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.

Contents

1 Product Introduction

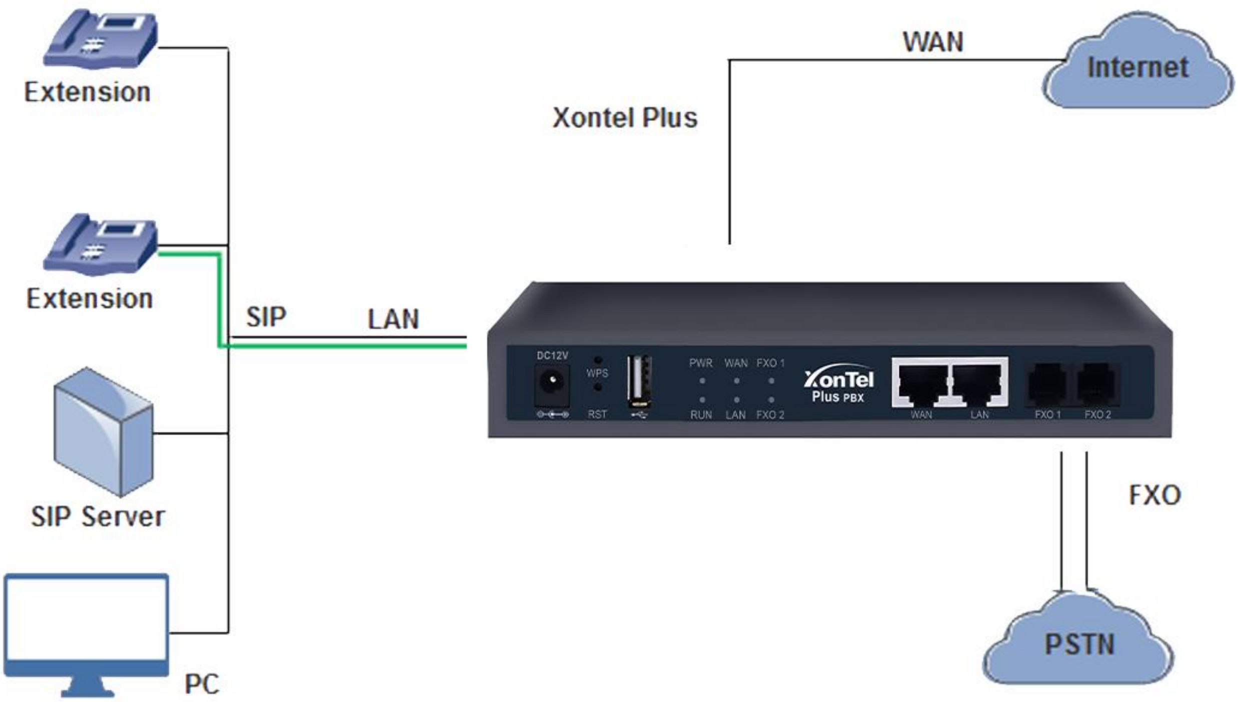
1.1 Overview

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:



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.
RUN	Running Indicator	Slow Flashing	The device is initialized successfully and is running normally
		On	The device is being initialized.
		Off	The device is not running normally.
FXO	FXO In-use Indicator	Fast Flashing	The FXO port is connected with PSTN line and is in idle status
		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.
WAN/LAN	Network Connection Indicator	Off	Network does not work or network cable is not connected to the WAN/LAN port.
		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

1.5.7 Environmental

●Power Supply: 12VDC, 2A

●Power Consumption: 18W

●Operating Temperature: 0 ℃ ~ 45 ℃

Storage Temperature: -20 ℃~80 ℃

●Humidity: 10%-90% (Non-Condensing)

●Dimensions：260×180×35mm（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

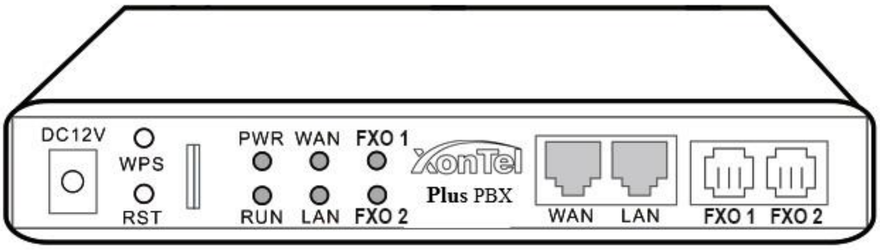
2 Quick Installation

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



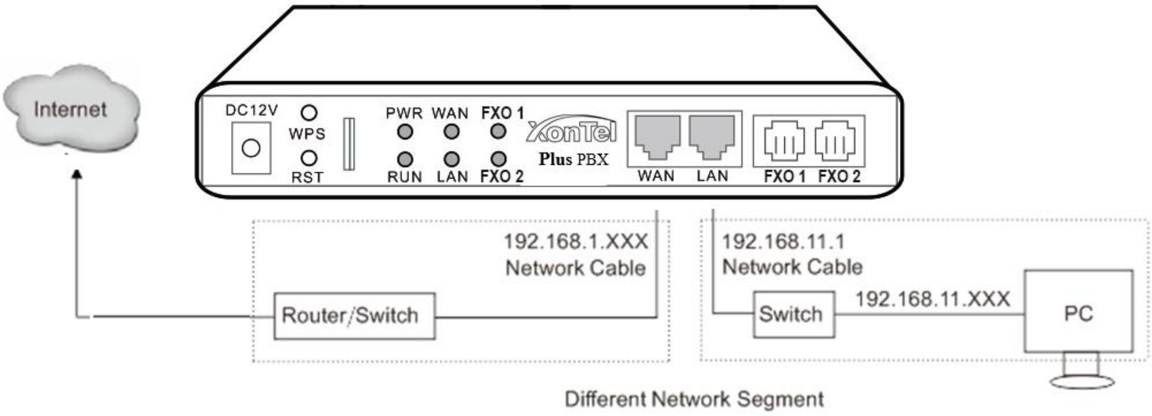
- Connect the power adapter to the power jack;
- Connect PSTN lines to the FXO ports;
- Connect network cable to the LAN port(s) and WAN port (please refer to 2.3 Network Connection);

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.

2.3.1 Network Connection Diagram under Route Mode

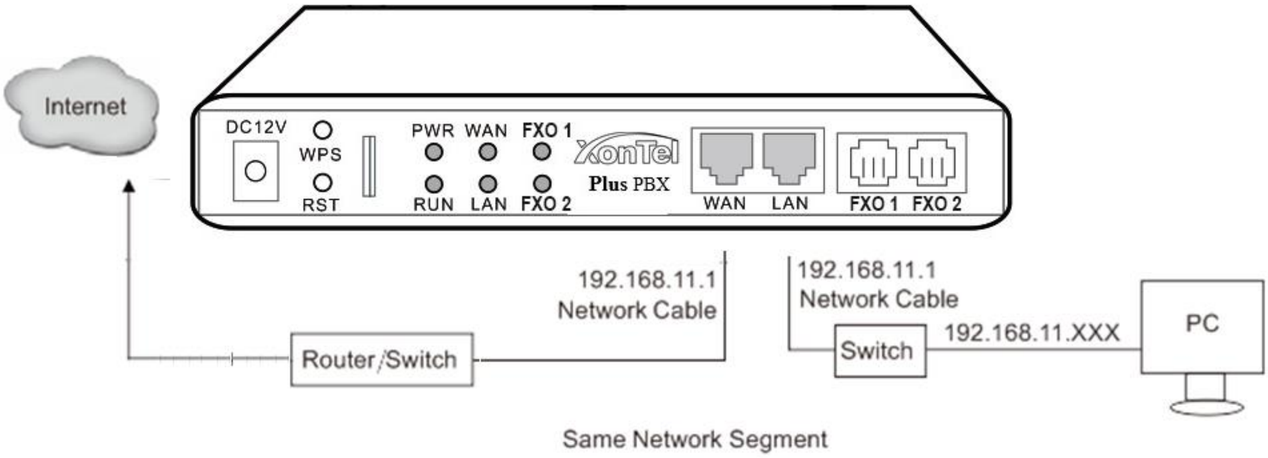
Under the route mode, the default IP address of WAN port is a DHCP IP address, while the default IP address of the LAN port is a static IP address, namely **192.168.11.1**.



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.

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**.

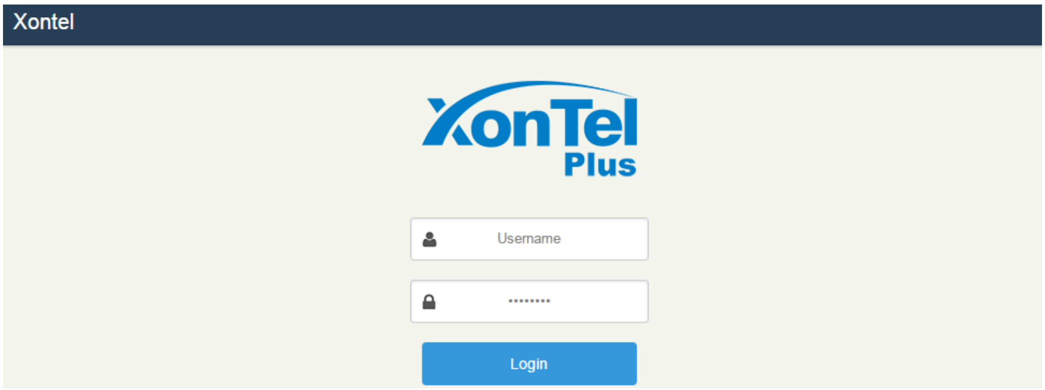
Check the connectivity between the PC and the XonTel Plus. Click **Start → Run** of PC and enter **cmd** to execute '**ping 192.168.11.1**' to check whether the IP address of LAN port runs normally.

