holding on by pressing the flash button or the flash hook.

When the called party presses the flash button or the flash hook once again, he or she will switch back to the first call.

## 3.3 Call Waiting

If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the calling party will hear a IVR voice '**Please hold on, the subscriber you dialed is busy**' and the called party will hear three beeps.

By pressing the flash button or the flash hook, the called party is able to switch between the new incoming call and the current call.

## 3.4 Call Transfer

## 3.4.1 Blind Transfer

Blind transfer is a call transfer in which the transferring party connects the call to a third party without notifying the third party.

Example: A gives a call to B and B wants to blindly transfer the call to C. Operation instructions are as follows:

- 1. A dials the extension number of B;
- 2. The extension of B rings, and B picks up the phone. Then A and B go into conversation;
- 3. B presses **\*3** to trigger blind transfer (at the same time, A can hear the waiting tone). Then B dials the extension number of C (end up with # or wait for 4 seconds);

4. The extension of C rings, B hangs up the phone and C picks up the phone. Then C and A goes into conversation.

Note:

- On the 'Call Control  $\rightarrow$  Feature Code' page, feature code service should be 'On'.
- If B hears continuous busy tones after he dials the extension number of C, it means the call has timed out.







#### 3.4.2 Attended Transfer

Attended transfer is a call transfer in which the transferring party connects the call to a third party after he confirms that the third party agrees to answer the call.

Example: A gives a call to B and B wants to attended transfer the call to C. Operation instructions are as follows:

- 1. A dials the extension number of B;
- 2. The extension of B rings, and B picks up the phone. Then A and B go into conversation;
- 3. B presses **\*2** to trigger attended transfer (at the same time, A can hear a waiting tone). Then B dials the extension number of C;
- 4. Then one of the following situations will happen:
  - **a.** If the extension of C cannot be reached because the dialing/call has timed out, C rejects the call or C is busy, B will automatically switch to the conversation with A.
  - **b.** The extension of C rings (at the same time, B can hear a ringback tone). If B hangs up the phone at this moment, A will continue to hear the waiting tone. Then if A also hangs up the phone, the extension of C will continue to ring. If C picks up the phone at this moment, the call will end directly.
  - **c.** The extension of C rings and then C picks up the phone. C and B go into conversation, and A will continue to hear a waiting tone. If it's B that hangs up the phone at this moment, C and A go into conversation. If it's C that hangs up the phone, B and A go into conversation.

## 3.5 Three-way Conference

#### When the SIP extension of XonTel Plus is the caller:

Step1. A dials the number of B and B picks up the phone, and then A and B go into conversation;

Step2. A presses the flash hook, and then dial the number of C after hearing the dialing tone.

Step3. C pick up the phone, and A and C go into conversation and meanwhile the call between A and B is kept holding.

Step4. Then, if A presses the flash hook and dials 1, the conversation will switch back to A and B; if A presses the flash hook and dial 2, the conversation will switch to A and C; if A presses the flash hook and dial 3, the conversation will switch to A, B and C (three-party conversation).

#### When the SIP extension of XonTel Plus is the callee:

Step1. B places a call to A, and A picks up the phone after the phone rings. And then C also gives a call to A (at the same time, A can hear a waiting tone).

Step2. If A presses the flash hook, A and C go into conversation and meanwhile the call between A and B is kept holding.

After that, if A dials 1, the conversation will switch back to A and B; if A dial 2, the conversation will switch to A and C; if A dials 3, the conversation will switch to A, B and C (three-party conversation).

Step2 (optional). When C is calling A and B hands up the phone during the process, A and C will automatically go into conversation.







## 3.6 Switching Between Two Calls

#### When the SIP extension of XonTel Plus is the caller:

Step1. A dials the number of B and B picks up the phone, and then A and B go into conversation;

Step2. A presses the flash hook, and then dial the number of C after hearing the dialing tone.

Step3. C pick up the phone, and A and C go into conversation and meanwhile the call between A and B is kept holding.

Step4. If A presses the flash hook again, and the call will be switched back to A and B. If A presses the flash hook once more, the call will be switched to A and C.

#### When the SIP extension of XonTel Plus is the callee:

Step1. B places a call to A, and A picks up the phone after the phone rings. And then C also gives a call to A (at the same time, A can hear a waiting tone).

Step2. If A presses the flash hook, A and C go into conversation and meanwhile the call between A and B is kept holding.

After that, if A presses the flash hook again, and the call will be switched back to A and B. If A presses the flash hook once more, the call will be switched to A and C.

## 3.7 Send or Receive Fax

#### 3.7.1 Fax Mode Supported

- T.38 (IP-based)
- T.30 (Pass-Through)

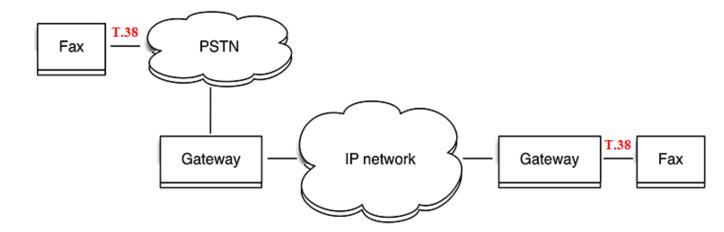






#### 3.7.2 Explanation of T.38 and Pass-through

T.38 is an ITU recommendation for allowing transmission of fax over IP networks in real time. Under the T.38 mode, analog fax signal is converted into digital signal and fax signal tone is restored according to the signal of peer device. Under the T.38 mode, fax traffic is carried in T.38 packages.



Pass-through: Under the pass-through mode, fax signal is not converted and fax traffic is carried in RTP packets. It uses the G.711 A or G711U codec in order to reduce the damage to fax signal.

## **3.8 Function of RST Button**

Press the RST button for different time length, and the XonTel Plus device will execute different function:

1. On the condition that the device is running normally, press the RST button for 3 to 6 seconds, the login password of the device will be restored to the factory default, and the network mode will become the route mode, with WAN address obtained through DHCP and LAN IP address defaulted as 192.16.11.1. At the meanwhile, the access ports of Http, Https, Telnet and SSH are restored to the default settings.

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Network / Access Control	
Web Server	
нттр	
Enable	.∞
HTTP Port	80
Allow WAN access	
HTTPS Port	443
Allow WAN access	
Telnet	
Enable	✓
Port	23
Allow WAN access	
SSH	
Enable	
Port	22
Allow WAN access	
	Cancel Save Reset
	Canter Save Reset

2. On the condition that the device is running normally, press the RST button for 6 to 12 seconds, and all configurations are restored to the default settings.

3. On the condition that the device is powered off, press the RST button and the WPS button, and connect the XonTel Plus PBX with power source. After about 30 seconds, the device will wipe out all configurations, rebuild a file system and then re-load a firmware version (this method is used in case of version fault).







## 3.9 Query IP Address and Restore Default Setting

Connect a PSTN line to the FXO port. Use a mobile phone to dial the number of the FXO port, and you can dial \*158 to query the IP address of LAN port and dial \*159 to query the IP address of WAN port.

If you want to restore XonTel Plus to default settings, you can press the **RST** button for 6 to 12 seconds or you can configure it on the Web interface.

On the Web interface, click **System**  $\rightarrow$  **Backup/Restore/Upgrade** and then select the parts (system, network or service) that need to be restored to default settings. Click **Reset** and then restart the device, and the selected parts will be restored to default settings.

System / Backup/Restore/Upgrade				
Upgrade Backup/Restore				
Choose backup files and download	System 🗹 Network 🗹 Service	Download		
Reset to defaults	System Detwork Service	Reset		
Restore from the backup	Choose File No file chosen	Restore		







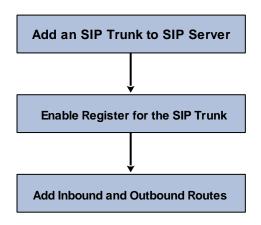
## **4** Configuration Wizard

## 4.1 Configuration Wizard

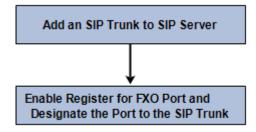
The following are the common ways to configure the XonTel Plus PBX.

### 4.1.1 XonTel Plus Regarded as Terminal and Registered to SIP Server

1. XonTel Plus Registered to SIP Server



2. FXO Port Registered to SIP Server



Note: Although 'Register' has been enabled for FXO port, calls through FXO port will take inbound and outbound routes as first priority. For outgoing calls, if outbound route cannot be matched, then the registered SIP trunk will be selected. For incoming calls, if inbound route cannot be matched, then the registered FXO port will be selected.

Generally, local extension number is taken as first priority for call routing selection, followed by DID, route and then registered port.

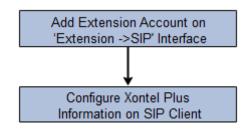




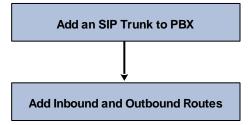


#### 4.1.2 Other SIP Clients registered to XonTel Plus

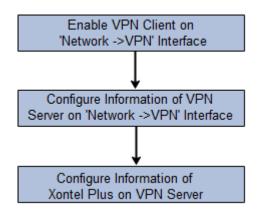
Under this mode, XonTel Plus is regarded as a SIP Server. Create an extension account first on the Extension  $\rightarrow$  SIP interface, and configure listening port on the Profile -> SIP interface. Then, configure the IP address, extension account and listening port of XonTel Plus on SIP client.



#### 4.1.3 XonTel Plus Connected to PBX through Trunking



4.1.4 XonTel Plus Serving as VPN Client





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## **5** Configurations on Web Interface

## 5.1 Introduction to Web Interface

Modify the IP address of PC to make it at the same network segment with that of LAN port of the XonTel Plus PBX (the default IP of LAN port is **192.168.11.1**).

Open a web browser on the PC and then enter the IP address of LAN port. Click **Login**, and the login GUI is displayed. Both the default username and password are admin.

The displayed login GUI is shown as follows:

		2		3	4 5	6
ontel Status			xtension Trunk	Call Control -	Auto Refresh	on Administrator : admin
This device is still th >>Go to password co		Please chan <u>o</u>	ge the default pa		e web interface !	
		Please chang	ge the default pa	ssword to protect th Performance CPU		0 (4%)
>>Go to password c	onfiguration		ge the default pa	Performance CPU	4/10	D (4%)
>So to password control of the password of	Xontel Plus		ge the default pa	Performance CPU Filesystem	4 / 100	984 kB (73%)
>>Go to password c System Device Model Device SN	onfiguration Xontel Plus DD59-A210-0001-	-0107		Performance CPU	4 / 100	
>>Go to password control of the second co	onfiguration Xontel Plus DD59-A210-0001- 4834-1327-392F	-0107 4-10 19:44:55		Performance CPU Filesystem	4 / 100	984 kB (73%)
So to password control of the password of t	Xontel Plus DD59-A210-0001- 4834-1327-392F 2.55.1.19 2020-04	-0107 4-10 19:44:55		Performance CPU Filesystem	4 / 100	984 kB (73%)

Index	Item	Description
1	XonTel	The name of the PBX; it can be edited on the <b>System</b> $\rightarrow$ <b>Setting</b> interface
2	Menu Bar	The menu bar of XonTel Plus
3	Password Change Reminder Or Unsaved Changes	If your password remains the default one, you will be advised to modify it. All changes to the configuration of the PBX need to be saved. Click <b>Apply</b> to enter into the page to save the changes; click <b>Revert</b> to return to original configuration.
4	Detailed Interface	The detailed configuration interface or display interface

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5	Auto Refresh Button	The button can be enabled or disabled. If it is enabled, the information on the <b>Status → Overview/SIP/PSTN/Current Call</b> interfaces will be refreshed automatically
6	User Role	The role of the current user logging into the Web. And the "exit" sign will pop up when the mouse moves over there. You can log out of the web from there

## 5.2 Status

The '**Status**' menu mainly displays all kinds of status information. It includes the following sub-menus: Overview, SIP, PSTN, DHCP Client List, Fail2ban, VPN, Parking Lot, Current Call, Call Queue, CDRs, Service, performance and About.

### 5.2.1 Overview

Log in the Web interface of XonTel Plus, click **Status**  $\rightarrow$  **Overview**, and the following interface will be displayed. On the interface, device model, firmware version as well as information about performance are shown, together with WAN network, LAN network and DHCP server.

System Device Model Device SN Hardware ID Firmware Version Local Time Uptime Cloud Server	Xontel Plus B901-0500-1018-0308 2C37-CB30-2734 2.55.1.23 2020-08-13 11:28:29 CST +0800 2020-08-22 00:18:37 1 d 0 h 18 m 13 s Disabled	Performance CPU Filesystem Memory	3 / 100 (3%) 15572 kB / 17964 kB (66%) 64924 kB / 212716 kB (30%)
WAN Network MAC Address Type IP Address Netmask Gateway Prefered DNS server Alternate DNS server RX / TX (Per Second) RX / TX (Total)		LAN Network MAC Address Type IP Address Netmask RX / TX (Per Second) RX / TX (Total)	D4-67-61-B9-07-C5 Static 10.108.122.2 255.255.255.0 0 Bytes (0 Pkts.) / 0 Bytes (0 Pkts.) 7.22 MB (24153 Pkts.) / 5.09 MB (19672 Pkts.)
DHCP Server Status Start Address End Address Gateway Expires Prefered DNS server Alternate DNS server	Disabled - - - -		

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## 5.2.2 SIP

Click Status → SIP, and the following interface will be displayed. On the interface, information of SIP profile, SIP Trunk and SIP extension is shown.

Status / SIP	1								
SIP Extension	SIP Trunk	SIP Profile							
Filter by Status	🗷 Register 🗹 Unre	egistered							
Index ‡ Na	me ‡ Extension	t Online t	Register Source	e ‡	Status ‡	Expires ‡	Agent ‡	Profi	le ‡
Status / SIP									
SIP Extension	SIP Trunk	SIP Profile							
Index I	Name	Address	Transport	Reg	Heartbeat	Status	Call In(F/T)	Call Out(F/T)	Profile
Status / SIP									
SIP Extension	SIP Trunk	SIP Profile							
Index	Name	Liste	ening Addr		State	Current Call	Call In	I(F/T)	Call Out(F/T)
1	lan_default	192.16	68.11.1:5060	F	RUNNING	0	0/	D	0/0

Belong To	Parameter	Explanation
	Filter by Status	You can choose Register or Unregister to filter SIP extensions
	Profile	The profile that is used by the SIP extension
SIP Extension	Status	SIP extension is registered or not. There are two statuses: Registered/Unregistered
	Heartbeat	If heartbeat is enabled, option message will be sent to peer device (the peer device is reachable)
SIP Trunk	Status	Green color means available, while red color means abnormal, unavailable or prohibited.
		There are five statuses: Running, Reged/Up, Noreg/Up, Trying-Down, Fail-Wait
	Profile	The profile that is used by the SIP trunk
	Name	The name of the SIP profile
	Listening Address	The current listening address and port of SIP
Profile	State	Green color means normal running, while red color means listening address and port of SIP is unavailable. There are two states: <b>Running and Down</b>

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### 5.2.3 **PSTN**

On the Status  $\rightarrow$  PSTN interface, information of FXO is shown. Green color means available or registered, while red color means abnormal, unregistered or prohibited.

FXO					
Port Mo	odule State	Parameter Status	SIP Register Status	Hook State	Line State
0	READY	ОК	Not Config	ONHOOK	OFFLINE
1	READY	ОК	Not Config	ONHOOK	OFFLINE

If 'SIP Register Status' is 'Registered', it means FXO has been registered to SIP server on the Trunk  $\rightarrow$  SIP/FXO interface respectively.

Belong To	Parameter	Explanation
	Module Status	There are two module statuses: Ready and Config Failed
	Parameter Status	There are two parameter statuses: OK and error
FXO	SIP Register Status	There are two SIP register statuses: Registered and Unregistered
	Hook State	There are two hook states: Onhook and Offhook
	Line State	There are two hook states: Online and Offline

## 5.2.4 DHCP Client List

XonTel Plus has a built-in DHCP server. When the DHCP server is enabled, it can assign IP addresses to the clients connected to it.

On the Status → DHCP Client List interface, information of DHCP clients connected to the XonTel Plus PBX, such as client name, Mac address and IP address, is shown.

Status / D	HCP Client List				
ID	Client Name	MAC Address	IP Address	Expiration	Status
1	GJFdeiphone	6C:8D:C1:05:A5:EE	192.168.11.173	2016-09-12 19:49:46	Online







#### 5.2.5 Fail2ban

On the **Status**  $\rightarrow$  **Fail2ban** interface, you can see currently-banned IP addresses t and historic banned IP addresses. You can also unban those IP addressed that have been blocked before.

Fail2ban is a log-parsing application that monitors system logs for symptoms of an automated attack on your device. When an attempted compromise is located, using the defined parameters, Fail2ban will add a new rule to block the IP address of the attacker, either for a set amount of time or permanently. Fail2ban can also alert you through email that an attack is occurring.

Statu	ıs / Fail2ban					
Curren	nt Ban List					
Inde	x IP	Ban time	Release time	Туре	Action	
Operat	tion History List					
Operat	tion History List IP	Common Ban Duration	Туре	Action	Operation time	Filter
			Type SSH	Action Ban	Operation time 2020/05/09 08:39:52	Filter

For the explanation of parameters related to fail2ban, please refer to the "Network ->Fail2ban" section.

#### 5.2.6 VPN

On the Status  $\rightarrow$  VPN interface, the online records and historical records of XonTel Plus as a L2TP client, a PPTP client, SSTP client and an OpenVPN client are displayed.

Meanwhile, the XonTel Plus PBX can also serve as a VPN server, such as L2TP server, PPTP server and OpenVPN server. Related online records and historical records are shown on the Status  $\rightarrow$  VPN ->OpenVPN Server or Status -> VPN -> L2TP/PPTP Server Access List interface.

OpenVF	N Client	OpenVPN Server	L2TP Client	PPTP Client	L2TP/PPTP Server Acce	ess List SSTP	GRE	
Online	Record							
Index	Protocol	IP Address	Gateway	Server Address	RX / TX Bytes	Login Time	Connection Tir	me
				This section contai	ins no records yet			
History	Records							
Index	Protocol	IP Address	Gateway	Server Address	RX / TX Bytes	Login Time	Connection Time	Filter
				This section contai				

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## 5.2.7 Current Call

On Status  $\rightarrow$  Current Call interface, the source, destination, calling number, called number, start time, answer time, state and duration of the current real-time call are shown. If there is no current call, no information will be shown

Status / Cur	rent Call								
Index	Src	Dest	Caller	Called	Start Time	Answer Time	State	Duration	Filter

## 5.2.8 Call Queue

On the Status  $\rightarrow$  Call Queue interface, you can see all the call queues and specific information of each call queue.

Call Queue consists of:

- Incoming calls being placed in the queue;
- Members that answer the queue (extensions or users that log in as agents);
- A strategy for how to handle the queue, such as dividing the calls between agents;
- Waiting calls.

Status / Call Queu	e					
Name	Number	Strategy	Agents Count	Waiting Calls	Answered Calls	Total Calls

## 5.2.9 Parking Lot

You can use the parking feature to park a call, and then retrieve the call either from your phone or another phone. After you park a call, the call is placed on hold, you can continue the conversation after retrieving it.

On the **Status -> Parking Lot** interface, the numbers that are parked and the parking duration are shown.

Status / Park	ing Lot		
Index	Parking Number	Source	Duration







#### 5.2.10 CDRs

Click Status  $\rightarrow$  CDRs, and you can set query criteria to query the CDRs (Call Detailed Records) that you want on the displayed interface. Meanwhile, you are allowed to clear CDRs or export CDRs through clicking the **Empty** or **Export** button. The maximum number of CDRs that can be saved is 5000.

CDRs cannot be saved on the **Status**  $\rightarrow$  **CDRs** interface unless the CDRs function has been enabled on the **System**  $\rightarrow$  **Setting** interface.

Status / CDRs											
CDRs Query Param Start Date Caller Source		• 5 •	1 *		End Date Called Destination		2020 <b>*</b>	5 🔹 1	9 *		
Min Duration				Query	Max Duration Reset						
CDRs List									Em	pty E	Export
Index Caller S	Source	Called	Destination	Start Time No Cl	End Time DRs yet !	Duration	Hangup E	By Code	c Hangu	p Cause	Filter

#### 5.2.11 Service

Click Status  $\rightarrow$  Service, and the service status of XonTel Plus is displayed. This function is enabled by default. The Web, SSH and Telnet service can be disabled and their ports can be modified on the Network  $\rightarrow$  Access Control interface. If no running status is shown, it means exception has occurred on the XonTel Plus device.

Besides, if syslog is disabled on the System  $\rightarrow$  Setting interface, the logs cannot be uploaded to the server, but log service is still running.

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Status / Service	
Running Status	
Msg Service	Running
Switch Kernel Service	Running
Log Service	Running
Upgrade Service	Running
Web	Running
SSH	Running
Telnet	Running
Remote Proxy	Stopped
NATS Server	Stopped

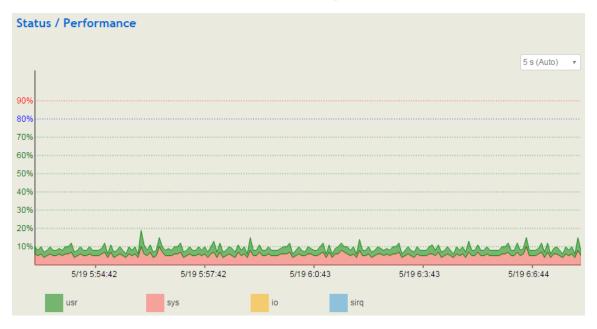






### 5.2.12 Performance

On the **Status ->Performance** Interface, you can see the performance statistics of the system.



### 5.2.13 About

On the Status  $\rightarrow$  About page, the device model, device SN, hardware ID, MAC address, boot image, root image and firmware Version of the XonTel Plus are displayed.

Status / About	
System	
Device Model	Xontel Plus
Device SN	B901-0500-1018-0114
Hardware ID	2C37-CB2F-202D
MAC Address	D4-67-61-B9-04-BD
Boot Image	10
Root Image	14
Firmware Version	2.55.1.19 2020-04-10 19:44:55 CST +0800







## 5.3 System

Configurations for hostname, time zone, NTP, login username & password, other user name, provision, TR069, operation log, service log, upgrade/backup/restore, IVR upload, Command Line, cloud server, API, event report, scheduled task, FTP server, disk manager and reboot can be carried out in the System section.

## 5.3.1 Setting

On the **System**  $\rightarrow$  Setting interface, you can modify the device name, set a new time zone, synchronize local time and enable CDRs, Syslog as well as built-in NTP server.

System / Setting		
General		
lostname	Xontel	
nguage	English	
ezone	Asia/Kuwait 🗸	
al Time	2021-09-09 11:45:12 Sync with brow	ser
Format	YYYY-MM-DD 🗸	
S	Enable ~	
ver Prompt	Disable	
g		
s rvice Log Level	Notice	
ble Syslog		
me Synchronization		
able builtin NTP server		
P server candidates	0.pool.ntp.org	
	1.pool.ntp.org	
	2.pool.ntp.org	
able builtin NTP server	0.pool.ntp.org	
	www.xontel.com	





Parameter	Explanation
Hostname	The name of the PBX. After it is configured, the name will be displayed on the left of the menu bar.
Language	You can choose the language of XonTel Plus, the default value is English
Timezone	You can choose a time zone you want. The default value is Asia/Kuwait
Local Time	The current time based on current time zone. It is synchronized with NTP.
CDRs	If it is enabled, CDRs will be saved automatically. 5000 CDRs call be saved at most and they can be queried on the <b>Status → CDRs</b> interface. If it is disabled, CDRs will not be saved
Service Log Level	There are eight levels, including Debug, Info, Notify, Warning, Error, Critical, Alert and Emergency
Enable Syslog	Whether to enable syslog
Time Synchronization	If NTP server is enabled, the XonTel Plus can be synchronized with the world standard time. Meanwhile, you're able to add or reduce NTP servers. Please consult local telecom operators or surf the internet for the address of NTP servers.
8	Delete a NTP Server
٠	Add a NTP Server







#### 5.3.2 User Manager

Click System  $\rightarrow$  User Manager, and you can modify the username name and password for logging in the XonTel Plus PBX. Factory defaults for username name and password are admin and XonTel respectively, so it is advised to modify them for security consideration.

The abovementioned username and password are also used to log in Web Interface, Telnet and SSH.

The super administrator of the device can add different users to the device and assign different roles for them, like observer, operator and administrator. Different roles can be allocated with different permissions to the functions.

System / User Manager					
Modify Password Current Username Old Password New Password Confirm New Password	adn	in Save			
Other User Manager					
Username Us	ser Group	Expiration	Description	Status	
	This :	section contains no values yet			New
System / User Manager /	New User				
Name					
User Group		Administrator	Ŧ		
New Password					
Confirm New Password					
Expiration		2030 • 5 • 19	•		
Description					
Status		Enable	*		
Web Access Permission					
Status		□ View			
System		□ View			
Network		View			
Profile		View			
Extension		□ View			
Trunk Call Control		View			
Can control		Cancel Save Re	eset		



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Parameter	Explanation
Name	The name of the new user. After it is established, the name and the password will be used to log into the web page of the device.
User Group	You can choose a role for the new user, such as administrator, operator and observer. The default value is administrator.
New Password	Setting the login password for the new user. The password needs to consist of 8 to 32 characters.
Expiration	The expiry date when the user cannot log in the device any more.
Status	Choose enable or disable.
Web Access Permission	The permissions to view status, system, network, profile, extension, trunk and call control.

### 5.3.3 Provision

Provision is used to make XonTel Plus automatically upgrade with the latest firmware stored on an HTTP server, an FTP server or a TFTP server.

As for how to configure XonTel Plus and HTTP/FTP/TFTP server for Provision, please make reference to the instruction guide of Provision.

Select the checkbox on the right of **Enable**, and you will see the following interface:

System / Provision	
Enable	×
Periodic Check	On r
Check Interval(s)	3600
URL	ftp://172.16.77.200/home
Username	star
Password	······ •
Proxy Address	
Username	
Password	•
	Cancel Save Reset







Parameter	Explanation
Periodic Check	Whether to enable periodic check. If it is enabled, the PBX will automatically check whether the firmware version stored on the URL is updated.
Check Interval	The interval to check whether the firmware version stored on the URL is updated. If it is 3600s, the PBX will check every 3600s.
URL	The URL of the HTTP/FTP/TFTP server:
	For example:
	ftp://172.16.77.200/home
	tftp://172.16.77.200/provision.xml
	http://test.domain.com/test
Username	The login username of the HTTP/FTP/TFTP server
Password	The login password of the HTTP/FTP/TFTP server

Note: Proxy Address, Proxy Username and Proxy Password are optional to be configured.

## 5.3.4 Operation Log

The logs tracing the operations carried out on the Web can be queried on the System → Operation Log interface. You are allowed to set query criteria to query the logs that you want and to export the logs through clicking the Export button at the top-right corner.

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Systen	n / Operation Lo	g				Export
Only late	st 100 records provide	ed to show,	if want to see more, you	can export it !		
Index	Time	Level	Access Source	Operation	Page	Filter
100	2020-05-19 Tue 06:18:59	Info	172.19.1.11:54156	View	system/provision	
99	2020-05-19 Tue 06:17:01	Info	172.19.1.11:54128	Add New Config	system/security/user/add	
98	2020-05-19 Tue 06:15:46	Info	172.19.1.11:54103	View	system/security	
97	2020-05-19 Tue 06:10:24	Info	172.19.1.11:54034	View	system/setting	
96	2020-05-19 Tue 06:09:18	Info	172.19.1.11:54013	View	status/about	
95	2020-05-19 Tue 06:08:03	Info	172.19.1.11:53992	View	status/performance	
94	2020-05-19 Tue 06:06:20	Info	172.19.1.11:53975	View	status/service	
93	2020-05-19 Tue 05:58:59	Info	172.19.1.11:53914	View	status/cdr	
92	2020-05-19 Tue 05:58:31	Info	172.19.1.11:53871	View	status/parking	
91	2020-05-19 Tue 05:58:28	Info	172.19.1.11:53871	View	status/callqueues/callqueues_info	)
90	2020-05-19 Tue 05:58:26	Info	172.19.1.11:53871	View	status/callqueues/callqueues_info	)

Note: Operation logs are generally used to locate faults by device manufacturer.







## 5.3.5 Service Log

Service logs (the running logs of XonTel Plus) can be exported on the **System**  $\rightarrow$  Service Log interface. Those logs are used for analyzing where a problem has occurred on the PBX.

System / Serv	vice Log		
Export			

## 5.3.6 Config Changes Log

On the System  $\rightarrow$  Config Changes Log interface, the configurations changed by administrator on the Web of the PBX are recorded.

System / Config Changes Log	Export
Mon May 18 02:35:36 2020	A
SSIP / Edit	
Password = xionghuaiwei Server Address = 172.28.88.187	
Fri May 15 09:26:46 2020	
SSTP / Edit	
Server Address = 13.250.48.100	
Fri May 15 09:24:28 2020	
SSTP / Edit	
	<b>*</b>

### 5.3.7 Backup/Restore/Upgrade

On the **System**  $\rightarrow$  **Backup/Restore/Upgrade** interface, you can back up or restore configuration data, and can upgrade XonTel Plus to a new version. But you need to restart the device for the change to take effect after executing restore or upgrade.

#### **Upgrade the Device**

System / Backup/Restore/Upgrade			
Upgrade Backup/Restore			
Please Select Upgrade Type	System •		
	Choose File No file chosen Upgrade		

Note: the file you choose to be upgraded on the above interface is a local file, while the version file upgraded through the Provision function is a file from HTTP/FTP/TFTP server.







System / Backup/Restore/Upgrade							
Upgrade Back	up/Restore						
Choose backup files Reset to defaults	and download	<ul> <li>✓ System </li> <li>✓ Network </li> <li>✓ System </li> <li>Network </li> <li>✓ Service</li> </ul>	Download Reset				
Restore from the ba	ckup	Choose File No file chosen	Restore				
	Restore to History Backup						
Index	User	Backup Time 2020-05-18 02:35:54					
1	admin	2020-05-18 02:35:54	() () () () () () () () () () () () () (				
2	admin	2020-05-15 09:27:04					
-							
4	admin	2020-05-14 13:43:45	() () () () () () () () () () () () () (				
5	admin	2020-05-14 13:43:00					
6	admin	2020-05-14 13:26:48	$\bigcirc \bigcirc \bigotimes$				
7	admin	2020-05-14 10:59:45	$\bigcirc \bigcirc \bigcirc \bigcirc$				
8	admin	2020-05-14 10:52:41	$(i) \supseteq \otimes$				

### Explanation of Backup/Restore/Upgrade menu

admin

admin

9

10

Upgrade	Choose a file to be upgraded (which is provided by XonTel), and then click Upgrade.
Download	You can download the configuration data to be backed up. Select any of the checkboxes on the left of System, Network and Service, and then click <b>Download</b>
Reset	Select any of the checkboxes on the left of System, Network and Service, and then click <b>Reset</b> , and configurations related to the selected part will be restored to factory defaults.
Restore	Choose a backup file, and then click <b>Restore</b> .

2020-05-14 10:52:09

2020-05-12 03:04:22



 $\bigcirc \bigcirc \bigotimes$ 

 $\bigcirc \bigcirc \bigcirc \bigcirc$ 





## 5.3.8 Voice

On the System  $\rightarrow$  Voice  $\rightarrow$  Voice you can upload an IVR file according to your needs. At present, only wav audio file is allowed. The format of the uploaded wav audio file must be: monaural, 8000hz, 16bit, and size of no more than 1M.

System / Voice				
Voice Voice Recor	rd			
Туре	Name	Description	Storage Location	Operation
Waiting Music	default waiting music	Default waiting/hold music, will play repleatly	Local	•
IVR	default ivr	Default IVR welcome audio	Local	•
Waiting Music 🗸			Local 🗸	Choose File Nosen Upload
The format of wav audio	file should be monaural, 8000hz	z, 16bit, and a size of no more than 1MB.		

Also on System  $\rightarrow$  Voice  $\rightarrow$  Voice Record you can record the IVR from your extension, just select the extension that

you want to record on it then click on	Start Record	to start record the IVR		
--	--------------	-------------------------	--	--

System / Voice					
Voice Voice Record					
Select Extension	SIP Extension / 301 🗸				
Туре	IVR 🗸				
Name	test				
Description	test				
Recording Storage Location	Udisk V Start Record	d			

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### 5.3.9 Command Line

On the System → Command Line interface, some commonly-used command lines can be directly selected in the draw-down box, and therefore user has no need to enter command lines on Telnet. In this way, the efficiency of problem diagnostics is greatly improved.