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.



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 →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.

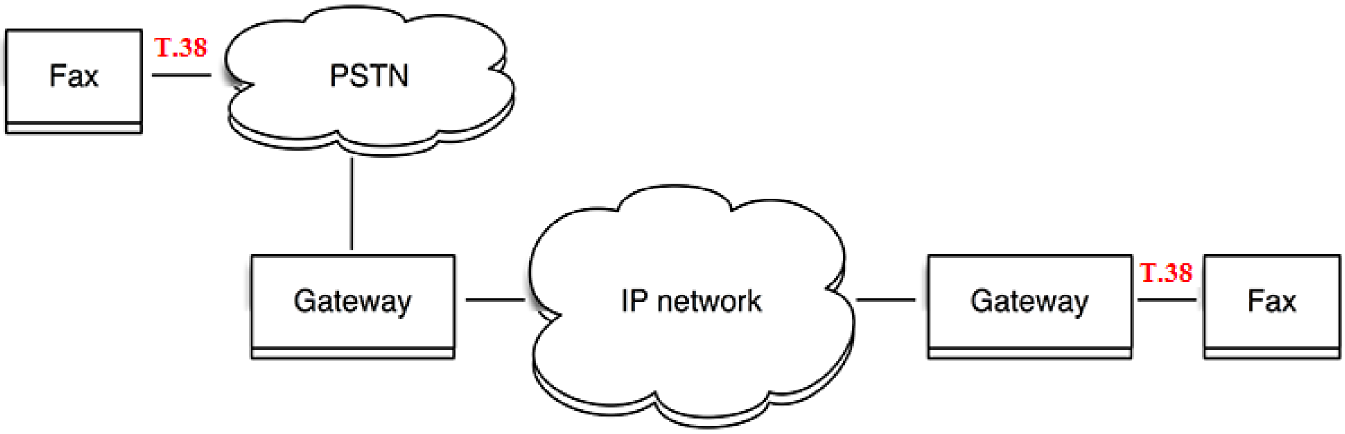
3.7 Send or Receive Fax

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.

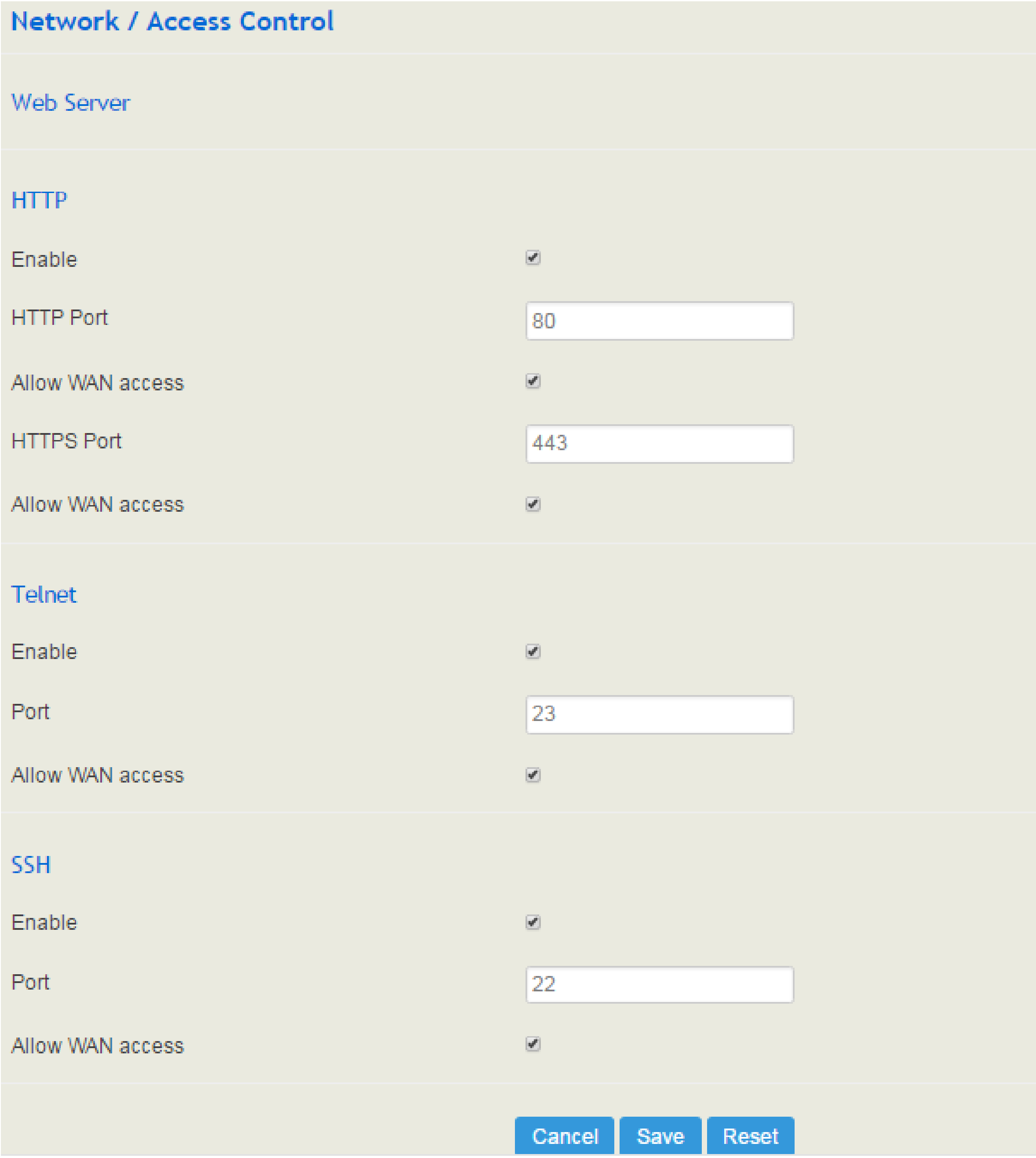


Pass-through: Under the pass-through mode, fax signal is not converted and fax traffic is carried in RTP packets. It uses the G.711 A or G711U codec in order to reduce the damage to fax signal.

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.



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.

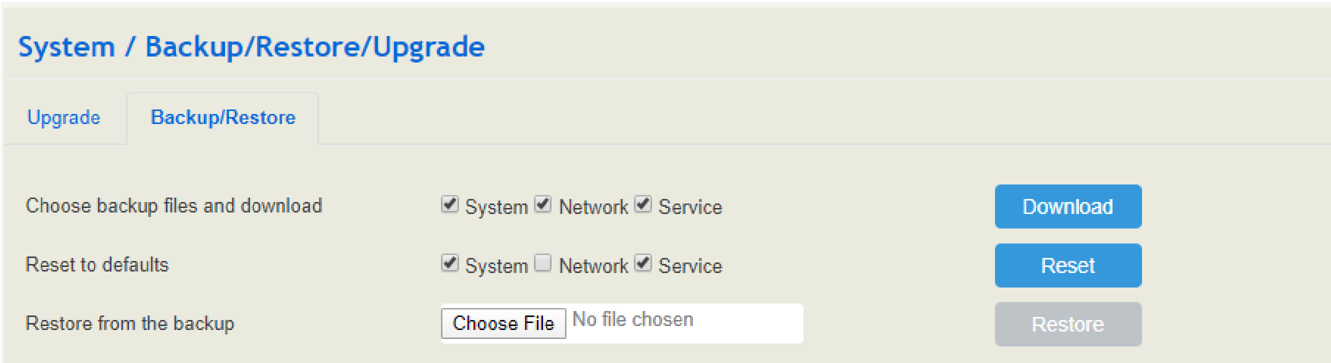
3. On the condition that the device is powered off, press the RST button and the WPS button, and connect the XonTel Plus PBX with power source. After about 30 seconds, the device will wipe out all configurations, rebuild a file system and then re-load a firmware version (this method is used in case of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



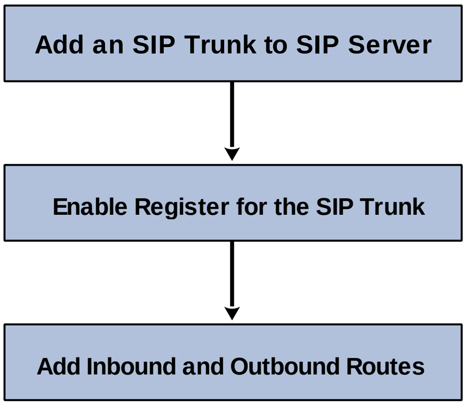
4 Configuration Wizard

4.1 Configuration Wizard

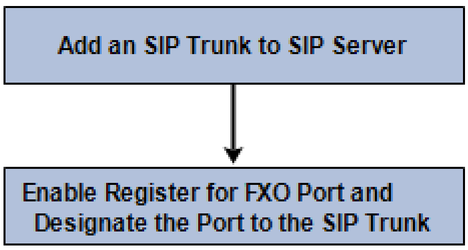
The following are the common ways to configure the XonTel Plus PBX.