Commonly-used command lines include fxo config, fxo status, gsm status, gsm bcch, gsm oper, sip status, sip profile and so on.

## System / Command Line

<b>•</b>	Execute	Save	Empty
· · · · · · · · · · · · · · · · · · ·			

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## 5.3.10 Cloud Service

Cloud service is mainly used to centrally manage all kinds of devices. Through cloud service, you can query the status of a device, upgrade devices at batch, log in or configure a device remotely. The XonTel Plus PBX provides Cloud service. Enter the IP address, service port and password of the Cloud server, and then the PBX will connect to the cloud server.

System / Cloud Service					
NMS Remote Proxy NATS Server					
Status	Enable ~				
Request method	HTTPS ~				
Server Address	185.16.18.200				
Server Port	20006				
Interface	WAN ~				
Protocol version number	1.0 ~				
	Cancel Save Reset				

System / Cloud Service	
NMS Remote Proxy NATS Server	
Status	Enable
Server Address	server02.dmcld.com
Server Port	3100
Password	•
	Cancel Save Reset
	www.xontel.com
nTel	Kuwait         KSA           Tel.: 1880005         Fax: 22413877





#### **NATS Server:**

XonTel Plus can work as a NATS client to send messages to a NATS server, and then the NATS server will open related ports to facilitate the connection with those clients or servers of users.

System / Cloud Service					
NMS Remote Proxy	NATS Server				
Status		Disable	~		
Server Address					
Username					
Password					
Heartbeat		Disable	~		
TLS Verification		Disable	~		
TLS Skip Server Verification		Disable	~		
Server Certificate		Choose File No file chosen			
Client Certificate		Choose File No file chosen			
Client Key		Choose File No file chosen			
		Save Reset			







## 5.3.11 **API**

XonTel Plus provides API (Application Programming Interface) to interwork with other devices or platforms. This function enables you to centrally manage devices through command lines.

System / API		
Status	Enable •	
Password	•••	•
	Cancel Save Reset	

## 5.3.12 Event Report

XonTel Plus allows the following events to be reported through NATS: device startup, call status, registering or unregistering of SIP extensions, availability or unavailability of SIP trunks, FXO status and update of CDR information. For event report through NATS, please refer to the configuration steps of NATS in the Could Server section.

System / Event F	Report	
System SIP FX	(O Recordin	g Log
Event Type		
SIP Extension Register/	Unregister (	
SIP Trunk Available/Una	available (	
APP Notification	t t	2
URL Info		http://pnxonpbx.xontel.com/pnxontel/pn.php?token=\${pn-tok}&caller=\${callee=\${callee}&callid=\${callee}
Register Ti	meout(s)	5
Parameter	List	<pre>\${pn-tok} : pn-tok \${caller} : Caller Number \${callee} : Destination Number \${callid} : Call-ID</pre>
		Save Reset







## 5.3.13 Schedule Task

On the **System**  $\rightarrow$  **Schedule Task** interface, you can set a scheduled time to reboot the XonTel Plus device, record backup, and back up CDRs, configuration backup or backup logs as shown in the figures below.

You can also make schedule backup to an http server (You can make your PC as http server by using http server application such as **hfs** software application).

System	/ Schedule Ta	ask			
Reboot	Record Backup	CDR Backup	Config Ba	ackup	Log Backup
Status Interval				Enable 7	v Day
Execution	n Time			0	▼ Hour 0 ▼ Min
				S	Save Reset

System / Schedule Task	
Reboot Record Backup CDR	Backup Config Backup Log Backup
Status	Enable
Interval	30 🗸 Day
Execution Time	0 V Hour 0 V Min
Backup to Server	
URL Info	http://192.168.8.169/recording
Max Retry	5 🗸
Delete After Backup	
	Save Reset

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System	/ Schedule T	ask					
Reboot	Record Backup	CDR Backup	Config Backup	Log Back	up		
Status			Enable	•			
Interval			1	✓ Day			
Executio	n Time		0	✓ Hour	0	✓ Min	
Backup	Туре		All			*	
CDR For	rmat		Sqlite				
Local Ba	ckup						
5	Storage Location		Udisk		`	·	
Backup t	o Server						
ι	JRL Info		http://1	92.168.8.16	9/cdrback		
C	Compress File						
				Save	Reset		

System / Schedule Task	
Reboot Record Backup CDR Backup	Config Backup Log Backup
Status	Enable 🗸
Interval	1 V Day
Execution Time	0 V Hour 0 V Min
Local Backup	
Storage Location	Udisk 🗸
Backup to Server	
URL Info	http://192.168.8.169/backupconfig
	Save Reset

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System / Schedule Task	
Reboot Record Backup CDR Backup	Config Backup
Status	Enable 🗸
Interval Execution Time	1 V Day 0 V Hour 0 V Min
Local Backup Storage Location	✓ Udisk ✓
Backup to Server	✓ http://192.168.8.169/logbackup
	Save Reset



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## 5.3.14 Email

On the **System**  $\rightarrow$  **Email** interface, you can configure an email client on XonTel Plus, which can be used to send or receive emails. The email client can also is used to test connection. But on top of that, SMTP, IMAP and POP 3 services need to be enabled for the email client.

System / Email	
Configuration Log	
Status	Enable v
Username	admin123
Password	••••••
	Connect Test Send Receive
Send(SMTP)	
Server Address	
Port	465
TLS Enable	
Email Address	
Receive	
Protocol	IMAP v
Server Address	
Port	993
TLS Enable	
Folder	INBOX
Message Query Interval(min)	5
Message Valid Time Range	Within 5 minutes
Numbers of Message Per Receive	5 💌







Username	Enter the address of email client
password	The password or authorization code of the email client
Server Address	The Address of the SMTP server, supported by the email client
Protocol	Choose IMAP or POP3. When POPS is selected, TLS port is 993 by default.
Message Query Interval (min)	The time interval to check whether there is a new email.
Message Valid Time Range	Only those emails received during this time range are addressed.
Number of Message Per Receive	The maximum number of emails that are received at one time. If the number exceeds, they will be received in batches.

On the System  $\rightarrow$  Email $\rightarrow$  Log interface, you can check Email logs as shown below.

System / Email		
Configuration Log		
log is empty.		

## 5.3.15 FTP Server

On the System **>** FTP Server interface, you can enable the FTP server function of XonTel Plus and configure related parameters such as username, password and access permissions. You can connect FTP clients to this FTP server and then access those files (like recording files and system logs) that are open on the XonTel Plus device through the 21 port.

Enable	Ŧ
admin123	
•••••	0
Disable	Ŧ
Disable	٣
Enable	Ŧ
Cancel Save Reset	
	admin123  Disable Disable Enable

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On the System  $\rightarrow$  FTP $\rightarrow$  Log interface, you can check FTP logs as shown below.

System / FTP Server		
FTP Server Log		
FTP Log		Export
Sat May 16 13:16:50 2020 [pid 8565] Sat May 16 13:16:50 2020 [pid 8565] Sat May 16 13:16:50 2020 [pid 8565]	<pre>[admin] FTP response: Client "103.145.12.13", "530 Permission denied." CONNECT: Client "103.145.12.13" FTP response: Client "103.145.12.13", "220 (vsFTPd 3.0.2)" FTP command: Client "103.145.12.13", "USER admin" [admin] FTP response: Client "103.145.12.13", "530 Permission denied."</pre>	Â

# 5.3.16 Disk Manager

On the **System**  $\rightarrow$  **Disk Manager** interface, you can see the memory usage of USB and SD card. USB memory are divided into three categories, including voicemail (40%), recording (50%) and Others(10%). You can also divide the proportion of each category, disconnect the USB or execute formatting on this interface.

System / Disk	Manager						
Recording	Voicemail	Others					
USB (15360M)					Re	size Remove	Format Disk
Recording : 7680 MB / (50%)				Voicemail : 3225.6 MB / (21%)		Others : 4454.4 MB / (29%)	
SD Card							
			SD Card	Not Found			

Note: XonTel Plus only supports USB of FAT and EXT4.

## 5.3.17 **Reboot**

On the System  $\rightarrow$  Reboot interface, you can click **Perform Reboot** to reboot the XonTel Plus PBX. After the device is rebooted, those configurations that have been saved will remain unchanged.









# 5.4 Network

XonTel Plus works in two modes: route mode and bridge mode. When it is under the route mode, the IP of WAN must be different from the IP of LAN. But when it is under the bridge mode, the IP of WAN and the IP of LAN are the same.

# 5.4.1 Setting

On the **Network**  $\rightarrow$  **Setting** interface, you can set the IP address of WAN port and LAN port.

Under the route mode, the default IP address of WAN port is a DHCP IP address, while the default IP address of the LAN port is **192.168.11.1**.

In fact, there are three kinds of IP addresses for selection for WAN port and LAN port, including Static IP address, DHCP and PPPOE.

### DHCP: Obtain IP address automatically.

XonTel Plus is regarded as a DHCP client, which sends a broadcast request and looks for a DHCP server to answer. Then the DHCP server automatically assigns an IP address to the XonTel Plus from a defined range of numbers.

Ne	twork / Setting		
Netv	vork Model	Route	,
WAN	4		
	Protoc ol	DHCP	
	Obtain DNS server address automatically	Image: A start of the start	
	Disable Private Internets(RFC2918) DNS responses	Ø	
	MTU	1500	
LAN			
	IP Address	192.168.11.1	
	Netmask	255.255.255.0	•
	MTU	1500	
		Cancel Save Reset	

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#### Set WAN IP as DHCP IP

WAI	4		
	Protocol	DHCP	٣
	Obtain DNS server address automatically		
	Disable Private Internets(RFC2918) DNS responses		
	MTU	1500	

### Note: When WAN IP is set as DHCP IP, please ensure that there is DHCP server working normally in the network.

#### **Static IP Address:**

Static IP address is a semi-permanent IP address and remains associated with a single computer over an extended period of time. This differs from a dynamic IP address, which is assigned ad hoc at the start of each session, normally changing from one session to the next.

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If you choose static IP address, you need to fill in the following information:

- IP Address: the IP address of the WAN port of the XonTel Plus; •
- Netmask: the subnet mask of the IP address of the XonTel Plus; •
- Default Gateway: the IP address of the router connected the XonTel Plus; •
- Prefered DNS server: the IP address of the primary DNS server •
- Alternate DNS server: the IP address of the secondary DNS server
- IP address 2: the second IP address of the WAN port of the XonTel Plus; •
- Netmask 2: the subnet mask of the second IP address of the XonTel Plus; •







#### Set WAN IP as Static Address

Network / Setting				
Net	vork Model	Route		
WAI	4			
	Protocol	Static address 🗸		
	IP Address	192.168.100.200		
	Netmask	255.255.255.0 🗸		
	Default Gateway	192.168.100.1		
	Prefered DNS server	8.8.8.8		
	Alternate DNS server	192.168.100.1		
	Disable Private Internets(RFC1918) DNS responses			
	IP Address 2	192.168.5.150		
	Netmask 2	255.255.255.0		
	MTU	1500		

#### **PPPoE:**

PPPoE is an acronym for point-to-point protocol over Ethernet, which relies on two widely accepted standards: PPP and Ethernet. PPPoE is a specification for connecting the users on an Ethernet to the Internet through a common broadband medium, such as a single DSL line, wireless device or cable modem. PPPOE IP address refers to IP address assigned through the PPPoE mode.

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If you choose PPPoE, you need to fill in to fill in the following information:

- Username: the account name of PPPoE
- Password: the password of PPPoE
- Server Name: the name of the server where PPPoE is placed







#### Set WAN IP as PPPoE IP

MAN	٧		
	Protocol	PPPOE •	
	Username	admin	
	Password	•••••	Ø
	Server Name		
	PPPOE Redial		
	Obtain DNS server address automatically	<b>v</b>	
	Disable Private Internets(RFC1918) DNS responses	<b>v</b>	
	MTU	1500	



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# 5.4.2 Access Control

The access ports of Web, Telnet and SSH, as well as relevant on-off controls, can be configured on the Network  $\rightarrow$  Access Control interface.

Network / Access Control			
Web Server			
НТТР			
Enable	×		
HTTP Port	80		
Allow WAN access	×		
HTTPS Port	443		
Allow WAN access	×		
Telnet			
Enable			
Port	23		
Allow WAN access	ø		
SSH			
Enable			
Port	22		
Allow WAN access			
	Cancel Save Reset		







### 5.4.3 Firewall

If the XonTel Plus works under the route mode, you can choose to enable the firewall and set filter rules to accept or reject certain destination IP addresses.

### **Configuration Procedures:**

- 1. Select On in the drop-down box on the right of Filter Rules Control
- 2. Select filter action, accept or reject;
- 3. Click the New button;
- 4. Fill in information of filter rule;
- 5. Click the Save button to save the configuration.

Filter Rules			
Priority Name Protocol Source IP/Port/MAC Destination IP/Port	Action	Status	
13 allow-classC All 192.168.0.1/255.255.0.0/*/* */* A	Accept	Enabled	в⊘⊗√
25 allow-classA All 10.0.0.1/255.0.0.0/*/* */* A	Accept	Enabled	
26 qcall All 62.150.1/255.255.255/*/* */*	Accept	Enabled	
27 XonTel All 78.89.170.173/255.255.255/*/* */*	Accept	Enabled	
28 gulfsip All 52.58.68.25/255.255.255.255/*/* */*	Accept	Enabled	
29 ooreedo All 188.0.0.1/255.0.0.0/*/* */*	Accept	Enabled	
30 Block-SSH All */*/* */22	Drop	Enabled	
31 Block-HTTP All */*/* */80	Drop	Enabled	
32 Block-HT All */*/* */443	Drop	Enabled	

#### Note:

- 🗹 : Edit information for the corresponding filter rule.
- (8) : Delete the corresponding filter rule.
- /\*: Information of Source or Destination is not completely filled in.

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### Create Filter Rule

Network / Firewall / Filter Rules / New				
Priority	32 •			
Name	class B			
Protocol	All			
Source IP	172.16.0.1/255.255.0.0			
Source Port				
Source MAC	00:00:00:00:00			
Destination IP				
Destination Port				
Action	Accept •			
Status	Enable			
	Cancel Save Reset			

### Explanation of Parameters for Filter Rule

Name	The name of the firewall filter rule.
Protocol	Choose UDP or TCP or All (both UDP and TCP)
Source IP	The IP address that you want XonTel Plus to accept or reject. It is the IP address of a host from local-area network; it can also be a string of IP addresses, for example, 172.16.11.1/15.
Source Port	The port of the source host which the accepted or rejected IP address belongs to
Source MAC	The Mac of the host which the accepted or rejected IP address belongs to
Destination IP	The IP address that you want XonTel Plus accept or reject. It is the IP address of a host from wide-area network; it can also be a string of IP addresses, for example, 152.16.11.11/19.
Destination Port	The port of the destination host which the accepted or rejected IP address belongs to
Action	Choose accept or Drop
Status	Enable or Disable the firewall filter rule

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# 5.4.4 DHCP Server

If there is a need, you can choose to enable the built-in DHCP server of XonTel Plus to assign IP addresses to PC or other clients that are in the same local-area network with XonTel Plus. Under this condition, the XonTel Plus PBX works like a router.

Network / DHCP Server	
Status	Enable
Start Address	192.168.11.99
End Address	192.168.11.198
Leasetime(Hour)	12
Gateway	192.168.11.1
Prefered DNS server	8.8.8.8
Alternate DNS server	192.168.11.1
	Cancel Save Reset

### Explanation of Parameters for DHCP Server

Status	Enable or disable DHCP server option
Start Address	The start IP address of the DHCP pool to be assigned
End Address	The end IP address of the DHCP pool to be assigned
Lease Time(Hour)	The validity period of the assigned IP address in hours
Gateway	The gateway of the DHCP pool to be assigned, it is optional to fill in
Preferred DNS server	The primary DNS of the client whose IP address is assigned by the built-in DHCP server; it is optional to fill in
Alternate DNS server	The secondary DNS of the client whose IP address is assigned by the built-in DHCP server; it is optional to fill in

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# 5.4.5 Port Mapping

When the XonTel Plus works under the route mode, port mapping allows a client in the wide-area network to visit a client in the local-area network.