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

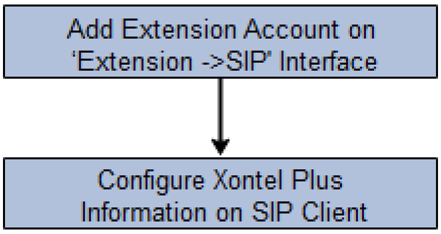


Note: Although 'Register' has been enabled for FXO port, calls through FXO port will take inbound and outbound routes as first priority. For outgoing calls, if outbound route cannot be matched, then the registered SIP trunk will be selected. For incoming calls, if inbound route cannot be matched, then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

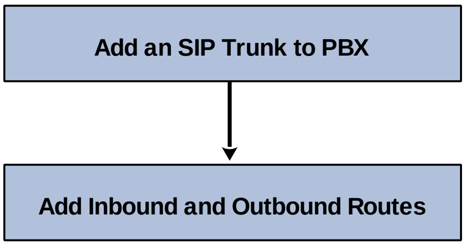
Generally, local extension number is taken as first priority for call routing selection, followed by DID, route and then registered port.

4.1.2 Other SIP Clients registered to XonTel Plus

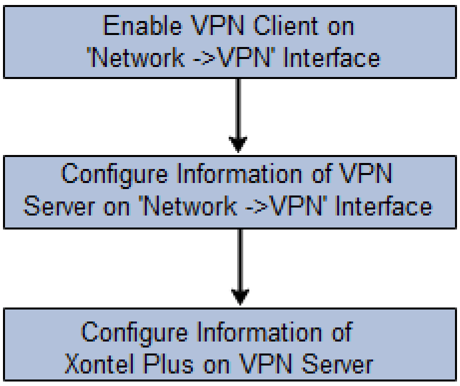
Under this mode, XonTel Plus is regarded as a SIP Server. Create an extension account first on the **Extension →SIP** interface, and configure listening port on the **Profile → SIP** interface. Then, configure the IP address, extension account and listening port of XonTel Plus on SIP client.



4.1.3 XonTel Plus Connected to PBX through Trunking



4.1.4 XonTel Plus Serving as VPN Client



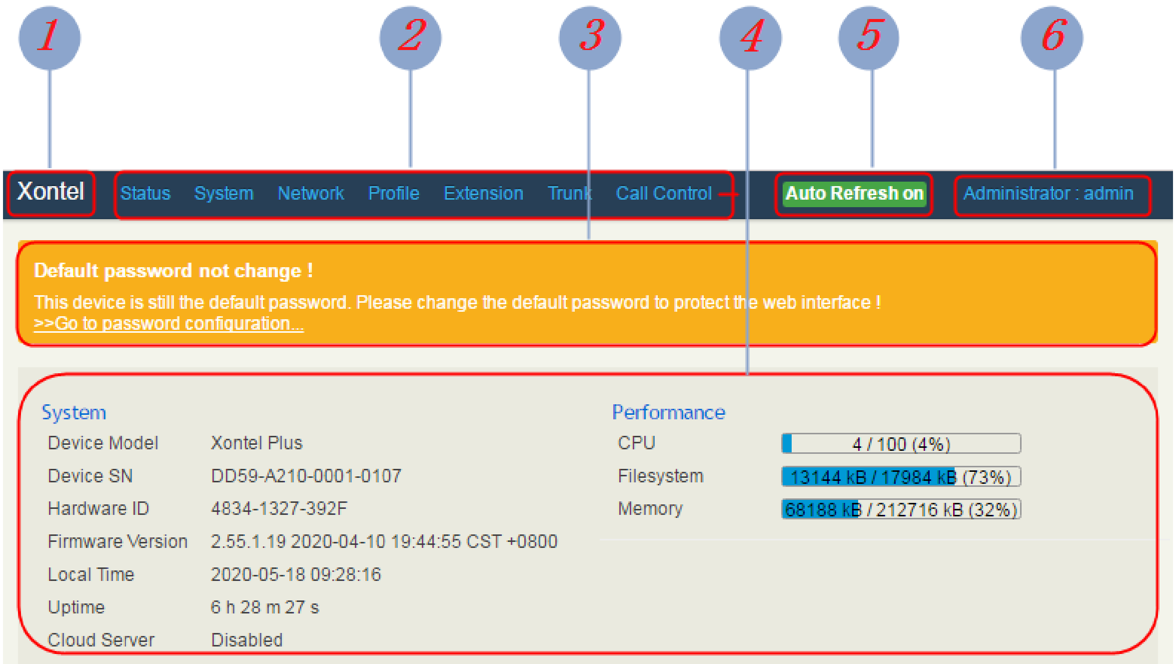
5 Configurations on Web Interface

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:



Index	Item	Description
1	XonTel	The name of the PBX; it can be edited on the **System → Setting** 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 **Apply** to enter into the page to save the changes; click **Revert** to return to original configuration.
4	Detailed Interface	The detailed configuration interface or display interface

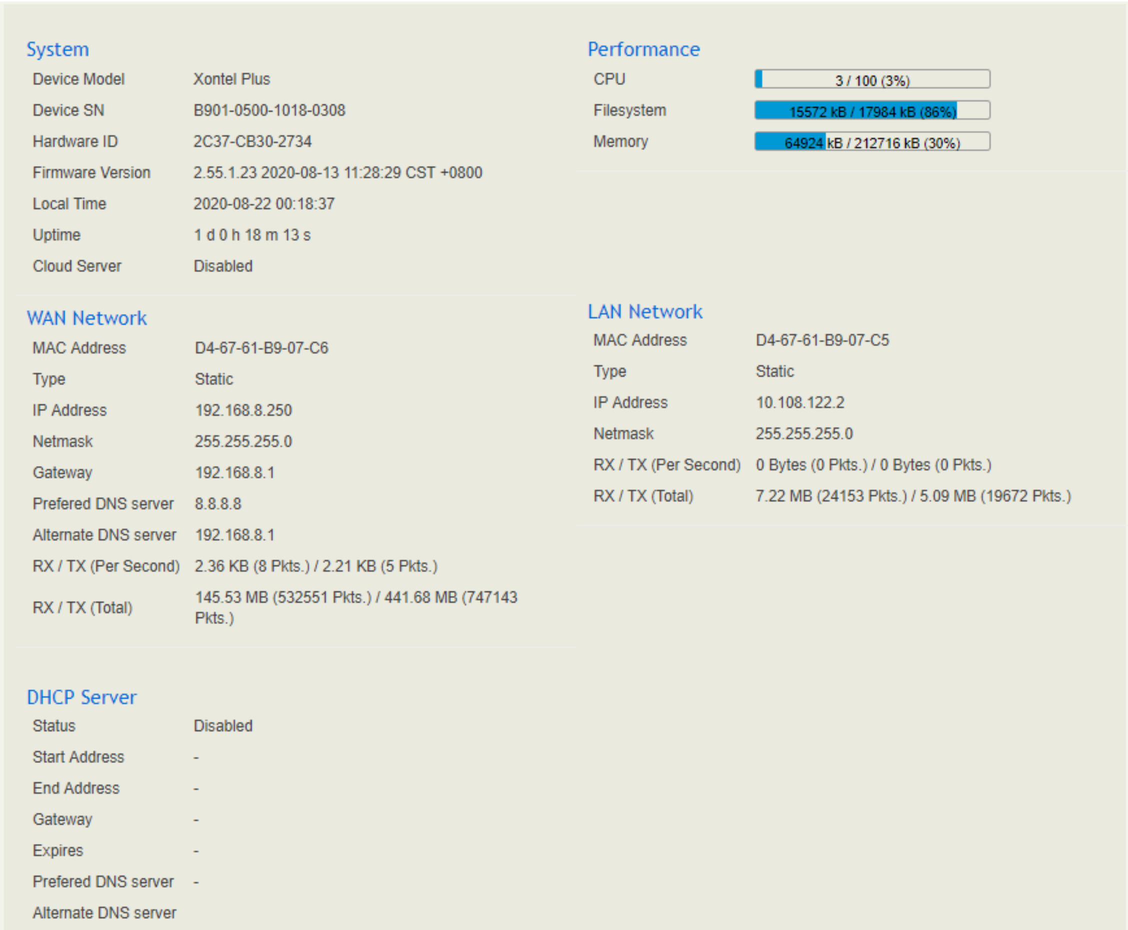
5	Auto Refresh Button	The button can be enabled or disabled. If it is enabled, the information on the **Status → Overview/SIP/PSTN/Current Call** interfaces will be refreshed automatically
6	User Role	The role of the current user logging into the Web. And the "exit" sign will pop up when the mouse moves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

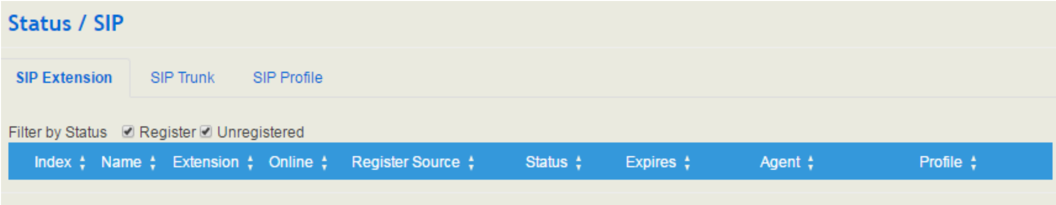
5.2.1 Overview

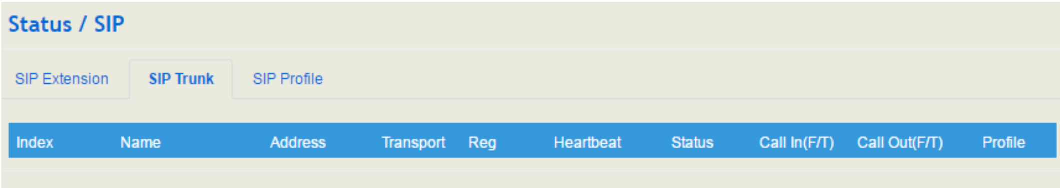
Log in the Web interface of XonTel Plus, click **Status → Overview**, and the following interface will be displayed. On the interface, device model, firmware version as well as information about performance are shown, together with WAN network, LAN network and DHCP server.