**Configuration Procedures:** 

1. Click **Network** → **Port Mapping**, and the following interface will be shown.

Network / Port Mapping							
Index	Name	WAN Port	Protocol	LAN IP	LAN Port	Status	
			This section cont	ains no values yet			
							New

- 2. Click the New button.
- 3. Fill in information on the following interface.

Network / Port Mapping / New	
Index	1 •
Name	
WAN Port	
Protocol	ТСР т
LAN IP	
LAN Port	
Status	Enable •
	Cancel Save Reset

Name	The name of this port mapping
WAN Port	The port of the client in the wide-area network, which is to visit local-area network
Protocol	Choose TCP, UDP or TCP/UDP
LAN IP	The IP address of the to-be-visited client in local-area network
LAN Port	The port of the to-be-visited client in local-area network (this port cannot conflict with the port of XonTel Plus)
Status	Choose enable or disable.

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4. Click the Save button to save the above configurations.







# 5.4.6 DMZ Setting

When the XonTel Plus PBX works under the route mode and the DMZ service is enabled, the clients in the wide-area network are allowed to have direct access to the clients in the DMZ (**demilitarized zone**).

Network / DMZ	
DMZ Status	Enabled •
DMZ IP Address	192.168.1.123
	Cancel Save Reset

# 5.4.7 Diagnostics

On the Network  $\rightarrow$  Diagnostics interface, you can use three network utilities including Ping, Traceroute and Nslookup to diagnose the network, and can capture data packages of the available network ports.

Network / Diagnostics		
Network Utilities Ping	Traceroute	Nslookup
Network Capture		
Network Interface	WAN	×
Logical Type	OR	Ψ.
Source IP		
Source Port		
Destination IP		
Destination Port		
Protocol		RP
	Start	

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**Ping** is used to examine whether a network works normally through sending test packets and calculating response time.

Instructions for using Ping:

1. Enter the IP address or domain name of a network, a website or a device in the input box of Ping, and then click Ping.

2. If related messages are received, it means the network works normally; otherwise, the network is not connected or is connected faultily.

Traceroute is used to determine a route from one IP address to another.

Instruction for using Traceroute:

- 1. Enter the IP address or domain name of a destination device in the input box of Traceroute, and then click **Traceroute**.
- 2. View the route information from the returned message.

**Nslookup** (Name Server Lookup) is a network command-line tool to obtain domain name of internet or to diagnose the problems of DNS.

Instruction for using Nslookup:

- 1. Enter a domain name and then click Nslookup.
- 2. View the DNS information from the returned message.

#### **Network Capture**

On the following interface, you can capture data packages of the available network ports. You can also set source IP, source port, destination IP or destination port to capture the packages that you want.

There is a "and"/" or "logical type. The "and" relationship can only capture a one-way message, or "or" relationship to fetch the interaction message between a particular IP.

Note: If there are multiple source or destination IP addresses, please use '|' to separate them, for example, 172.16.115.12|172.16.115.15.

After package capturing is completed, save the captured packages on a computer and then use a tool to analyze them.







## 5.4.8 **DDNS**

On the Network  $\rightarrow$  DDNS interface, you can enable DDNS (Dynamic Domain Name Service) service and set related parameters.

If DDNS (Dynamic Domain Name Server) service is enabled, when the IP address bound to a domain name changes, the new IP address will be sent to the DDNS, and thus user can visit the device via the new IP address or domain name and incoming calls can arrive the device via the domain name.

Network / DDNS		
DDNS Service	Enable	٠
Service Providers List	dyn.com	Ŧ
Domain	yourhost.dyndns.org	
Username	your_username	
Password	•••••	0
IP Source	External Address	Ŧ
IP Check URL	http://checkip.dyndns.com	Ŧ
IP Check Period(m)	10	
Force Update Interval(h)	72	
Retry Interval When Fail(s)	60	
	Cancel Save Reset	

# 5.4.9 **VPN**

VPN (Virtual Private Network) is a network technology that creates a secure remote network connection over a public network through encrypted tunnel and conversion of data's destination address. XonTel Plus can serve as a VPN client to connect with VPN server.

XonTel Plus supports the following VPN protocols:

- 1. **OpenVPN** is a kind of VPN based on the application layer of OpenSSL. It allows VPN clients to use a shared key, certificates or username/password to authenticate themselves.
- 2. Layer 2 Tunneling Protocol (L2TP) is a protocol used to package data of PPP link layer and transmit the data between two sites over the Internet through a tunnel.
- 3. **Point-To-Point Tunneling Protocol (PPTP)** is another tunneling protocol used to connect a remote client to a private server over the Internet. PPTP is an enhanced security protocol which supports VPN. And its security can be enhanced through PAP (Password Authentication Protocol) and EAP (Extensible Authentication Protocol).









4. Secure Socket Tunneling Protocol (SSTP) is a form of virtual private network (VPN) tunnel that provides a mechanism to transport PPP traffic through an SSL/TLS channel. SSL/TLS provides transport-level security with key negotiation, encryption and traffic integrity checking. The use of SSL/TLS over TCP port 443 allows SSTP to pass through virtually all firewalls and proxy servers except for authenticated web proxies

#### XonTel Can work as a VPN client as shown below:

A. XonTel Plus works as a OpenVPN client

Network / VPN						
OpenVPN L21	IP PPTP SSTP					
VPN / OpenVPN						
OpenVPN Client	OpenVPN Server	CA Certification Revocation	Log			
Config Mode Status Default Route Accept Push Route		Import from File(.ovp 🗸 Disable 🖍 Enable ✓				
Proto Device Remote Server		0 0 0				
Root Ca Certificate Client Certificate Client Key		0 0 0				
Auth Username						
Auth Password Certificate		Browse No file selected.				
		للنفنق كفنفي				

Please note that the certificate that you will upload in PBX format name must be client.ovpn.







#### B. XonTel Plus Works as L2TP Client

Network / VPN					
OpenVPN L2TP PPTP SSTP					
VPN / L2TP					
L2TP Client L2TP Server					
Status	Enable				
Default Route	Disable 🗸				
Server Address	52.17.125.56				
Username	200				
Password	•••				
	Cancel Save Reset				

Status	Whether to enable the L2TP client function (XonTel Plus works as L2TP client)
Default Route	Whether to enable default route; If default route is enabled, data are transmitted between L2TP client and L2TP server through VPN route; if it is not enabled, data are transmitted between L2TP client and L2TP server through network's outbound route.
Server Address	The server address of the L2TP server that assigns account to L2TP client
Username	The username of the account assigned by L2TP server to L2TP client
Password	The password of the account assigned by L2TP server to L2TP client

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C. XonTel Plus Works as PPTP Client

Network / VPN					
OpenVPN L2TP PPTP SSTP					
VPN / PPTP					
PPTP Client PPTP Server					
Status	Disable				
Default Route	Disable				
Data Encryption	Enable				
Server Address					
Username					
Password	•				
	Cancel Save Reset				

Status	Whether to enable the PPTP client function (XonTel Plus works as PPTP client)
Default Route	Whether to enable default route; If default route is enabled, data are transmitted between PPTP client and PPTP server through VPN route; if it is not enabled, data are transmitted between PPTP client and PPTP server through network's outbound route.
Data Encryption	Whether to encrypt data during data transmission
Server Address	The server address of the PPTP server that assigns account to PPTP client
Username	The username of the account assigned by PPTP server to PPTP client
Password	The password of the account assigned by PPTP server to PPTP client







#### D. XonTel Plus Works as SSTP Client

Network / VPN	
OpenVPN L2TP PPTP SSTP	
Status	Enable
Default Route	Disable 🗸
Server Address	sstp.xontel.net
Username	basel
Password	••••••
	Cancel Save Reset

Status	Whether to enable the SSTP client function (XonTel Plus works as SSTP client)
Default Route	Whether to enable default route;
	If default route is enabled, data are transmitted between SSTP client and SSTP server through
	VPN route; if it is not enabled, data are transmitted between SSTP client and SSTP server
	through network's outbound route.
Server Address	The IP address of the SSTP server that assigns account to SSTP client
Username	The username of the account assigned by SSTP server to SSTP client
Password	The password of the account assigned by SSTP server to SSTP client







#### XonTel Plus can work as a VPN Server as shown below:

A. XonTel Plus works as a OpenVPN Server

Network	c / VPN								
OpenVPN	OpenVPN L2TP PPTP SSTP								
VPN / Op	VPN / OpenVPN								
OpenVPN C	Client Op	enVPN Server	CA Certifi	cation Revoo	ation Log				
Server Ins	tance								
Index	Server	Device Mode	Proto	Port	Isolation	Max Clients	CA	Status	
				CA	not created				
User List									
User List Index	L	lser Name	Valid	Period		Server	Stai	tus	

Network / VI	PN							
OpenVPN L2	тр ррт	P SSTP						
VPN / OpenV	'PN							
OpenVPN Client	OpenVP	N Server CA	Certificat	ion Revocation	Log			
Index	Name	Key Size	City	Organization	Organization	Email	Status	
Index	Name	Ney 0126		s section contains i	Unit no values vet	Linai	Otatus	
								New







OpenVPN L2TP PPTP SSTP						
VPN / OpenVPN						
OpenVPN Client OpenVPN Server CA Certific	ation Revocation Log					
Certificate / New						
Index	1					
Index						
Name						
Key Size	1024					
Country	AD					
State or Province						
City						
Organization						
Organization Unit						
Email Address						
Status	Enable					
	Cancel Save Reset					

Network / VF	'n			
OpenVPN L21	P PPTP SSTP			
VPN / OpenV	PN			
OpenVPN Client	OpenVPN Server CA	Certification Revocation	Log	
Index	User Name		Valid Period	Revoke Time
Index		This section contain		

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#### B. XonTel Plus works as a L2TP Server

Network / VPN		
OpenVPN L2TP PPTP SSTP		
VPN / L2TP		
L2TP Client L2TP Server		
Status	Enable	
Start Address	192.168.11.220	
End Address	192.168.11.229	
	Save	
Index Username	Description	Status
Т	This section contains no values yet	
		New

Network / VPN				
OpenVPN L2TP PPTP SSTP				
VPN / L2TP				
L2TP Client L2TP Server				
User / New				
Index	1			
Username	200			
Password	••••			
Description L2TP Plus				
Status	Enable			
	Cancel Save Reset			

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#### C. XonTel Plus works as a PPTP Server

VPN / PPTF	0		
PPTP Client	PPTP Server		
Status		Enable Y	
Data Encryption		Enable	
Gateway		192.168.11.239	
Start Address		192.168.11.230	
End Address		192.168.11.238	
		Save	
Index	Username	Description	
		This section contains no values yet	
			New

VPN / PPTP	
PPTP Client PPTP Server	
User / New	
Index	1
Username	admin
Password	••••••••••••
Description	PPTP XonTel Plus
	Cancel Save Reset







## 5.4.10 Static Route

On the Network  $\rightarrow$  Static Route interface, you can configure static routes for the network.

Network / Static Route / New	
Index	1 *
Name	Static Route-1
Target IP	192.168.1.102
Netmask	255.255.255.0
Gateway	172.16.1.5
Interface	WAN •
Status	Enable
	Cancel Save Reset

Name	The name of the static route
Target IP	The destination host of the static route
Netmask	The netmask of the static route, default: 255.255.255.0
Gateway	The gateway address of the static route
Interface	The outbound interface of the static route, namely WAN port or LAN port
Status	The static route is enabled or disabled







### 5.4.11 Hosts

On the **Network**  $\rightarrow$  **Hosts** interface, you can add a host file. After enabling the hosts file, you can visit the corresponding host by inputting the alias or domain name of the host. The format of the hosts file is as follows: IP address host alias/domain name.

The hosts file contains the mapping relationship between IP address and hostname//domain name. And the mapping relationship allows quick and convenient access to the host.

Network / Hosts	
Status	Enable
	1 192.168.100.200 basel.xontel.net
Hosts List	
	Save Reset







## 5.4.12 Fail2ban

Fail2ban is used to scan system logs and update firewall rules to reject the IP addresses that show malicious signs (for example, too many login failures) for a specified amount of time.

On the Network → Fail2ban interface, you can configure rules for Fail2ban. For XonTel Plus, Fail2ban is generally targeted SSH and SIP.

Network / Fail2ban		
SSH		
Status		
Ban Duration(second)	600	
Max Retry Duration(second)	600	
Max Retry	5	
White List		$\oplus$
Black List		۲
CID		
SIP Status	ø	
Ban Duration(second)	600	
Max Retry Duration(second)	600	
SIP Register Max Retry	5	
SIP Invite Max Retry	20	
White List		⊕
Black List		$\oplus$
	Cancel Save Reset	

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SSH/SIP				
Ban Duration(Second)	The time period during which the IP addresses that conform to the banning rule or are in the backlist are prohibited.			
	Range: 60-315360000 seconds			
Max Retry Duration(second)	The time period during which the maximum retries have been executed and then the corresponding IP address will be banned.			
	For example, if this parameter is set as 60 seconds and the maximum number of retries is set as 10, an IP address will be banned in case that it has tried 10 times during 60 seconds. Range: 5-3600			
Max Retry	The maximum number of retries during a specific time. For example, if this parameter is set as 10 and the max retry duration is set as 60 seconds, an IP address will be banned in case that it has tried 10 times during 60 seconds. Range: 5-3600			
White List	Those IP addresses that are in the white list will not be banned by Fail2ban.			
Black List	Those IP addresses that are in the black list will not be banned by Fail2ban.			

Note: If an IP address does not receive any response after it has sent out SSH/SIP attempts, and the network is reachable, you can go to the **Status**  $\rightarrow$  **Fail2ban** interface to check whether the IP address is banned or not as shown below.

### Status / Fail2ban

#### Current Ban List

current b	an Eise				
Index	IP	Ban time	Release time	Туре	Action
0	45.143.220.95	2020/05/17 16:25:09	2028/08/03 16:25:09	SIP REGIS	858
1	185.53.88.171	2020/05/09 06:38:27	2028/07/26 06:38:27	SIP INVITE	S ≤
2	62.173.147.235	2020/05/09 20:42:40	2028/07/26 20:42:40	SIP INVITE	<28 €
3	45.143.220.62	2020/04/28 15:14:46	2028/07/15 15:14:46	SIP INVITE	S ≤
4	144.217.255.187	2020/05/20 20:07:26	2028/08/06 20:07:26	SIP INVITE	<28 €
5	45.143.220.7	2020/05/11 13:41:15	2028/07/28 13:41:15	SIP INVITE	S ≤
6	45.143.220.22	2020/04/29 03:19:57	2028/07/16 03:19:57	SIP INVITE	S ≤







# 5.5 Profile

The Profile menu includes the following sub-menus: SIP, FXO, Codec, Number, Time, Manipulation, Speed Dial, AutoCLIP, Recording and Voicemail.

# 5.5.1 **SIP**

On the **Profile**  $\rightarrow$  **SIP** interface, you can set SIP information such as listening port, which will be used in extension and trunk. Multiple SIP profiles can be configured for one XonTel Plus device, so you can choose different SIP profiles according to different needs.

Profil	e / SIP								
Index	Name	Interface	Listening Port	DTMF	Session Timeout	Codec Priority	Incodec Profile	Outcodec Profile	
1	lan_default	LAN	5060	RFC2833	Off	Remote	2-< fast >	2-< fast >	<i>(i)</i> <b>(2</b> × ()
2	wan_default	WAN	5060	RFC2833	Off	Remote	1-< default >	1-< default >	() ₫⊗
									New
Index				2					
Name				V	van_default				
Local Li	stening Interfa	ce		L.	VAN	~			
2000.12	o.ogoa					· ·			
Local Li	stening Port			5	5060				
NAT				C	Dff	~			
Progres	s Timeout(s)			E	55				
	Send Type				RFC2833	~			
Dimire	ind Type				(F02033				
RFC283	33-PT			1	01				
Detect I	nband When C	all in IVR		C	Dff	~			
Process	DTMF as Hol	d/Unhold		C	Dff	~			
DDACK					~ "				
PRACK				C	Off	~			
Session	Timer			C	Dff	~			
Extensio	on Register Lo	ck		C	Dff	~			
Trunk R	eg Num to the	Same Add	per Second	1		~			
Caller N	lumber Source			r	From: User Part	~			
Janor IV					.em. ooorr art	Ť			

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Br.				
6	Called Number Source	To: User Part	-	
	Inbound Codec Negotiation Priority	Remote	~	
	Inbound Codec Profile	1-< default >	~	
	Outbound Codec Profile	1-< default >	-	
	CNG(Comfort Noise Generator)	On	~	
	Bypass Media(SIP to SIP)	Off	~	
	Proxy Media(SIP to SIP)	Off	~	
	Detect Extension is Online	Off	~	
	Ignore ACK	Off	~	
	BLF	Off	~	
	CID Header	P-Asserted-Identity	-	
	PickUp Caller Refresh Method	Off	-	
	Allow Unknown Call	Off	~	
	Inbound Source Filter	0.0.0/0		Ð
	QoS	Off	~	
	Signal Encryption	Off	~	
	RTP Encryption	Off	~	
	User Agent	Hostname / Full Firmware Ver	-	
	Timer T1(ms)	500		
	Timer T2(ms)	4000		
	Timer T4(ms)	4000		
	Timer T1X64(ms)	32000		







Name	The name of the SIP profile			
Loool Listoping Interface	The local listening interface of this SIP profile. It can be WAN port, LAN port, Open VPN, L2TP, PPTP and SSTP.			
Local Listening Interface	If the SIP profile is used by a SIP trunk, the interface filled in here is the listening port for the SIP trunk.			
Local Listening Port	The local listening port of this SIP profile. If the SIP profile is used by a SIP trunk, the port filled in here is the listening port for the SIP trunk.			
NAT	Starting NAT can speak on different networks, including four: UPNP/NAT-PMP, IP Address, Stun, DDNS			
Progress Timeout(s)	If the parameter is set as 50 seconds, it means that the call will be considered as timeout in case that no one answers the call during 50 seconds.			
DTMF Send Type	DTMF is short for Dual Tone Multi Frequency There are three DTMF modes, including SIP Info, INBAND, RFC2833			
RFC2833-PT	RFC2833 payload coding			
Process DTMF as Hold/Unhold	By default, this parameter is off. When it is set as on, DTMF will be addressed as call hold/unhold.			
PRACK	Provisional Response Acknowledgement			
Session Timer	Session Expires: The validity period of a SIP session. When a SIP session times out, an invite message needs to be sent to refresh the session, otherwise, the session ends; It is 1800 seconds by default			
	Min Session Expires: the minimum validity period to respond to a SIP session. Session Refresh Method: re-INVITE or UPDATE			
	<b>From: User Part</b> : to obtain the caller number from the user part contained in the 'From' field.			
Caller Number Source	<b>From: Display Name</b> : to obtain the caller number from the display name contained in the 'From' field.			
	<b>To: User Part</b> : to obtain the caller number from the user part contained in the 'To' field. <b>Contact: User Part</b> : to obtain the caller number from the user part contained in the 'Contact' field.			
	<b>From: User Part</b> : to obtain the called number from the user part contained in the 'From' field.			
Called Neurlan C	<b>From: Display Name</b> : to obtain the called number from the display name contained in the 'From' field.			
Called Number Source	<b>To: User Part</b> : to obtain the called number from the user part contained in the 'To' field.			
	<b>Contact: User Part</b> : to obtain the called number from the user part contained in the 'Contact' field.			

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Inbound Codec Negotiation Priority	To take the remote device or the local device as priority for inbound codec negotiation Assume local device supports PCMA, PCMU, G.729 and G.723, while the remote device supports G.723 and G.729 If remote device is taken as codec negotiation priority, G.723 will be the codec mode, since the remote device supports G.723 and G.729 and G.723 is prior to G.729				
Inbound Codec Profile	The codec profile supported by SIP for inbound calls				
Outbound Codec Profile	The codec profile supported by SIP for outbound calls				
Bypass Media(SIP to SIP)	Whether to allow SIP to communicate with the server directly				
Detect Extension is Online	Whether to detect the SIP extension using this SIP profile is online or not				
	Off: Disable CID header.				
CID Header	• Remote-Party-ID: whether to send Remote-Party-ID or not.				
	• P-Asserted-Identify: whether to send P-Asserted-Identify or not.				
	The extension has two ways to get the real CID when make call pickup.				
Pickup Caller Refresh	Off: Disable Pickup Caller Refresh method.				
Method	• re-INVITE: Send reinvite to PBX to get real caller ID when pickup the call.				
	• Update: Send update to PBX to get real caller ID when pickup the call.				
Allow Unknown Call	If this function is enabled, incoming calls from unknown sources are allowed. Unknown sources are those IP addresses that do not fall into the source range configured for SIP trunks or SIP extensions				
Inbound Source Filter	The source of inbound calls, which is allowed. It can be an IP address or a network segment. If it is a network segment, the format is 172.16.0.0/16 or 172.16.0.0/255.255.0.0, which means calls from the network segment of 172.16 is allowed to come in. 0.0.0.0 means calls of any source is allowed to come in				
QoS	Whether to enable QoS. QoS is a technology used to solve network delay or congestion				
User Agent	Then content of the 'user agent' field in SIP packets				
Encryption	Whether to encrypt this SIP profile				
Timer T1(ms)	The value of timer T1 in SIP protocol. Default value is 500ms				
Timer T2(ms)	The value of timer T2 in SIP protocol. Default value is 4000ms				
TimerT4(ms)	The value of timer T4 in SIP protocol. Default value is 5000ms				
Timer T1X64(ms)	The value of timer T1X64 in SIP protocol. Default value is 32000ms				







# 5.5.2 FXO

On the **Profile** → **FXO** interface, you can configure the driving parameters of FXO port, including tone standard, dial timeout, ring timeout, hook-flash detection, DTMF parameters, CID-related parameters, impedance and so on.

Profile	/ FXO						
Index	Name	Tone Group	Digit Timeout(s)	Dial Timeout(s)	Detect Polarity Reverse	Detect Caller ID	
1	default	Kuwait	4	10	Off	Detect after ring/5000ms	(i) 🖻 😣
							New

Click and corresponding configuration interface will pop up.

Profile / FXO / Edit		
Index	1	
Name	default	
Tone Group	China	~
Register Param	China	~
Digit Timeout(s)	4	
Dial Timeout(s)	10	
Detect Polarity Reverse	Off	~
Delay Offhook(s)	3	
Detect Caller ID	Off	~
Dial Delay(ms)	400	



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TMF Parameters	
DTMF Send Interval(ms)	100
DTMF Duration(ms)	100
DTMF Gain	6dB 🗸
DTMF Detect Threshold	-40db
DTMF Terminator	#
Send DTMF Terminator	Off
usyTone Detect Parameters	
Detect Tone counts	8
Detect Tone Delta(ms)	50
Intermittent Ratio	1:1
	Cancel Save Reset

Name	The name of this FXO profile
Tone Group	The national standard of dialing tone, busy tone and ring tone; default value is China
Digit Timeout (s)	The timeout value for dialing a digit of a telephone number; When the time of dialing a digit exceeds this value, the system will think the dialing has completed; Default value is 4 seconds
Dial Timeout (s)	The timeout value for dialing the first telephone number after off-hook; Default value is 10 seconds
Ring Timeout (s)	The timeout value for the ringing of the FXO port when there are incoming calls
No Answer Timeout (s)	The timeout value for ending a call which goes out through the FXO port, when nobody answers the call.
Detect Polarity Reverse	Whether to enable 'detect polarity reverse'. If 'detect polarity reverse' is on, call tolls will be calculated based on the changes in voltage. If 'detect polarity reverse' is off, you need to set the time for off hook delay and call tolls will be calculated starting from the set time.
Detect Caller ID	Detect before ring: the CID will be shown before ringing; otherwise, CID will be displayed after ringing; Detect after ring: the CID will be shown after ringing; otherwise, CID will be displayed

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	before ringing
	Off: the CID will not be shown
DTMF Detect Timeout(s)	The timeout value to detect CID (in DTMF format)
Dial Delay(ms)	The delay time of dialing. Default value is 400ms
DTMF Send	The minimum interval between the sending of two DTMF tone
Interval(ms)	DTMF: Dual Tone Multi Frequency
DTMF Gain	Signal gain of DTMF
DTMF Duration (ms)	The minimum duration of a DTMF tone
DTMF Detect Threshold	The threshold for the device to detect DTMF
DTMF Terminator	The terminator for ending DTMF detection. It means when the terminator is detected, the system will think the dialing is completed and begin to process call.
Send DTMF Terminator	Whether to send DTMF terminator
Detect Tone Counts	Set the number of busy notes to check
Detect Tone Delta	Set the error size to check the busy tone
Intermittent Ratio	The intermittent ratio to detect busy tone

# 5.5.3 **Codec**

XonTel Plus supports six audio codec modes, including G729, G723, G722 PCMU PCMA and OPUS. XonTel Plus also supports six video codec modes, including VP8, H264, H263, H261, H263-1998 and H263-2000. You can adjust the priority of these modes according to you needs.

Profile / (	Codec				
Index	Name	Audio Codec	,	/ideo Codec	
1	default	PCMA@20ms, PCMU@20ms, G722@20ms	VP8, H264, H263,	H263-1998, H263-2000, H261	๔⊗
					New
	🙁 : De	lit codec profile. elete the corresponding codec profile or a co : Create a new codec profile	dec mode.		
		V	www.xor		
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Profile / Codec / Edit					
Index	1				
Name	default				
	PCMA 20ms V 🗙				
	PCMU v 20ms v 🛇				
Audio Codec	G722 ~ 20ms ~ 🛇				
Audio Codec	G723 - 30ms - 🛇				
	G729 ~ 20ms ~ 🛇				
	OPUS v 20ms v 🛇				
	VP8				
	H264 🗸 🗸				
	H263 🗸 🗸 😒				
Video Codec	H263-1998 🗸 🔀				
	H263-2000 🗸 🚫				
	H261 🗸 🗸 🔀				

# 5.5.4 Number

On the **Profile**  $\rightarrow$  **Number** interface, you can set a prefix for calling numbers or called numbers. When the prefix of a calling number or a called number matches the set prefix, the call will be passed to choose a route.

Index	Name	Caller Prefix	Caller Length	Called Prefix	Called Length	
1	international	*	*	^(\d{12,20})\$	*	<b>2</b> 0
4	kuwait local	*	*	^[2569]\d{7}\$ ^18\d{5}\$ ^1\d	*	<b></b>
						Ne
		Marrie	ding number profile l see the following int	erface:		



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Profile / Number / New	
Index	1 *
Name	test
Caller Number	
Length	5
Prefix	1 #
	2 *
Called Number	
Length	5
Prefix	1 # 2 *
	Cancel Save Reset

Name	The name of the number profile		
Prefix of Caller Number	The prefix of the calling number. It supports multiple prefixes, multiple rules for "or" relationships .It supports regular expression		
Prefix of Called Number	The prefix of the called number. It supports regular expression. It Supports multiple prefixes, multiple rules for "or" relationships.		
Length	The length of the calling number or called number. For example,: 4 6 7 means the calling number or called number must be 4 digits, 6 digits or 7 digits except the prefix		

#### **Regex (Regular Expression) Syntax**

^	Matches the starting position in a number string. For example, ^134 matches the numbers starting with 134
\$	Matches the ending position of a string. For example, 2\$ matches the numbers ending with 2.
	Separates alternate possibilities. For example, 2 3 4 means 2,3or 4.

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١	Marks the next character as a special character, a literal, a backreference, or an octal escape			
[]	Matches a single character that is contained within the bracket. For example, [123] matches 1, 2, or 3. [0-9] matches any digit from "0" to "9".			
[^]	Matches any one character except those enclosed in []. For example, [^9] matches any character except 9.			
•	Matches any single character except the newline character. For example, 3.4 matches 314, 324, 334, 344.			
?	Indicate there is zero or one of the preceding elements. For example, colour matches both color and colour			
*	Indicate there is zero or more of the preceding elements. For example, $ab*c$ matches ac, abc, abbc, abbc, and so on.			
+	Indicates there is one or more of the preceding elements. For example, ab+c matches abc, abbc, abbbc, and so on, but not ac			
\d	Mark any digit, equal to [0-9]			
\D	Mark any character that is not a digit, equal to [^0-9]			
\s	Mark any blank character such as a space or a tab.			
$\setminus S$	Mark any character that is not a blank character			

#### **Examples:**

^0755	Matches the phone numbers with starting digits of 0755.		
^0755 ^8899 ^0110	Matches the phone numbers with starting digits of 0755, 8899 or 0110.		
^[1][358][0-9]{9}\$	Matches the phone numbers with the first digit as 1, the second digit as 3, 5 or 8, the left nine digits as any of 0 to 9.		

Note: the matching of number prefix also supports some digits that are not conform to the format of regular expression. For example, 0755 matches the numbers starting with 0755, and 0755|8899|0110 matches the numbers starting with 0755, 8899 or 0110.

### 5.5.5 Time

On the **Profile**  $\rightarrow$  **Time** interface, you can set a time period for calls to choose routes. If the local time when a call is initiated falls into the set time period, the call will be passed to choose the corresponding route.







Click the **New** button, and you will see the following interface:

Profile / Time / New	
Index	1 *
Name	Timer 1
Date Period	2020-05-19~2020-05-28
Weekday	Mon U U U Wed U Thu Fri Sat
	Sun
Time Period	00:00~23:59 €
	Cancel Save Reset

Name	The name of the number profile		
Date Period	Configure the starting date and ending date of a period		
Weekday	Choose the desired week days		
Time Period	Choose the desired starting time and ending time of the day		

# 5.5.6 Manipulation

Number manipulation refers to the change of a called number or a caller number during calling process when the called number or the caller number matches the preset rules.

Profile / Manipulation				
Index	Name	Caller: Prefix/Suffix/Replace	Called: Prefix/Suffix/Replace	
1	22245888	//->22245888	//	๔⊗
				New

Click the New button, and you will see the following interface:





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Profile / Manipulation / New	
Index	1
Name	Manipulation 1
Caller	Y
Delete Prefix Count	
Delete Suffix Count	
Add Prefix	
Add Suffix	
Replace by	
Called	Г
	Cancel Save Reset

Name	The name of this manipulation profile		
Delete Prefix Count	The number of digits that are deleted from the left of the caller number or calling number		
Delete Suffix Count	The number of digits that are deleted from the right of the caller number or calling number		
Add Prefix	The prefix added to the caller number or the calling number		
Add Suffix	The suffix added to the caller number or the calling number		
Replace by	The number which replace the caller number or the calling number		
۲	If the checkbox on the right of Caller is selected, it means the caller number will be manipulated; if the checkbox on the right of Called is selected, it means the called number will be manipulated.		

Note: During number manipulation, deletion rules are carried out first, followed by adding rules. If 'Replace by' has been set, deletion rules and adding rules are invalid.

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### 5.5.7 Speed Dial

On the Profile → Speed Dial interface, you can set one-digit or two-digit peed dial numbers for SIP calls. For example, if the short number (speed dial number ) is set as 1, the long number is set as 8000, and this speed dial profile is applied to an SIP extension, the SIP extension only needs to dial 1 and the call will be directed to the extension number of 8000.

Profile / S	peed Dial				
Index	Name		Abbreviated Number Table		
1	speeddial		test/1/30/Enable	<i>(i)</i> <b>e</b> <sup></sup>	
				New	
Profile /	Speed Dial / N	lew			
Index			1 *		
Name			Speeddial1		
Abbreviate	ed Number Table				
Name	Short Number	Long Number	Status		
_					
1	1	8000	Enable T 🕣		
			Cancel Save Reset		

After that you can enable the Speed Dial profile in the extension settings as shown below.

Call Forward No Reply	Off	
NAT	On 🗸	
Call In Filter	Off	
Call Out Filter	Off	
Speed Dial	1-< speeddial >	$\leq \square$

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### 5.5.8 AutoCLIP

AutoCLIP is mainly used to SIP trunks and FXO trunks. AutoCLIP helps record the outgoing and incoming calls of a trunk.