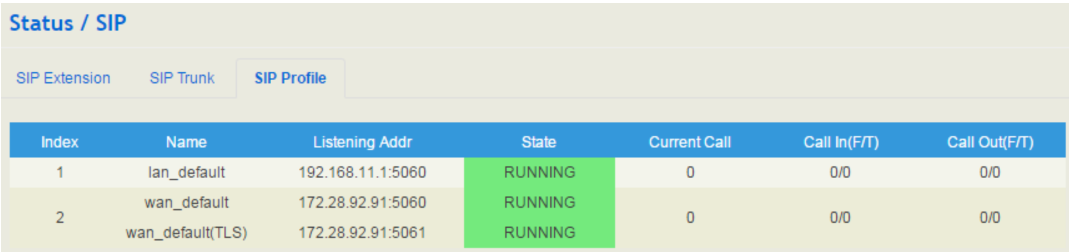


5.2.2 SIP

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



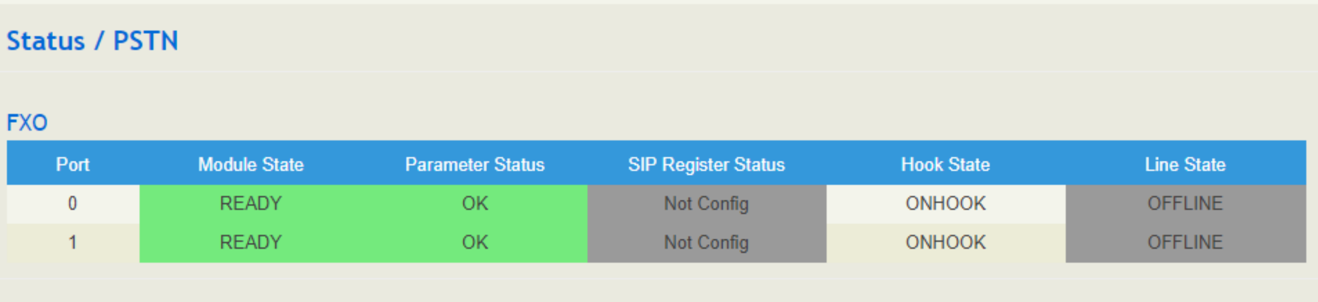




Belong To	**Parameter**	**Explanation**
SIP Extension	Filter by Status	You can choose Register or Unregister to filter SIP extensions
	Profile	The profile that is used by the SIP extension
	Status	SIP extension is registered or not. There are two statuses: Registered/Unregistered
SIP Trunk	Heartbeat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

5.2.3 PSTN

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


Status / PSTN

FXO

Port	Module State	Parameter Status	SIP Register Status	Hook State	Line State
0	READY	OK	Not Config	ONHOOK	OFFLINE
1	READY	OK	Not Config	ONHOOK	OFFLINE

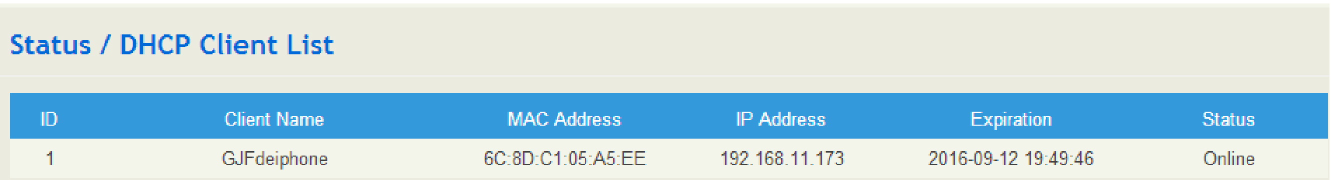
If 'SIP Register Status' is 'Registered', it means FXO has been **registered to SIP server** on the **Trunk → SIP/FXO** interface respectively.

Belong To	**Parameter**	**Explanation**
FXO	Module Status	There are two module statuses: Ready and Config Failed
	Parameter Status	There are two parameter statuses: OK and error
	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.

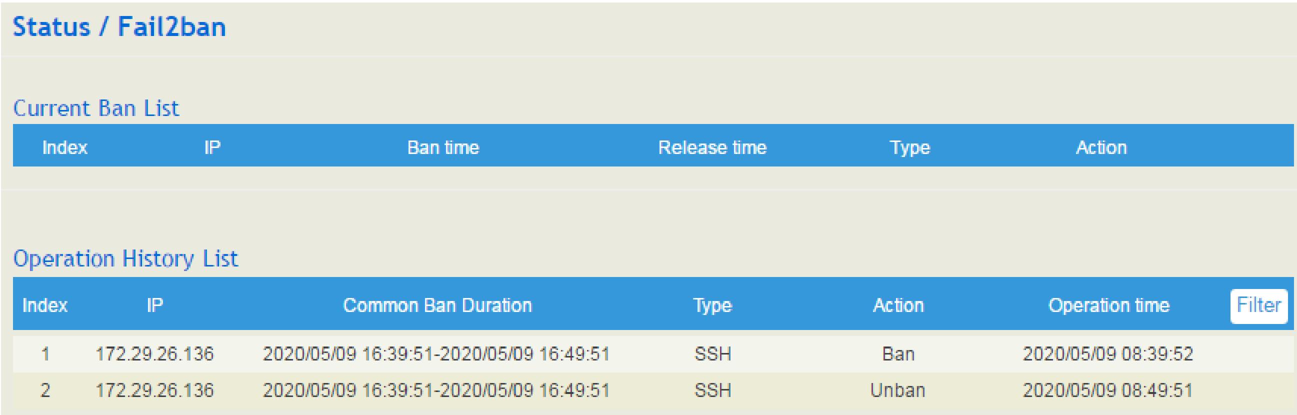

Status / DHCP Client List

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 →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.


Status / Fail2ban

Current Ban List

Index	IP	Ban time	Release time	Type	Action

Operation History List

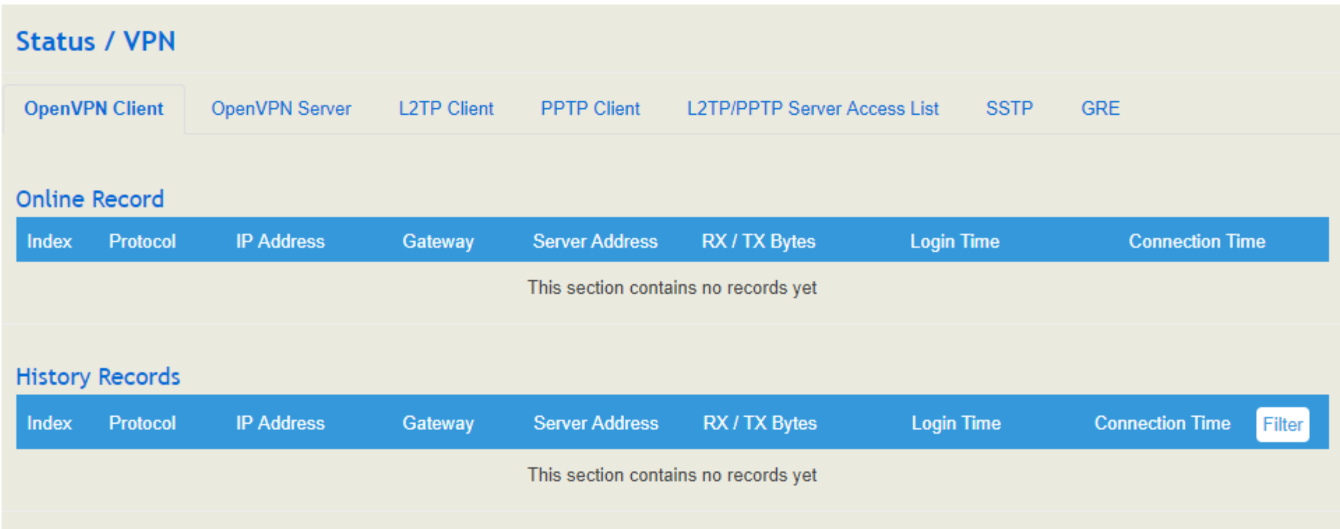
Index	IP	Common Ban Duration	Type	Action	Operation time	Filter
1	172.29.26.136	2020/05/09 16:39:51-2020/05/09 16:49:51	SSH	Ban	2020/05/09 08:39:52	
2	172.29.26.136	2020/05/09 16:39:51-2020/05/09 16:49:51	SSH	Unban	2020/05/09 08:49:51	

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

5.2.6 VPN

On the **Status → 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 → VPN ->OpenVPN Server** or **Status -> VPN -> L2TP/PPTP Server Access List** interface.