Profile / AutoCLIP								
Configuration Record								
Index	Name	Delete Used Record	Record Strategy	Record Expire(h)	Match Outgoing Trunk			
Index 1	Name Autoclip	Delete Used Record On	Record Strategy Missed Calls	Record Expire(h) 8	Match Outgoing Trunk On	₫⊗		

Profile / AutoCLIP	
Configuration Record	
Profile / AutoCLIP / Edit	
Index	1
Name	Autoclip
Record Strategy	Missed Calls
Record Expire(h)	8
Delete Used Record	
Match Outgoing Trunk	
Enable number matching rules when it fails	On 🖌
Number matching rules	
Number rules (regular) Remove prefix	Add Prefix
/d+	0
	Cancel Save Reset

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Index	The index of AutoCLIP profile
Name	The name of AutoCLIP profile
Record Strategy	You can choose missed calls or all calls. If missed calls is selected, XonTel Plus will record the missed calls of the trunk. If all calls are selected, all the calls going through the trunk will be recorded
Record Expire (hour)	The validity period of a record. For example, if this parameter is set as 2 hours, the record will be valid in 2 hours since the record is generated. During the validity period, if there is coming call for the extension number contained in the record, the call will directly led to the extension without routing.
Delete Used Record	By default, this parameter is disabled. If this parameter is selected, those records that have been used to match extension number or trunk will be deleted.
Match Outgoing Trunk	By default, this parameter is enabled. If this parameter is enabled, those calls going through the trunks in the record can coming in without routing.
Number matching rules	for example, if you dial <b>0505443281</b> out, but <b>0505443281</b> call back with caller number is <b>505443281</b> , may match fail. Then you call config it to add 0 to match again

#### After configuring AutoCLIP profile, you have to enable it in PBX outgoing Trunk as shown below

Register	On 🗸
Username	22248999
Auth Username	22248999
Password	•••••••
Specify Transport Protocol on Register URL	Off
Expire Seconds	1800
Retry Seconds	60
From Header User Part	Register User
From Header Display Name	Caller's Number
From Header Host	Server Address
Heartbeat	Off
AutoCLIP Profile	1-< Autoclip >

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You can check AutoCLIP records for PBX outgoing trunk as shown in the figure below

Profile / AutoCLIP								
Configura	ation Record							
	Source	Caller Number	Destination	Expires	Options			
	SIP Trunk/22245888	67070182	SIP Extension/36	2020-05-21 10:39:18	$\otimes$			

## 5.5.9 Recording

On the **Profile**→ **Recording** interface, you can choose SD card or Udisk (USB) as master/slave storage location.

#### How to Record Calls:

Configure a recording profile (or choose one of the two default recording profiles), and then add it to a SIP route. When there are calls going through the route and match the recording profile, the calls will be recorded.

Profile / Recording									
Confi	guration Re	cording List							
Master Storage Location			SD Card	v					
Slave S	Storage Location		Udisk	Ŧ					
			Sav	/e					
_									
Index	Name	Strategy	Recording Direction	Stereo	Min Duration(s)	Silence Detect			
1	auto_record	Auto Recording After Answer	Inbound & Outbound	Off	1	Off/-/-/-	₫⊗		
2	manual_record	Manual Recording After Answer	Inbound & Outbound	Off	1	Off/-/-/-	₫⊗		
							New		







Profile / Recording		
Configuration Recording List		
Index	3	۳
Name	Recording 3	
Strategy	Auto Recording After Answer	٣
Recording Direction	Inbound & Outbound	Ŧ
Stereo	On	T
Min Duration(s)	1	
Silence Detect	On	٣
Initial Silence Timeout(s)	10	
Final Silence Timeout(s)	20	
Silence Detect Threshold	200	Ŧ
	Cancel Save Reset	

Index	The index of the recording profile. Range: 1-32
Name	The name of the recording profile, used to identify the recording profile
Strategy	Auto Recording after Answer: start recording after the callee pick up the phone. Ban Recording: ether caller or callee enables his function, and then the call in both directions will not be recorded. Manual Recording after Answer: press *1 to start recording after the callee answers the call.
Recording Direction	<ul> <li>Inbound &amp; Outbound: If this recording profile is added to SIP extension, both inbound and outbound calls will be recorded.</li> <li>Inbound: If this recording profile is added to SIP extension, only inbound calls will be recorded.</li> <li>Outbound: If this recording profile is added to SIP extension, only outbound calls will be recorded.</li> <li>Note: If this recording profile is added to routing, this parameter is invalid and all calls going through the routing will be recorded.</li> </ul>
Min Duration	If the actual recording time is shorter than this value, the recording file will not be saved.
Silence Detect	Select on or off.
Initial Silence Timeout(s)	If the time of initial silence is shorter than this timeout value and there is voice afterwards, the recording will not stop. If the time of initial silence is longer than this timeout value, and there is voice afterwards, the recording will stop when the recording time reaches the preset value.

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Final Silence Timeout(s)	<ul> <li>If the time of final silence is shorter than this timeout value and there is voice afterwards, the recording will not stop.</li> <li>If the time of final silence is longer than this timeout value, and there is voice afterwards, the recording will stop before the call ends.</li> <li>Note: The XonTel Plus device will not execute final silence detection unless the initial silence is shorter than its timeout value.</li> </ul>
Silence Detect Threshold	The threshold for silence detection.

#### Enable calls recording for the extension.

Allow Being Monitored	
Monitor Mode	Disable
Voicemail	Off
Recording Profile	1-< auto_record >
SIP Profile	2-< wan_default >
Status	Enable
	Cancel Save Reset

#### Enable calls recording in the route configuration for the incoming/outgoing calls.

Action	
Callback	
Distinctive Ringtone(Alert-Info)	None
Manipulation	Off
Destination	SIP Extension / 36 / 36
Recording Profile	1-< auto_record >
Failover Action	
	Cancel Save Reset







You can click **Recording List** in the recording profile to view the recording files which show the caller/called number, recording duration and so on. You can also play, download or delete the recording files on this interface.

Profile	/ Recording						
Configurati	on Recording List						
Query Par	am					E	ixpand 👻
Index	Time 🚽	Caller	Source	Called	Destination	Duration	Operation
1	2020-09-30 21:43:43	99444230	SIP Trunk/22245888	0096522245888	SIP Extension/36	01:32	▶ ┇ ⊗
2	2020-09-30 18:54:15	25655336	SIP Trunk/22245888	0096522245888	SIP Extension/38	00:14	

# 5.5.10 Voicemail

On the **Profile**→ **Voicemail** interface, you can configure the location, number and duration of a voicemail.

Profile / Voicer	mail		
Configuration	Message List		
Master Storage Loca	tion	SD Card	٣
Slave Storage Locati	on	SD Card	Ŧ
Max Messages Per U	Jser	50	
Maximum of Login At	ttempts	3	
Maximum of Operation	on Failure	3	
Min Message Time(s	ec)	3	- -
Max Message Time(r		2	Ŧ
Auto Play New Mess		2	
	aye		
Play CID Number			
Play from Latest Mes	sage		
Play Message Date		Before Playing Message	v
		Cancel Save Res	et









Master/Slave Storage Location	Select SD card or Udisk (USB)
Max Message Per User	If this maximum number of messages is reached, a prompt voice "the mail box is full" will be played.
Maximum of Login Attempts	If this maximum number of attempts (by dialing *98 to log in the voicemail box) is reached, the call will hang up.
Maximum of Operation Failure	When a call enters into the voicemail box and the caller dial inexistent DTMF repeatedly, the caller will be forced to log out the voicemail box after the repetition times exceed this value.
Min Message Time (second)	The minimum duration of a voicemail
Max Message Time (second)	The maximum duration of a voicemail.
Auto Play New Message	If this parameter is on, new messages will be played automatically. If it is off, a prompt voice "please dial 1 to listen to new message" will be given.
Play CID Number	If this parameter is on, the caller number will be played together with messages.
Play from Latest Message	If this parameter is on, the latest messages will be played first.
Play Message Date	When to play message date. You can choose 'Before Playing Message', 'After Playing Message' and 'Never'.

#### How to use voicemail:

Go to the **Extension**  $\rightarrow$  **SIP** interface, click **New** to create new SIP extension and enable the voicemail function for it, and then calls that times out will enter into voicemail.

Voicemail	On 🗸
Password	···
Message Forward Email	
Recording Profile	1-< auto_record >
SIP Profile	2-< wan_default >
Status	Enable
	Cancel Save Reset

#### <u>Note</u>

To use Message Forward Email (voicemail to Email) option, please configure PBX Email settings.







You can click **Message List** in the voicemail profile to view the voicemail files which show the caller/called number, message duration and so on. You can also play, download or delete the message files on this interface.

Profile	e / Voicemail	_						
Configura	ation Message List							
Index	Time	Caller	Source	Called	Destination I	Message Type	Duration	Operation
1	2020-05-18 15:47:55	0546683186	FXO Trunk/Port 0	388	SIP Extension/388	Common	00:03	
2	2020-04-14 13:41:24	0555418678	FXO Trunk/Port 0	388	SIP Extension/388	Common	00:03	$\blacktriangleright \downarrow \otimes$
3	2020-04-07 15:10:17	0542200402	FXO Trunk/Port 1	388	SIP Extension/388	Common	00:07	
	Configura Index 1 2	Index         Time           1         2020-05-18 15:47:55           2         2020-04-14 13:41:24	Message List           Index         Time         Caller           1         2020-05-18 15:47:55         0546683186           2         2020-04-14 13:41:24         0555418678	Message List           Index         Time         Caller         Source           1         2020-05-18 15:47:55         0546683186         FXO Trunk/Port 0           2         2020-04-14 13:41:24         0555418678         FXO Trunk/Port 0	Message List         Message List           Index         Time         Caller         Source         Called           1         2020-05-18 15:47:55         0546683186         FXO Trunk/Port 0         388           2         2020-04-14 13:41:24         0555418678         FXO Trunk/Port 0         388	Message List         Message List           Index         Time         Caller         Source         Called         Destination           1         2020-05-18 15:47:55         0546683186         FXO Trunk/Port 0         388         SIP Extension/388           2         2020-04-14 13:41:24         0555418678         FXO Trunk/Port 0         388         SIP Extension/388	Message List       Message List         Index       Time       Caller       Source       Called       Destination       Message Type         1       2020-05-18 15:47:55       0546683186       FXO Trunk/Port 0       388       SIP Extension/388       Common         2       2020-04-14 13:41:24       0555418678       FXO Trunk/Port 0       388       SIP Extension/388       Common	Message List       Message List         Index       Time       Caller       Source       Called       Destination       Message Type       Duration         1       2020-05-18 15:47:55       0546683186       FXO Trunk/Port 0       388       SIP Extension/388       Common       00:03         2       2020-04-14 13:41:24       0555418678       FXO Trunk/Port 0       388       SIP Extension/388       Common       00:03

# 5.5.11 PIN List

PIN List are used to manage lists numerical passwords that can be used to access restricted features such as outbound routes.

Profile / Pl	N List		
Index	Name	PIN List	
		This section contains no values yet	
			New

Index   Index 1   Name International   PIN List   Name Password   Status   basel   Emma   5678	Profile / PIN L	list / New		
PIN List       Name     Password       basel     1234	Index		1	~
Name     Password     Status       basel     1234     Enable	Name		International	
basel 1234 Enable V	PIN List			
	Name	Password	Status	
			Cancel Save	Reset

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# 5.6 Extension

# 5.6.1 **SIP**

On the Extension  $\rightarrow$  SIP interface, you can configure the SIP accounts registered in the XonTel Plus by SIP clients (here by XonTel Plus is regarded as a SIP server).

Extension / SIP	Imp	oort From File <del>-</del>	Export New	Batch New		Delete
Index	D 🛊 DID 🛊	Password 🛔	Register Source 🛔	Profile 🛔	Status 🛔	Filter
□ • 1 30 30		On	Any	2-< wan_default >	Enabled	() ₫ ()
□ • 2 31 31		On	Any	2-< wan_default >	Enabled	()₫⊘
<b>•</b> 3 32 32		On	Any	2-< wan_default >	Enabled	()₫⊘
Extension / SIP / Edit						
Index	1	~	)			
Name	30					
Extension	30					
Password	•••••		•			
Outbound CID						
DID			$\oplus$			
Max Concurrent Register	2	~	)			
Max Concurrent Call	1	~				
Ring Timeout(s)	50					
			)			
Original Called Number Location(Send INVITE)	Off	~	ļ			
Register Source	Any	~	]			
Call Pickup	Ring Group	~	]			
Call Waiting	Off	~	]			
Do Not Disturb	Off	~	]			

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Call Forward Unconditional	Off	<b>~</b>
Call Forward Unregister	Off	~
Call Forward Busy	Off	~
Call Forward No Reply	Off	¥
NAT	On	✓
Call In Filter	Off	<b>~</b>
Call Out Filter	Off	~
Speed Dial	1-< speeddial >	~
Allow Being Monitored		
Monitor Mode	Disable	<b>~</b>
Voicemail	Off	~
Recording Profile	1-< auto_record >	~
SIP Profile	2-< wan_default >	~
Status	Enable	~
	Cancel Save Rese	t

Name	The name of this SIP extension
Extension	The SIP account of the extension registered in XonTel Plus by a SIP client
Password	The password of the SIP account registered in XonTel Plus by a SIP client
Outbound DID	Outbound Direct Inward Dialing. Overrides the caller ID when dialing out a trunk. Leave this field blank to disable the outbound caller ID feature for this user. If you leave it blank, the system will use the route or trunk caller ID, if set.
DID	Direct Inward Dialing; if the called number is same with DID, the call will be directly forwarded to the extension, rather than choosing a route. Users can set multiple DID.
Max Concurrent Register	XonTel Plus PBX supports SIP forking. <b>SIP forking</b> refers to the process of "forking" a single SIP call to multiple SIP endpoints. The value of Concurrent Registrations limits how many SIP endpoints the extension can be registered.
Max Concurrent Call	Maximum simultaneous calls to/from one extension
Ring Timeout(s)	Customize the timeout in seconds. Phone will stop ringing over the time defined.

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Register Source	If 'Any' is chosen, all SIP clients are allowed to register the SIP account of this extension; if 'Specified' is chosen, only the SIP client with the specified IP address or network segment is allowed to register the SIP account of this extension.				
Call Waiting	If a calling party places a call to a called party which is otherwise engaged, and the called party has the call waiting feature enabled, the calling party will hear an IVR voice.				
	Allows extension to answer another extension incoming call.				
Call Pickup	<b>Ring Group</b> : Extension can pick up the call that is ringing at the other extension <b>that is in the same ring group only</b> .				
	Local extension: Extension can pick the call that is ringing at the local extensions				
	Off: Disable Call Pickup feature from this extension				
Do Not Disturb	If 'Do Not Disturb' feature is enabled, calls cannot reach the called party.				
Call Forward Unconditional	If 'Call Forward Unconditional' feature is enabled, all coming calls will be forwarded to a preset number.				
Call Forward Unregister	When the SIP extension is not registered, you can transfer all the calls to the set number				
Call Forward Busy	If 'Call Forward Busy' feature is enabled, new coming call will be forwarded when the corresponding local port is busy.				
Call Forward No Reply	If 'Call Forward No Reply' feature is enabled, calls will be forwarded when nobody answer the calls during a specified period.				
NAT	If NAT is enabled, the IP address of SIP extension in LAN will be turned into the outbound IP address of public network, thus making NAT traversal possible				
Call In Filter	When you breathe in to SIP, you match the relevant filter conditions				
Call Out Filter	When the SIP is called out, the filter conditions are matched				
Speed Dial	Set speed dial profile that will be used on this extension				
Allowing Being monitored	Check this option to allow this user to be monitored.				
	Decide how you will monitor another extension's current call.				
	<b>Disable</b> : you will not be allowed to monitor other's call.				
Monitor Mode	□ <b>Listen Mode</b> : you can only listen to the call, but can't talk (default feature code: <b>*222</b> ).				
	□ Whisper Mode: you can talk to the extension you're monitoring without being heard by the other party (default feature code: *223).				
	□ <b>Barge-in Mode</b> : you can talk to both parties (default feature code: <b>*224</b> ).				
	Check this box to enable voicemail for this extension.				
Voicemail	<b>Password</b> : Voicemail password used to access voicemail system. This password can contain only numbers.				
	Message Forward Email: Check this box to send voicemail to the extension Email address. To use this feature, "Email Settings" need to be configured correctly.				

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Recording Profile	Set calls recording profile that will be used on this extension
SIP Profile	The SIP profile that is selected for the extension
Status	If it is enabled, this SIP extension is registered to the XonTel Plus device; Otherwise the SIP extension is not registered

# 5.6.2 Ring Group

On the Extension  $\rightarrow$  Ring Group interface, you can group SIP extensions together and set strategy for choosing the SIP extension to ring under a ring group.