Status / VPN

OpenVPN Client | OpenVPN Server | L2TP Client | PPTP Client | L2TP/PPTP Server Access List | SSTP | GRE

Online Record

Index	Protocol	IP Address	Gateway	Server Address	RX / TX Bytes	Login Time	Connection Time

This section contains no records yet

History Records

Index	Protocol	IP Address	Gateway	Server Address	RX / TX Bytes	Login Time	Connection Time	Filter

This section contains no records yet

5.2.7 Current Call

On **Status →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 / Current Call

Index	Src	Dest	Caller	Called	Start Time	Answer Time	State	Duration	Filter

5.2.8 Call Queue

On the **Status →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 Queue

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.

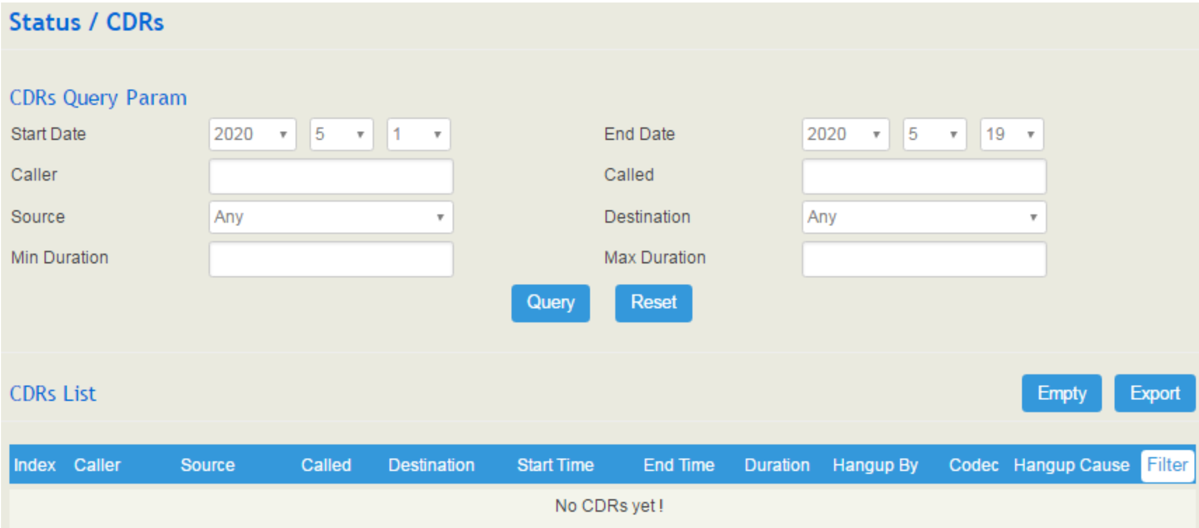
Status / Parking Lot

Index	Parking Number	Source	Duration

5.2.10 CDRs

Click **Status → 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.

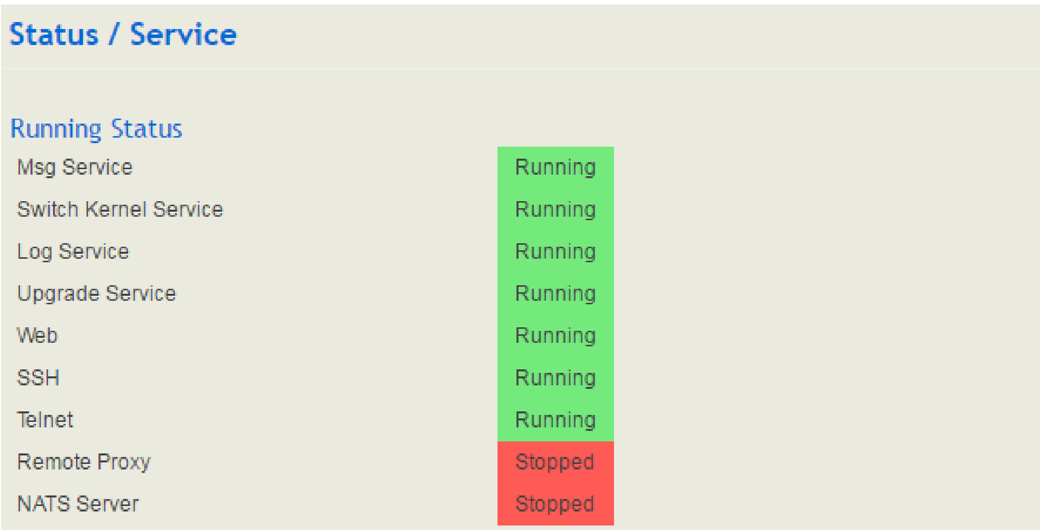
CDRs cannot be saved on the **Status → CDRs** interface unless the CDRs function has been enabled on the **System → Setting** interface.



5.2.11 Service

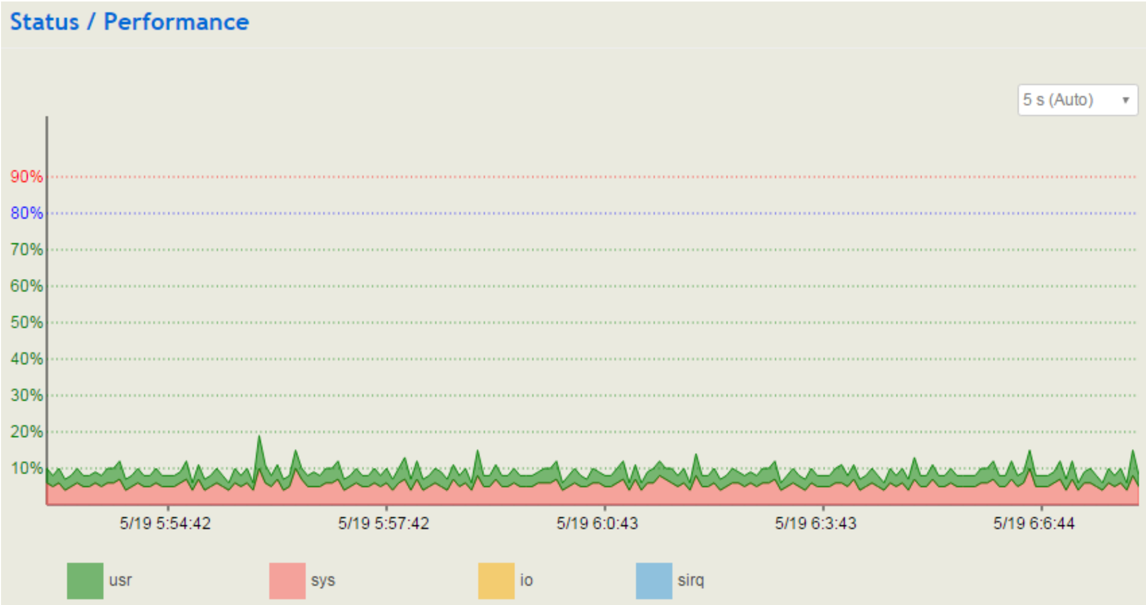
Click **Status → Service**, and the service status of XonTel Plus is displayed. This function is enabled by default. The Web, SSH and Telnet service can be disabled and their ports can be modified on the **Network → Access Control** interface. If no running status is shown, it means exception has occurred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