Extension / Ring Group									
Index	Name	Number	Members Strategy						
1	incomingcalls	6200	SIP Extens	sion-< Kuwait-Ali >	SIP E	Extensio	on-< Egypt >	Simultaneous	() ₫⊗
									New
Exten	sion / Ring G	roup / New							
Index				2	~				
Name				incoming calls2					
Members	Select			Select All Source lis SIP Extension / Kuwa 301	_	>	Select All Target list 0/2 SIP Extension / test / 303 SIP Extension / Egypt / 302		
Strategy				Sequence(Ascending)	~				
Ring Grou	ıp Number			6300					
DID									
Ring Time	e(5s~200s)			25					
When no	answer transfer to			SIP Extension / Kuwait-	Ali / 3 🗸				
				Cancel Save	Reset				

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Name	The name of this ring group
Members Select	Select the SIP extension or several SIP extensions. Add extension to the ring group by adding it to the target list.
Strategy	The strategies for choosing which SIP extension to ring, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random
Ring Group Number	The number of the ring group
DID	Same with Ring Group Number; it is optional to fill in
Ring Time (5-200s)	The duration of ring when there is an incoming call. Range: 5s to 200s
When no answer transfer to	Set failover destination for the created ring group



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# 5.6.3 Paging Group

On the Extension -> Paging Group interface, you can group SIP extensions into a paging group and then if there calls given from FXO/SIP to the paging group, the calls will be directed to one extension of the paging group according to the preset strategy.

Extension / Paging Group				
Index Name	Members	Strate	Strategy	
	This section contains no va	lues yet		New
Extension / Paging Group / New				
Index	1	~		
Name	Paging Group			
Paging Group Number	7000			
Strategy	1-way Paging	~		
Members Select	<ul> <li>Select All Source list 0/17</li> <li>SIP Extension / 33 / 33</li> <li>SIP Extension / 34 / 34</li> <li>SIP Extension / 35 / 35</li> <li>SIP Extension / 36 / 36</li> <li>SIP Extension / 37 / 37</li> <li>SIP Extension / 38 / 38</li> <li>SIP Extension / 39 / 39</li> <li>SIP Extension / 40 / 40</li> <li>SIP Extension / 41 / 41</li> <li>Cancel Save Reset</li> </ul>		Select All Target list 0/3 SIP Extension / 30 / 30 SIP Extension / 31 / 31 SIP Extension / 32 / 32	<ul><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li><li></li></ul> <li></li>

Name	The name of this paging group
Paging Group Number	The number of the paging group.
Strategy	<ul> <li>1-way paging: members of the paging group only can listen to the voice of presenter and cannot answer the call.</li> <li>2-way intercom: members of the paging group can have conversation with the presenter, but members cannot talk to each other.</li> </ul>
Members Select	Select the SIP extensions that are added into the paging group. Add extension to the paging group by adding it to the target list.

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# 5.6.4 Call Queue

On the **Extension**  $\rightarrow$  **Call Queue** interface, you can set a strategy of how to handle the queue, members that answer the queue, waiting music and other parameters for a call queue.

With VoIP, call queue is a concept used in inbound call centers. Call centers use an automatic call distributor to distribute incoming calls to specific resources or agents within the center. This feature is ideal and necessary for answering calls in a fair and orderly manner, especially business VoIP. It is important for business with a large inbound call volume. VoIP makes it possible to manage these calls in an efficient and organized way, without the high cost of a third-party system.

Extension /	Call Queue		
Call Queue D	ynamic Agent Login Setting		
Index Name	Number	Members	Strategy
		This section contains no values yet	
			New
Extension /	Call Queue		
Call Queue	Dynamic Agent Login Setting		
Login Suffix		*	
Logout Suffix		**	

- Login Suffix: The suffix for a member of the call queue to log in. Example call queue number 6300, to login press 6300\*
- Logout Suffix: The suffix for a member of the call queue to log out.







Extension /	/ Call	Queue /	New
-------------	--------	---------	-----

Index	4	~
Name		
Strategy	Simultaneous	~
Call Queue Number		
Agent Wrap Time(5s~300s)	45	
Agent wrap nine(os~300s)	15	
Agent Ring Time(5s~300s)	15	
Menu Tone	Off	<u>v</u>
ment fore		v
Waiting Music	Default Tone	~
Max Wait Time(0s~300s)	60	
· · ·		
Call Forward Timeout	Hangup	~
Leave When Queue Empty	On	~
Call Forward Queue Empty	Hangup	~
Max Queue Length	0	
Call Forward Exceed Length	Hangup	~
Max No Answer	0	
Enable Position Announcement	Off	~
Members Select	Select All Source list 0/7	Select All Target list 0/0
	SIP Extension / Eqaila / 2	
	SIP Extension / Manager	
	<ul> <li>SIP Extension / Mahboul</li> <li>SIP Extension / 501 / 501</li> </ul>	
	C OI EXtension / JUT / JU	

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Name	The name of this queue
Strategy	<ul> <li>Simultaneous: All available agents will ring simultaneously until one answer.</li> <li>Liner: rings agents in the order specified in the queue configuration.</li> <li>Random: ring a random agent.</li> <li>Memory Round Robin: Round Robin with Memory, remembers where it left off in the last ring pass.</li> <li>Least Recent: ring the agent which was least recently called.</li> <li>Fewest Calls: ring the agent with the fewest completed calls.</li> </ul>
Call Queue Number	The number of the queue number.
Agent Wrap Time(5s~300s)	How many seconds after the completion of a call an Agent will have before the queue can ring with a new call.
Agent Ring Time(5s~300s)	The number of seconds an agent's phone can ring before we consider it a timeout.
Menu Tone	Announcement played to callers once prior to joining the queue.
Waiting Music	Select the "Music on Hold" for this queue.
Max Wait Time(0s~300s)	Defines the maximum number of seconds a caller can wait in a queue before being pulled out.
Call Forward Timeout	Set the failover destination for the caller who pulled out from the queue.
Leave When Queue Empty	If enabled, callers already on hold will be forced out from the queue when no agents available.
Call Forward Queue Empty	Set the failover destination for the caller who forced out from the queue when no agents available to answer his call (queue empty).
Max Queue Length	Maximum number of callers who can wait in the queue.
Call Forward Exceed Length	Set the failover destination for the caller who forced out from the queue when the queue exceeds the length.
Max No Answer	when the extension is not answering the calls (include ring timeout, reject, offline) more than the value, the status of this extension will become ON-Break and the extension will not get any call till its login again.
Enable Position Announcement	Announce position of caller in the queue.
Members Select	Select the SIP extensions that are added into the queue.

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# 5.7 **Trunk**

# 5.7.1 SIP

SIP trunk can realize the connection between XonTel Plus and IPPBX or SIP servers.

Trunk /	SIP							
Index	Name	Realm	Transport	Heartbeat	Register	SIP Profile	Status	
1	Fastelco	10.196.32.33:5060	UDP	Off	Off	1-< peer >	Enabled	()₫⊘⊗
								New
Name								
Address								
Port								
Outbound Pro	оху							
Port								
Transport		l	JDP	~				
Register			Dn	~				
Usernar	ne							
Auth Us	ername							
Passwo	rd			٥				
Specify	Transport Protocol or	n Register URL	Dff	~				
Expire S	Seconds		1800					
Retry Se	econds	6	60					
From Header	User Part		Caller's Number	~				
From Header	Display Name		Caller's Number	~				
From Header	Host		ocal Address	~				
Heartbeat			Dff	~				
AutoCLIP Pro	ofile		Dff	~				
DNIS			Off	~				
SIP Profile		•	I-< lan_default >	~				
Outbound Co	dec Profile	•	I-< default >	~				
Extra Param								
Status		1	Enable	~				

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Name	The name of the SIP trunk.
Address	The IP address or domain name of the SIP devices or servers.
Port	The SIP listening port of the peer SIP devices or servers.
Outbound Proxy	If outbound proxy is used, enter the IP address or domain name of the proxy server.
Port	If outbound proxy is used, enter the listening port of the proxy server.
Transport	Transport protocol: TCP or UDP or TLS.
Register	If it is on, the SIP trunk will send register request to the peer device.
Username	The username of this SIP trunk.
Auth Username	The username used for register authentication by this SIP trunk.
Password	The password used for register authentication by this SIP trunk.
Specify Transport Protocol on Register URL	Whether to specify transport protocol on register URL.
Expire Seconds	The validity period after the SIP trunk is registered successfully. When the time expires, the SIP trunk will send register request to the server. Default value is 1800 seconds.
Retry Seconds	When the SIP trunk fails to be registered, the interval to send register request. Default value is 60 seconds.
From Header User Part	Choose Caller's Name, Caller's Display Name, Custom or Register User.
From Header Display Name	Choose Caller's Name, Caller's Display Name, Custom or Register User.
From Header Host	Choose Local Address, Server Address or Custom.
Heartbeat	If heartbeat in on, heartbeat (options) messages will be sent to examine the connection with servers. The default value is 'Off'.
Heartbeat Period(s)	The interval of sending heartbeat (options) messages in seconds.
AutoCLIP Profile	Choose an AutoCLIP profile.
DNIS	If this option is on, a trunk name will be displayed as caller ID (name) when there is an incoming call on this trunk.
SIP Profile	The SIP profile of the SIP Trunk; make reference to Profile $\rightarrow$ SIP section
Outbound Codec Profile	The codec profile which will be used when make outbound calls through this SIP trunk.
Extra Param	Here you can add the parameter you want, it will add it in request line of INVITE or FROM or TO. For example, you input " <b>user=phone</b> ", it will add it in SIP INVITE message.
Status	If it is enabled, it means the SIP Trunk can be used; otherwise, the SIP trunk is unavailable.

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# 5.7.2 FXO

FXO Trunk interconnects the PSTN with XonTel Plus. Calls from the PSTN can come into the PBX and calls can go out from the PBX to search telephone numbers under the PSTN.

runk / FXO						
XO Automatch Impedance	Busytone Learn	ing				
ort Extension Autodial Num	Register to SIP Server	RX Gain	TX Gain	Impedance	Profile	
0 8000	Off	0 dB	4 dB	600 Ohm	1-< default >	ď
1 8001	Off	0 dB	4 dB	600 Ohm	1-< default >	ď
FXO Automatch Impedance	Busytone Learnin	g				
Trunk / FXO / Edit						
Port		0				
Extension		8000				
Autodial Number						
Register to SIP Server		On		~		
Master Server		SIP Trunk	k / 22204441	~		
Slave Server		Not Confi	ig	~		
Username						
Auth Username						
Password				ø		
Specify Transport Protocol on	Register URL	Off		~		
Expire Seconds		1800				
Retry Seconds		60				
Display Name / Username Format		Caller ID	/ Caller ID	~		
Display Name / Username Format v	when CID unavailable		lame / Extension	~		
Gain Configure Mode		General S	Settings	~		
TX Gain(IP->PSTN)		+4 dB		~		
RX Gain(PSTN->IP)		0 dB		~		
Impedance		600 Ohm		~		
AutoCLIP Profile		Off		~		
FXO Profile		1-< defau	ılt >	~		

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Port	The FXO port number.
Extension	The extension account of the FXO port, which is used to register.
Autodial Number	The autodial number of the FXO port when there are incoming calls.
Register to SIP Server	If it is enabled, the FXO trunk will be registered to the SIP trunk that has been set. Default is off.
Master Server	The address and port of the master SIP server. It is generally the IP address of a SIP trunk. Please make reference to Trunk $\rightarrow$ SIP section
Slave Server	The address and port of the slave SIP server.
Username	Username of the FXO port account, used for the authentication of registration.
Auth Username	Username of this FXO trunk, which is used during register authentication.
Password	Password of this FXO trunk, which is used during register authentication.
Specify Transport Protocol on Register URL	Whether to specify transport protocol on register URL.
Expire Seconds	The validity period after the FXO trunk is registered successfully. When the time expires, the FXO trunk will send register request to the server. Default value is 1800s
Retry Seconds	When the FXO trunk fails to be registered, the interval to send register request. Default value is 60s
Display Name/Username Format	The format to display caller information, including: Caller ID/Caller ID Display Name/ Caller ID Extension/ Caller ID Caller ID/ Extension Anonymous
Display Name / Username Format when CID unavailable	Set the caller's caller id format when the main number is not detected.
Gain Configure Mode	Choose General Settings or Advanced Settings.
TX Gain(IP→PSTN)	The TX Gain for the transmitting channel of FXO Port.
RX Gain(PSTN $\rightarrow$ IP)	The RX Gain for the receiving channel of FXO Port.
Impedance	The impedance (SLIC) matched with phones.
AutoCLIP Profile	Choose an AutoCLIP profile or keep it off.
FXO Profile	The FXO profile that is selected for this FXO port.
Status	If it is on, this FXO trunk can be used, otherwise, the FXO trunk is unavailable.







#### **FXO** Automatch Impedance:

Click the **Detection** button, and the XonTel Plus PBX will automatically detect the most-matched impedance.

Trunk / FXO					
FXO Automa	atch Impedance	Busytone Lear	ning		
FXO			Port 0	*	
Automatch Mode			Simple	×	
Current Impedanc	е		600 Ohm		
Current Transhybr	id Balancing Para	m	0	1	
DTMF			1234567890123456789	Start	
Automatch Optimu	ım Impedance				
Automatch Optimu	ım Transhybrid Bal	ancing Param		1	
			Cancel Save		

#### **FXO Busytone Learning:**

Click the **Detection** button, and the XonTel Plus PBX will automatically detect the most-matched cadence.

Trunk / FXO				
FXO Automatch Impedance	Busytone Learning			
FXO		Port 1		
Current Candence		260,240,0,0,0,0,0,0	l	
Destination Number		1234567890#	Start	
Original Cadence				
Automatch Optimum Cadence				
		Cancel Save		

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# 5.8 Call Control

This section is to configure routes or route groups for incoming and outgoing calls through XonTel Plus, as well as IVR, Feature Codes and so on.

# 5.8.1 Setting

Call Control / Setting		
Voice		
Disconnect call when no RTP packet	2	
Period without RTP packet(10s~300s)	60	
Packet Loss Concealment(PLC)		
Echo Path Change Detection(EPCD)		
Non-Linear Processor(NLP)	Low	٣
Echo Gain	-4dB	Ŧ
Echo Canceller Tail Length(ms)	128	Ŧ
DTMF Min Detect Interval(ms)	0	5
RTP Port Range	10000-20000	5
Tone		
Waiting Music	Default Tone	*
Route		
Local extension call	2	
FXO extension dial out	8	
FAX		
Send Mode	T.30	Ŧ
Tone Detection by Local		
SDP Param		
a=X-fax		
a=fax	•	
a=X-modem	•	
a=modem		

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Disconnect call when no RTP packet	If it is enabled, and no RTP packets are received in PBX, calls will be disconnected.
Period without RTP packet(10s~300s)	If no RTP packets are received within the present time, calls will be disconnected.
Packet Loss Concealment(PLC)	Whether to enable the 'Packet Loss Concealment' function.
Echo Path Change Detection(EPCD)	Whether to enable the 'Echo Path Change Detection' function.
Non-Linear Processor(NLP)	Choose Off, Low, Normal and High.
Echo Gain	Default value: -4dB.
Echo Canceller Tail Length(ms)	Default value is 128.
DTMF Min Detect Interval(ms)	The minimum time for DTMF detection
RTP Port Range	Enter the start port and of end port RTP packets
Waiting Music	Choose a tone as waiting music (music on hold)
Local extension call	If it is enabled, calls between local extensions do not need routes.
FXO extension dial out	Whether to dial out FXO extension
Fax Mode	T30 or T38
Tone Detection by Local	If it is enabled, XonTel Plus will detect fax tones automatically during a call and the call will be switched into fax mode after a fax tone is detected.
SDP Param 'a=X-fax'	Attribute parameter 'a=X-fax' is carried in SDP
SDP Param 'a=fax'	Attribute parameter 'a=fax' is carried in SDP
SDP Param 'a=X-modem'	Attribute parameter 'a=X-modem' is carried in SDP
SDP Param 'a=modem'	Attribute parameter 'a=modem' is carried in SDP







### 5.8.2 Route Group

On the **Call Control**  $\rightarrow$  **Route Group** interface, you can group SIP trunks, SIP extensions, and FXO trunks together according to your needs and set strategy for choosing which trunk or extension as the destination route under a route group.