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



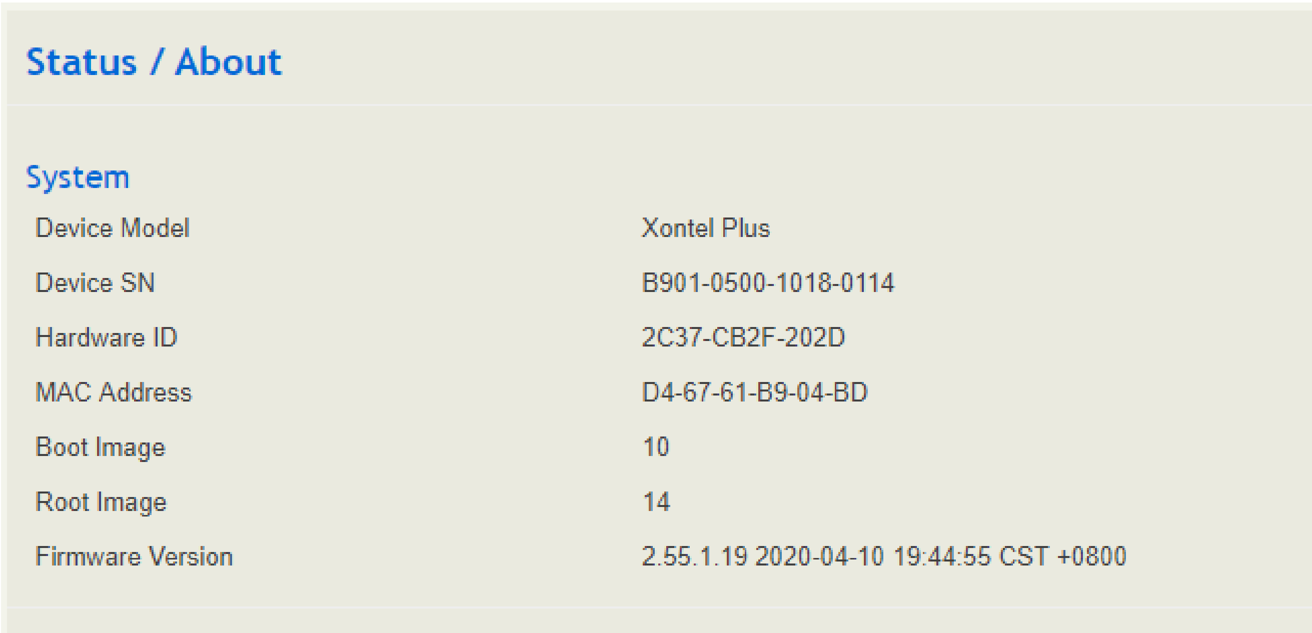
5.2.12 **Performance**

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



5.2.13 **About**

On the **Status → 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

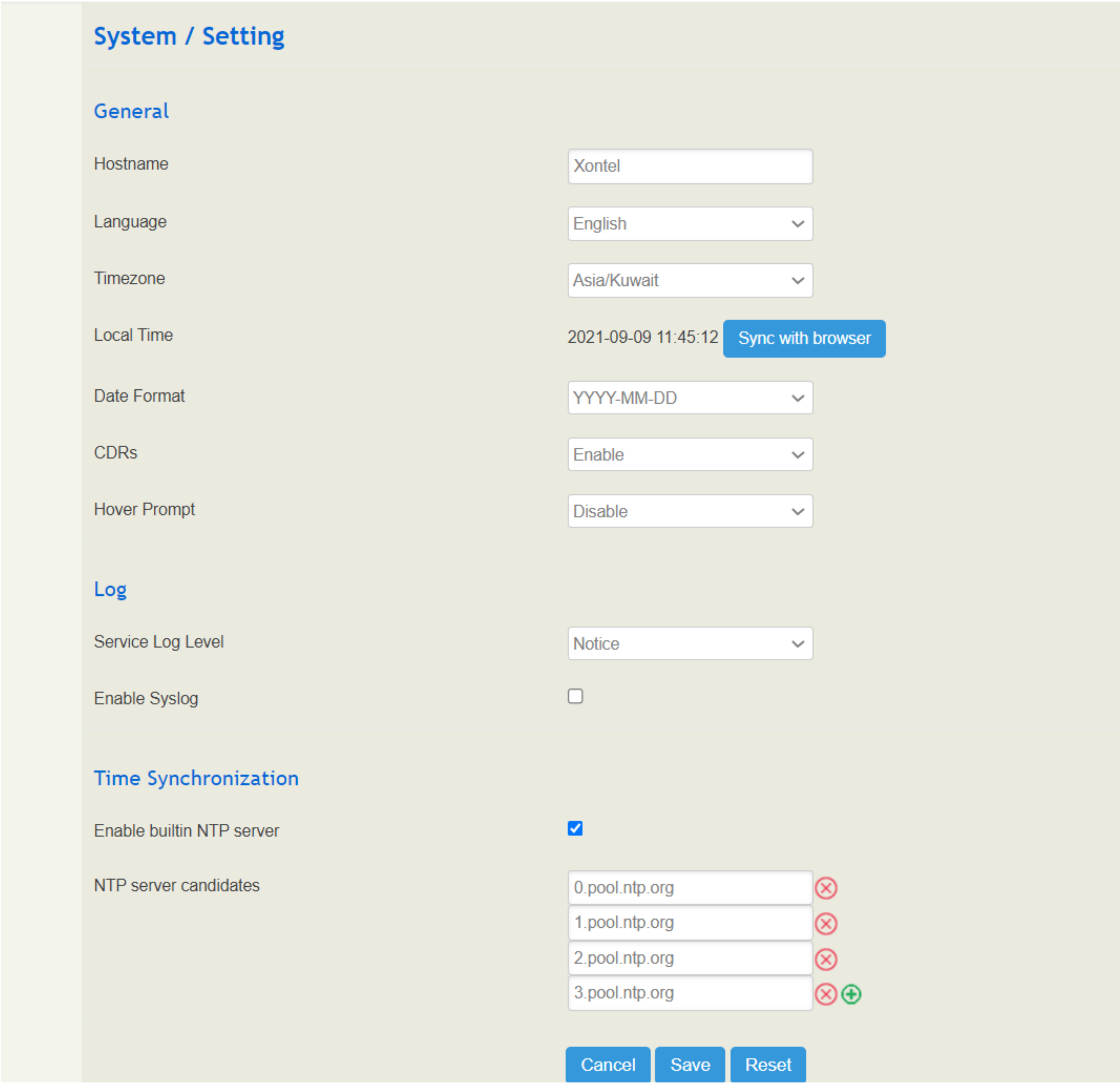
Device Model	Xontel Plus
Device SN	B901-0500-1018-0114
Hardware ID	2C37-CB2F-202D
MAC Address	D4-67-61-B9-04-BD
Boot Image	10
Root Image	14
Firmware Version	2.55.1.19 2020-04-10 19:44:55 CST +0800

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 → 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.



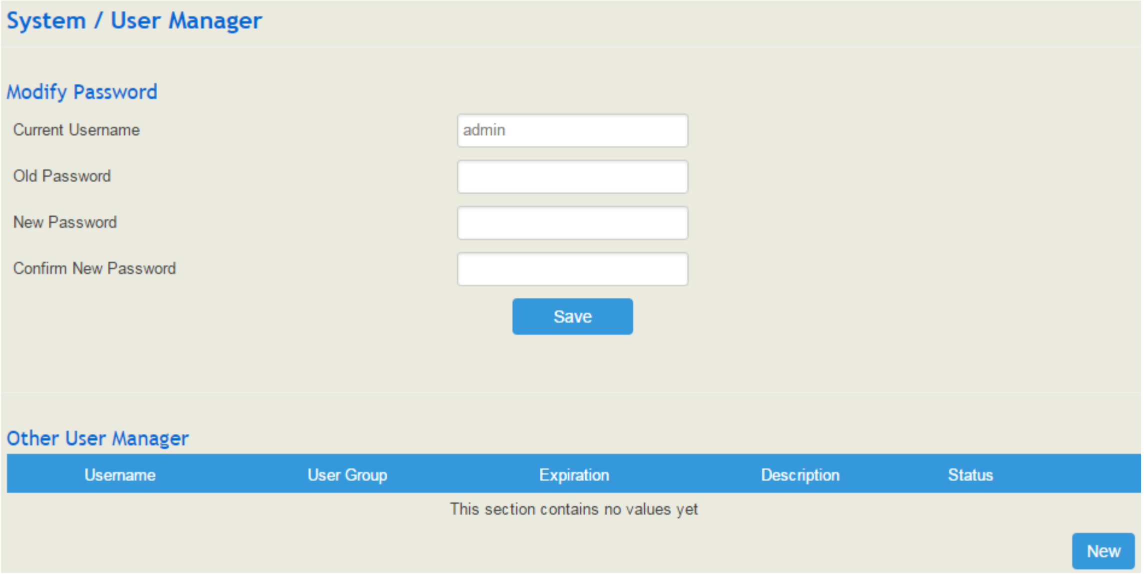
Parameter	Explanation
Hostname	The name of the PBX. After it is configured, the name will be displayed on the left of the menu bar.
Language	You can choose the language of XonTel Plus, the default value is English
Timezone	You can choose a time zone you want. The default value is Asia/Kuwait
Local Time	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 **Status → CDRs** 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.
	Delete a NTP Server
	Add a NTP Server

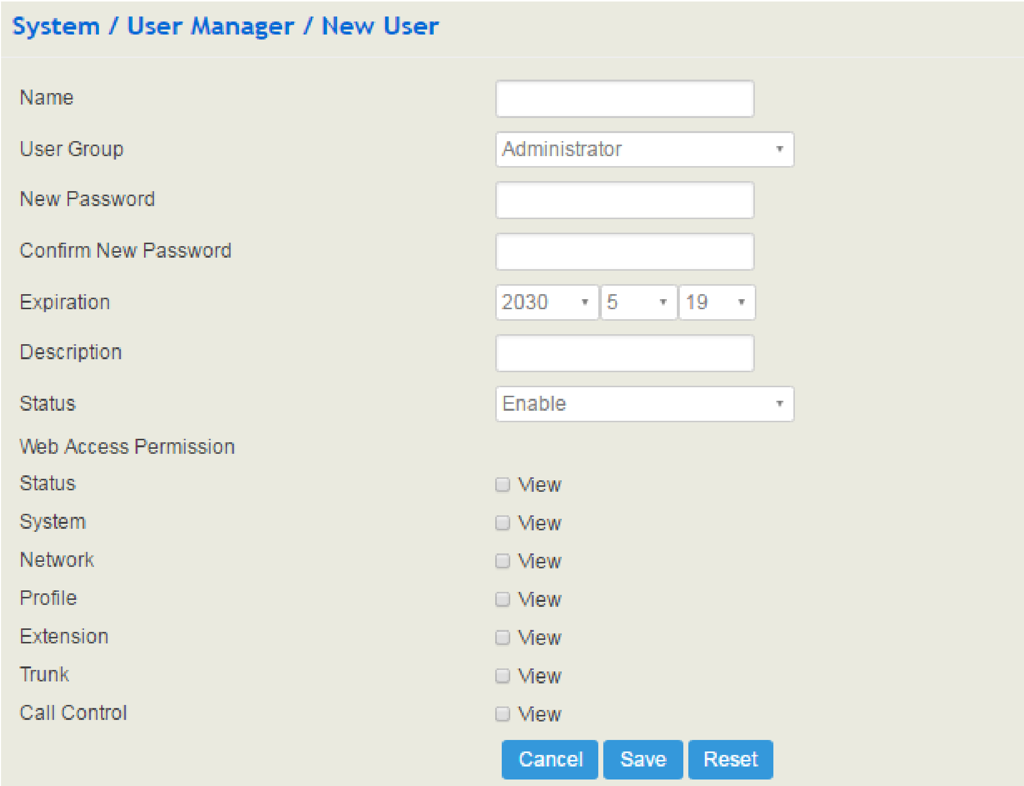
5.3.2 User Manager

Click **System → 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.