Call Co	ontrol / R	loute Group				
Index	Name		Members		Strategy	
1	out	SIP Trunk-< 22215999 >	SIP Trunk-< 22216999 >	SIP Trunk-< 22248999 >	Sequence(Ascending)	๔⊗
						New

Call Control / Route Group / New		
Index Name Members Select	2 V Group out Select All Source list 0/24 SIP Extension / 34 / 34 SIP Extension / 35 / 35 SIP Extension / 36 / 36 SIP Extension / 37 / 37 SIP Extension / 38 / 38 SIP Extension / 39 / 39 SIP Extension / 40 / 40 SIP Extension / 41 / 41	<ul> <li>Select All Target list 0/3</li> <li>FXO Trunk / Port 0</li> <li>FXO Trunk / Port 1</li> <li>SIP Trunk / 22216999</li> <li>✓</li> <li>✓</li> </ul>
Strategy	SIP Extension / 42 / 42 Sequence(Ascending) Cancel Save Reset	

Name	The name of the route group.
Members Select	Select SIP extension(s), SIP trunk or FXO trunks.
Strategy	The strategies for choosing which route under the route group as the destination route, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random







### 5.8.3 Route

On the **Call Control**  $\rightarrow$  **Route** interface, you can configure routes for incoming calls and outgoing calls.

Call Control / Route / New		
Priority	29	~
Name		
Condition		
Source	SIP Trunk / GSMTRUNK	~
Number Profile	Off	~
		×
Caller Number Prefix		
Called Number Prefix		
Time Profile	Any	~
1. March		
Action		
Callback		
Delay before Callback(s)	10	
Distinctive Ringtone(Alert-Info)	None	~
Manipulation	Off	~
Destination	SIP Trunk / GSMTRUNK	~
Deschauori	SIF HURK/ GSWITKUNK	Y
Password Type	Off	~
Recording Profile	Off	~
Failover Action		
Condition	Busy Timeout Unav	vailable
Other Condition Code		
Manipulation	Off	~

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The priority for choosing the route; the higher value, the lower priority.
The name of the route.
The condition under which the route will be used
The source of the call; it can be SIP extension, FXO trunk, SIP trunk a customized source or any.
The profile of the caller number and the called number; please make reference to the Profile $\rightarrow$ Number section. The default value is 'Off'.
Note: it cannot be simultaneously used with the following parameters of 'caller number prefix' and 'called number prefix'.
The prefix of caller number; it supports regular expression.
The prefix of called number; it supports regular expression.
The profile of time during which the route can be used; make reference to the <b>Profile → Time</b> section
Include manipulating number and sending call to destination.
This feature allows callers to hang up and get called back to XonTel Plus. This feature could reduce the cost for the users who work out of the office using their own mobile phones. <b>Delay before Callback(s):</b> Set the number of seconds before PBX calling back a caller.
The system supports mapping to custom ring tone files. For example, if you configure the distinctive ringing for custom ring tone to "Family", the ring tone will be played if the phone receives the incoming call.
Please note that the IP phone must support this feature also.
If it is on, the caller number or called number of the route will be manipulated; make reference to the <b>Profile</b> → <b>Manipulation</b> section.
The destination of the route.
If you enable this option user extension has to insert the password to use the route. • Single Pin: Use single password.
• PIN List: Select the configured PIN list for list of passwords.
Choose Off or a recording profile.
The processing when a call through this route fails.

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### 5.8.4 Feature Codes

XonTel Plus provides convenient telephone functions. Connect an extension and dial a specific feature code, and you can query corresponding information after hearing IVR.

The following is the corresponding function of each feature code:

Call (	Control / Feature Code				
Feature	Code Service		Dn ▼ Sa	we	
Index	Feature	Key	Description	Status	
1	Inquiry LAN IP	*158	Inquiry LAN IP	Enabled	₫ Ø
2	Inquiry WAN IP	*159	Inquiry WAN IP	Enabled	20
3	Inquiry Phone Number	*114	Inquiry Phone Number	Enabled	20
4	Network Work Mode	*157*	Dail *157*0 to set route mode.Dail *157*1 to set bridge mode	Enabled	
5	IP Address Config Mode	*150*	*150*1#-Static, *150*2#-DHCP	Enabled	₫ Ø
6	Configure IP Address	*152*	Set IPv4 Address 192.168.1.10 by dial *152*192*168*1*10#	Enabled	
7	Configure Gateway	*156*	Set IPv4 Gateway 192.168.1.1 by dial *156*192*168*1*1#	Enabled	20
8	Configure Subnet Mask	*153*	Set IPv4 Netmask 255.255.0.0 by dial *153*255*255*0*0#	Enabled	
9	Restart Device	*111	Restart Device	Disabled	r 🗸
10	Call Waiting Activate	*70	Enable Call Waiting service	Enabled	20
11	Call Waiting Deactivate	*71	Disable Call Waiting service	Enabled	20
12	Blind Transfer	*3	Example:*38000#,you can blind transfer to the extension number 8	Enabled	20
13	Attended Transfer	*2	Example:*28000#,you can attended transfer to the extension numb	Enabled	20
14	Call Forwarding Uncondition Activate	*72	Enable Call Forwarding Uncondition service.Example:*728000,set	Enabled	20
15	Call Forwarding Uncondition Deactivate	<b>*</b> 073	Disable Call Forwarding Uncondition service	Enabled	20
16	Call Forwarding Busy Activate	*90	Enable Call Forwarding Busy service.Example:*908000,set the call	Enabled	20
17	Call Forwarding Busy Deactivate	*91	Disable Call Forwarding Busy service	Enabled	20
18	Call Forwarding No Reply Activate	*52	Enable Call Forwarding No Reply service.Example:*528000,set th	Enabled	20
19	Call Forwarding No Reply Deactivate	*53	Disable Call Forwarding No Reply service	Enabled	20
20	DND Activate	*78	Enable Do Not Disturb service	Enabled	20
21	DND Deactivate	*79	Disable Do Not Disturb service	Enabled	20
22	Call Pickup	*4	Pick up the ringing extension, Example:*48000, pick up the extensi	Enabled	₫ Ø
23	WAN Access Control	*160*	*160*1# - Allow HTTP WAN access, *160*0# - Deny HTTP WAN a	Enabled	20
24	Voicemail Service	*98	*981# - Leave messages, *982# - Play messages	Enabled	20
25	Callback Service	*163	Callback the last received call	Enabled	20
26	Recording Service	*1	Start or stop recording when manual recording	Enabled	20
27	Call Park	*6	Example: *8, you can park another part during the call. *8100, you	Enabled	20
28	Call Monitor	*22	*222 - Listen Mode, *223 - Whisper Mode, *224 - Barge-in Mode. E	Enabled	₫ Ø
29	Auto Answer	*80	Make an intercom with a specific extension user, Example: dial *80	Enabled	20

To a disable a specific feature code, click the button •



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To a enable a specific feature code, click the button •



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Feature Code	Corresponding Function
*158	Dial *159 to inquiry LAN IP
*159	Dial *158 to inquiry WAN IP
*114	Dial *114 to inquiry phone number
*157*	Dial *157*0 to set route mode; dial *157*1 to set bridge mode
*150*	Dial *150*1 to set IP address as static IP address; dial *150*2 to set IP address as DHCP IP address
*152*	Dial *152* to set IPv4 address, for example:
	Dial *152*192*168*1*10# to set IPv4 address as 192.168.1.10
*156*	Dial *156* to set IPv4 Gateway address, for example:
	Dial *156*192*168*1*1# to set IPv4 Gateway address as 192.168.1.1
*153*	Dial *153* to set IPv4 netmask, for example:
	Dial *153*255*255*0*0*# to set IPv4 netmask as 255.255.0.0
*111	Dial *111 to restart the XonTel Plus device
*70	Dial *70 to enable the call waiting service
*71	Dial *70 to disable the call waiting service
*3	Dial *3 to trigger blind transfer, for example:
	Dial *38000, and you can blind transfer to the extension number 8000
*2	Dial *2 to trigger attended transfer, for example:
	Dial *28000, and you can attend transfer to the extension number 8000
*72*	Enable unconditional call forwarding service. Example: Dial *72*8000, and calls will be unconditionally forwarded to extension number 8000
*073	Disable unconditional call forwarding service
*90*	Enable the 'call forwarding on busy' service. Example: Dial *90*8000, and calls will be
	forwarded to extension number 8000 when the called number is busy
*91	Disable the 'call forwarding on busy' service
*52*	Enable the 'call forwarding on no reply' service. Example: Dial *52*8000, and calls will
	be forwarded to extension number 8000 when there is no reply from the called number
*53	Disable the 'call forwarding on no reply' service
*78	Enable the 'Do Not Disturb' service
*79	Disable the 'Do Not Disturb' service



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*4	Pick up the ringing extension.
	Example: Dial *48000, and you can take the incoming call of extension number 8000
*160*	Dial *160*1# to allow HTTP WAN access, Dial *160*0# to deny HTTP WAN access
*98	Dial *98 to check voicemail. The system will prompt you for password.
	*981# - Leave messages.
	*982# - Play messages
*163	Callback the last received call.
*1	Start or stop recording when manual recording is applied.
*6	Park the call.
	By default, PBX will generate parking lot range for the parked calls starting from
	the first parked call (1-100) and you can change this range according to your needs.
	Example: During the call dial *6 to park the call then go to another extension and dial
	*6100 to pickup the number 100 from parking lot.
*22	Monitor the extension call.
	Dial *222 to initiate Listen monitoring. In this mode you can only listen to the call but
	can't talk.
	Dial *223 to initiate Whisper monitoring. In this mode you can listen and talk to the
	monitored extension without being heard by the other party.
	Dial *224 to initiate Barge-in monitoring. In this mode you can listen and talk with both
	parties.
	Example: Dial *2231000 to monitor the extension 1000 in whisper mode.
	Note: To monitor an extension, you need to configure monitor settings for this
	extension first.
*80	Dial *80 and an extension number to page that extension.
	Example: Dial *80300, then the extension number 300 will be picked up.

#### Notes:

- 1. A voice prompt indicating successful configuration will be given after each configuration procedure. Please do not hang up until hearing this voice prompt.
- 2. You can edit and customize your feature codes as shown below.







Cal	l Contro	/ Feature	Code /	Edit
-----	----------	-----------	--------	------

Index	29
Feature	Auto Answer
Кеу	*80
Description	Make an intercom with a specific extension user, Example: dial *801000, then the extension 1000 will be automatically picked up.
Status	Enable •



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### 5.8.5 **IVR**

On the **Call Control**  $\rightarrow$  **IVR** interface, you can carry out specific configurations for the IVR which has been uploaded from the **System**  $\rightarrow$  **Voice** interface.

Callcontrol / IVR / Ne	w					
Index			2		~	
Name						
Menu Tone			Off		~	
Repeat Loops			3			
Enable Direct Extension			Off		Ŷ	
Select Invalid Times			3			
Select Invalid Tone			Off		Ý	
Destination Invalid Times			3			
Destination Invalid Tone			Off		Ý	
Response Timeout(s)			5			
Digit Timeout(s)			3			
Response Timeout Tone			Off		Ŷ	
Exit Tone			Off		Ŷ	
Status			Enable		Ŷ	
Menu						
DTMF	Tone		Destination			
0	Off	~	Extension	× (	SIP Extension /	30 ~

Name	The name of the IVR
Menu Tone	Choose Off or a voice prompt that you uploaded in System $\rightarrow$ Voice interface.
Repeat Loops	If it is set as '3', the call will be hanged up after the IVR has been repeated for three times during timeout.
Enable Direct Extension	Whether to allow direct dialing of extensions during the playing of IVR.
Select Invalid Times	Number of times to retry when receiving an invalid/unmatched response from the caller. Please note that this option will be ignored when " <b>Enable Direct Extension</b> " option is enabled.

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Select Invalid Tone	Prompt to be played when an invalid/unmatched response is received from the caller. Please note that this option will be ignored when " <b>Enable Direct Extension</b> " option is enabled.	
	Please note that this option will be ignored when Enable Direct Extension option is enabled.	
Destination Invalid Times	The number of times to retry when receiving an invalid/unmatched extension number from the calling side if " <b>Enable Direct Extension</b> " option is enabled.	
Destination Invalid Tone	Prompt to be played when an invalid/unmatched extension number is received from the calling side if " <b>Enable Direct Extension</b> " option is enabled.	
Response Timeout(s)	If no DTMF tone is received during the time that you have set in seconds, the IVR will be repeated or the call will be hanged up. The default value is 10 seconds.	
Digit Timeout(s)	How long (in seconds) PBX wait for the caller to enter an option on their phone keypad before PB consider it time out and it follows the Timeout Destination.	
Response Timeout Tone	Prompt to be played if no DTMF tone is received during the time that you have set in <b>Timeout</b> .	
Exit Tone	Prompt to be played for IVR exit.	
Status	If it is disabled, the IVR cannot be seen in the destination of route.	
	DTMF: It can be 0-9 quick-dial numbers, *, #, others or timeout.	
Menu	Destination: the destination of the IVR; it can be an extension or a trunk.	
	For example, if DTMF is configured as 1,2,3 and others, and the telephone key that is pressed is not 1, 2 or 3, the IVR will choose the destination of 'others.	
	When the playing of the IVR times out, and user does not press any telephone key, the IVR will choose the destination of 'timeout'.	
	When the destination is a trunk, user does not need to pre-configure the called number, and the system will prompt the user to dial the called number.	







### 5.8.6 Diagnostics

In case that call cannot be connected or voice has quality problem, you can enter into the **Call Control**  $\rightarrow$  **Diagnostics**  $\rightarrow$  **Call Trace** to collect fault-related information and then send it to technical support to locate fault.

**Operation Procedures:** 

1. Select the module that need to be traced. For example, if a call from SIP to FXO has voice problem, you can select SIP message, FXS/FXO and Voice, and then click the **Start** button.

Call Control / Diagnostics				
Call Trace	SIP Test			
r				
Select the module you want to trace		□ SIP Stack 🗹 SIP Message 🗹 FXS/FXO □ DSP □ Voice		
		Start		

2. Give a call, and come back to the **Call Control** → **Diagnostics** interface after the call ends then click **Stop & Download** to stop the call trace and download the tracing file.

Unapplyed Changes:1 >>Apply >>Revert	
r	
Call Control / Diagnostics	
Call Trace	
Select the module you want to trace	SIP Stack SIP Message FXS/FXO DSP Voice
Call Trace is working	
	Stop & Download

3. In order to locate faults more quickly, you sometimes need to enter into the **System**  $\rightarrow$  Service Log interface, click export, and then send this exported file and the tracing file to technical support,







Also to do SIP test, you can enter into the Call Control  $\rightarrow$  Diagnostics  $\rightarrow$  SIP Test to collect related information and then send it to XonTel technical support to locate fault.

when you do this, the XonTel Plus PBX will run as specific mode as SIPpy, need reboot the device to resume.

Call Control / Diagnostics	
Call Trace SIP Test	
,	
SIP Test will stop all call service ! Please restart device after test !	
Source IP:Port : Destination IP:Port	
Scenario File Message Flow Message Detail Statistics	Start
1	







### **6** Glossary

Glossary	Description	
ARP	Address Resolution Protocol	
CID	Caller Identification	
DNS	Domain Name Server	
DDNS	Dynamic Domain Name Server	
DHCP	Dynamic Host Configuration Protocol	
DMZ	Demilitarized Zone	
DND	Do Not Disturb	
DTMF	Dual Tone Multi Frequency	
FTP	File Transfer Protocol	
HTTP	HyperText Transfer Protocol	
LAN	Local Area Network	
L2TP	Layer 2 Tunneling Protocol	
PPTP	Point-to-Point Tunneling Protocol	
MAC Address	Media Access Control Address	
NAT	Network Address Translation	
Ping	Packet Internet Gopher	
SIP	Session Initiation Protocol	
ТСР	Transmission Control Protocol	
UDP	User Datagram Protocol	
RTP	Real Time Protocol	
PPPOE	Point-to-point Protocol over Ethernet	
QoS	Quality of Service	
UPnP	Universal Plug and Play	
VLAN	Virtual Local Area Network	
NTP	Network Time Protocol	
STUN	Simple Traversal of UDP over NAT	
PSTN	Public Switched Telephone Network	