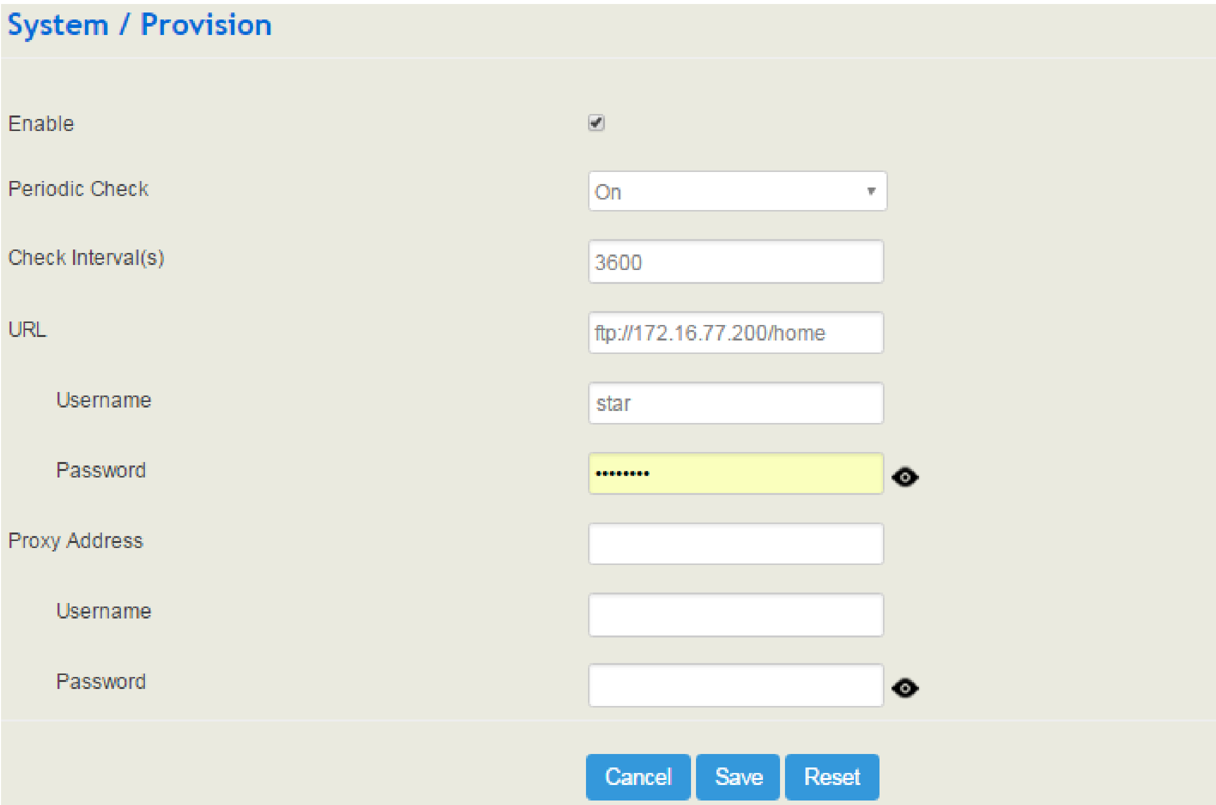
Parameter	Explanation
Name	The name of the new user. After it is established, the name and the password will be used to log into the web page of the device.
User Group	You can choose a role for the new user, such as administrator, operator and observer. The default value is administrator.
New Password	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:

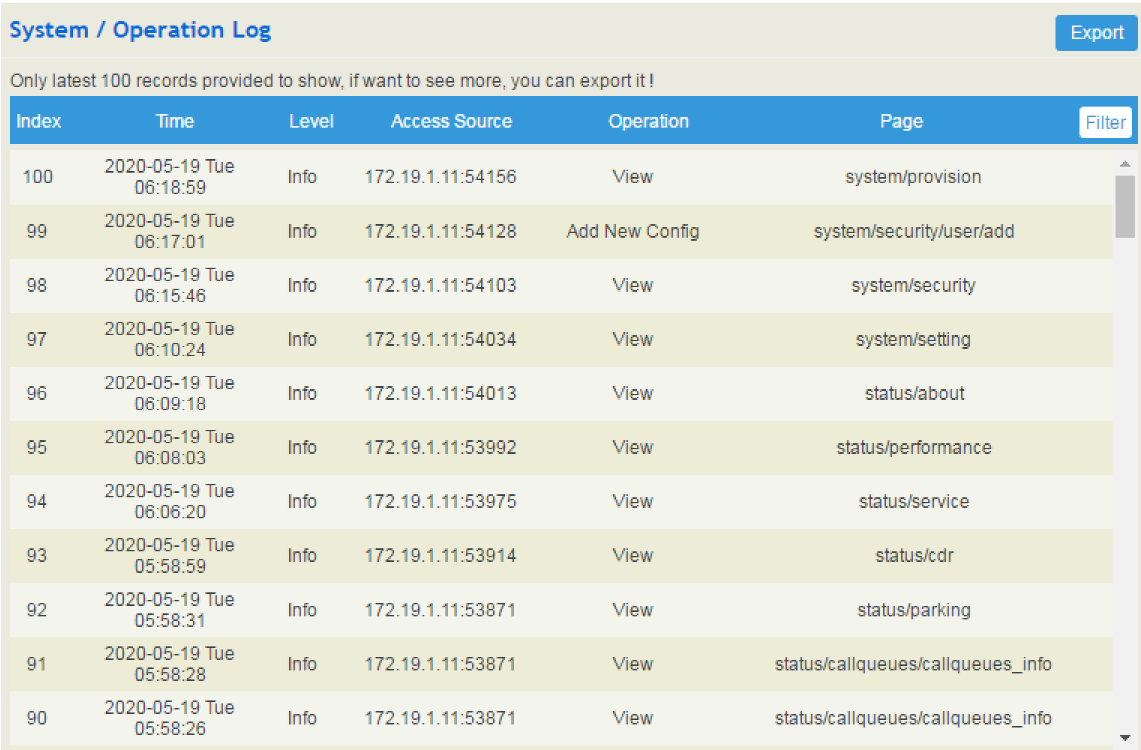


Parameter	Explanation
Periodic Check	Whether to enable periodic check. If it is enabled, the PBX will automatically check whether the firmware version stored on the URL is updated.
Check Interval	The interval to check whether the firmware version stored on the URL is updated. If it is 3600s, the PBX 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.



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 → Service Log** interface. Those logs are used for analyzing where a problem has occurred on the PBX.

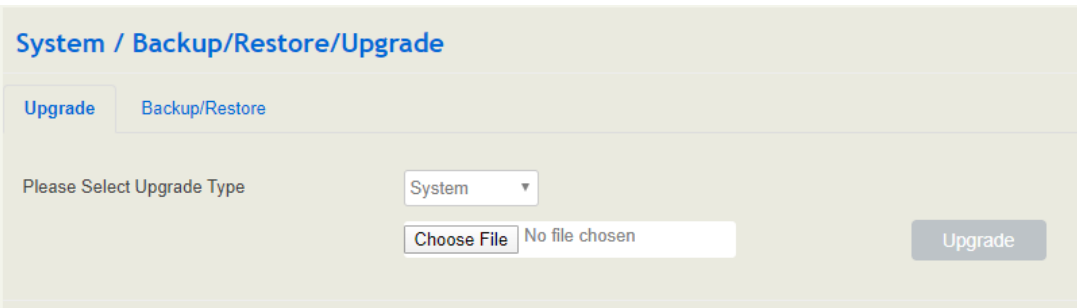
5.3.6 Config Changes Log

On the **System →Config Changes Log** interface, the configurations changed by administrator on the Web of the PBX are recorded.

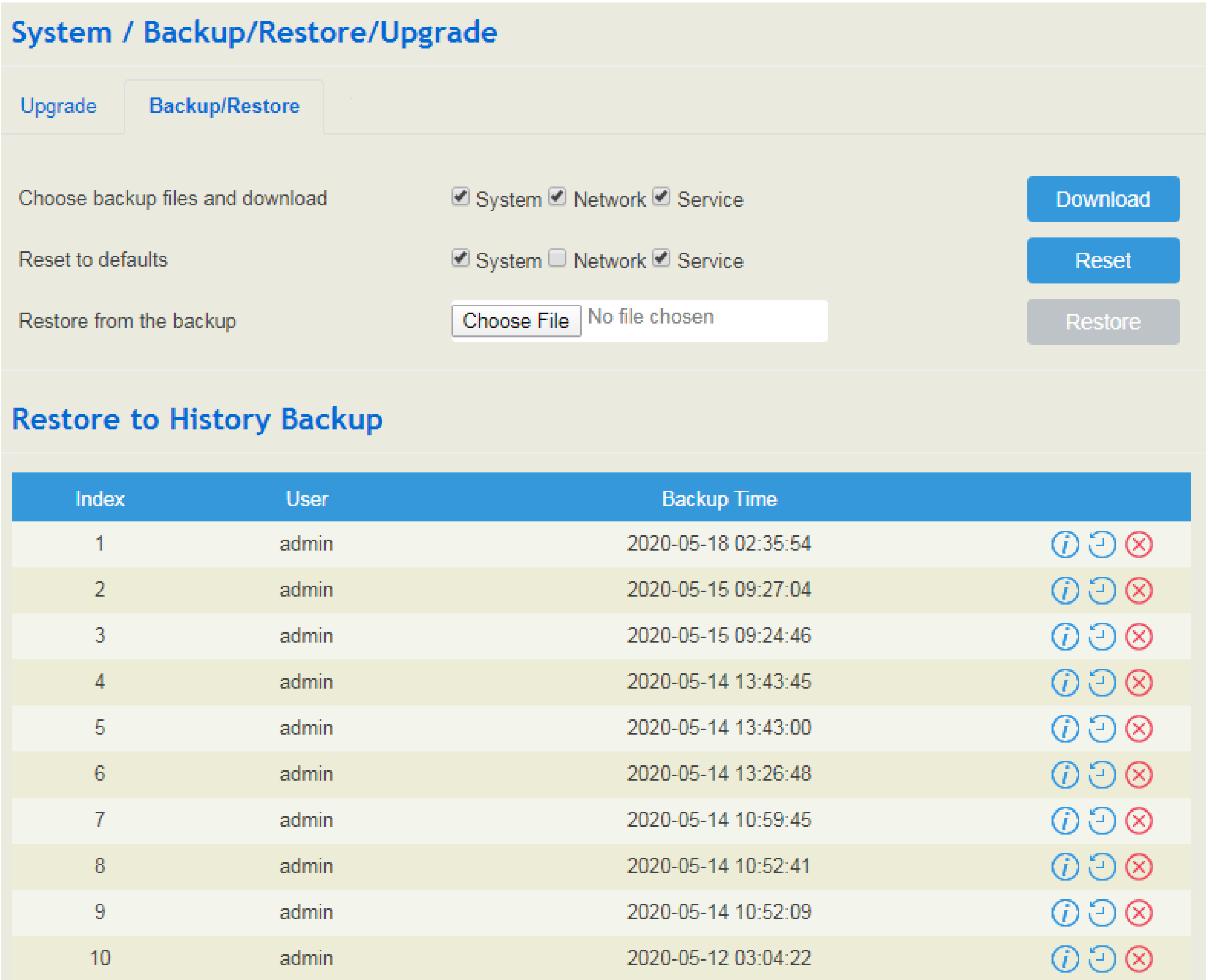
5.3.7 Backup/Restore/Upgrade

On the **System → 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



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.

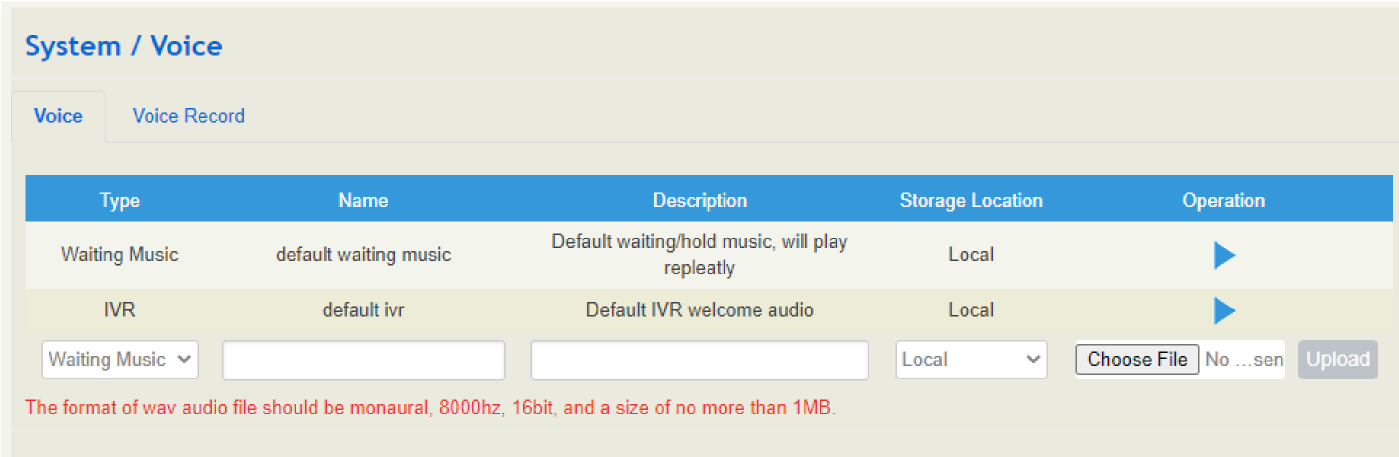


Explanation of Backup/Restore/Upgrade menu

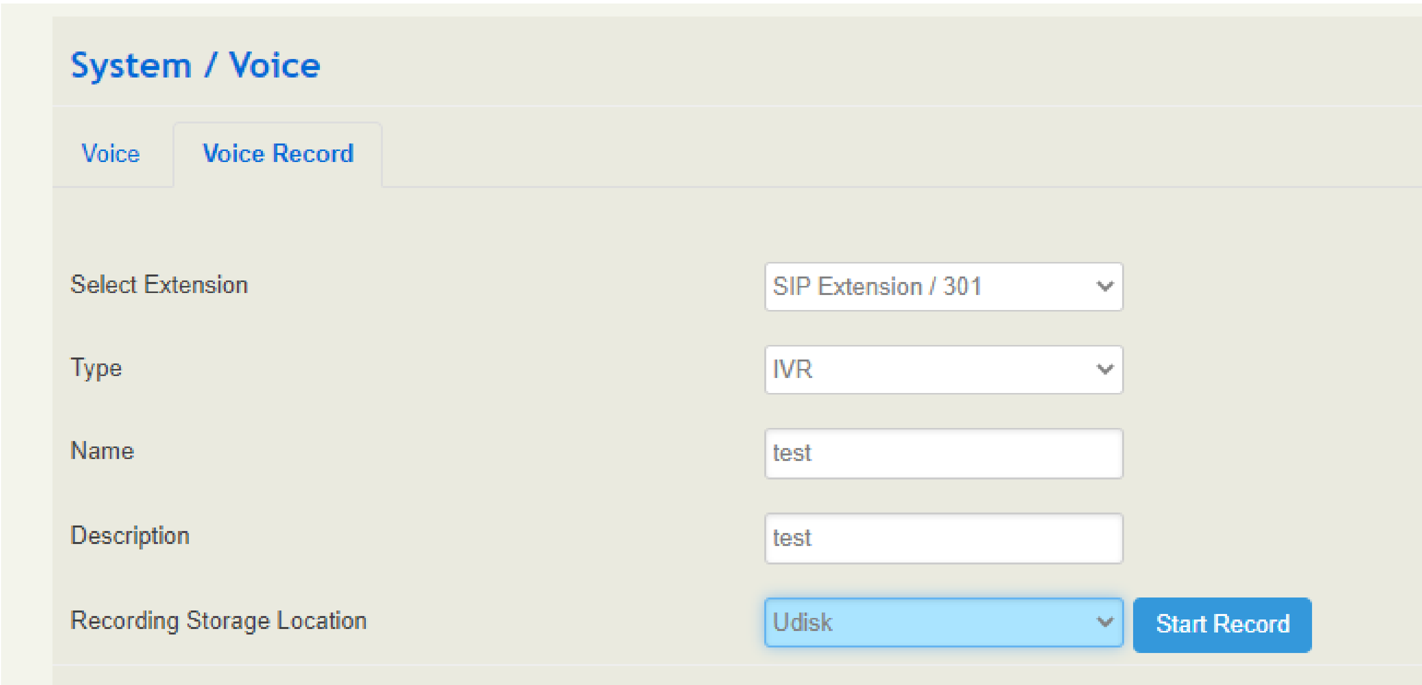
Upgrade	Choose a file to be upgraded (which is provided by XonTel), and then click **Upgrade**.
Download	You can download the configuration data to be backed up. Select any of the checkboxes on the left of System, Network and Service, and then click **Download**
Reset	Select any of the checkboxes on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

5.3.8 Voice

On the **System → Voice → 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.



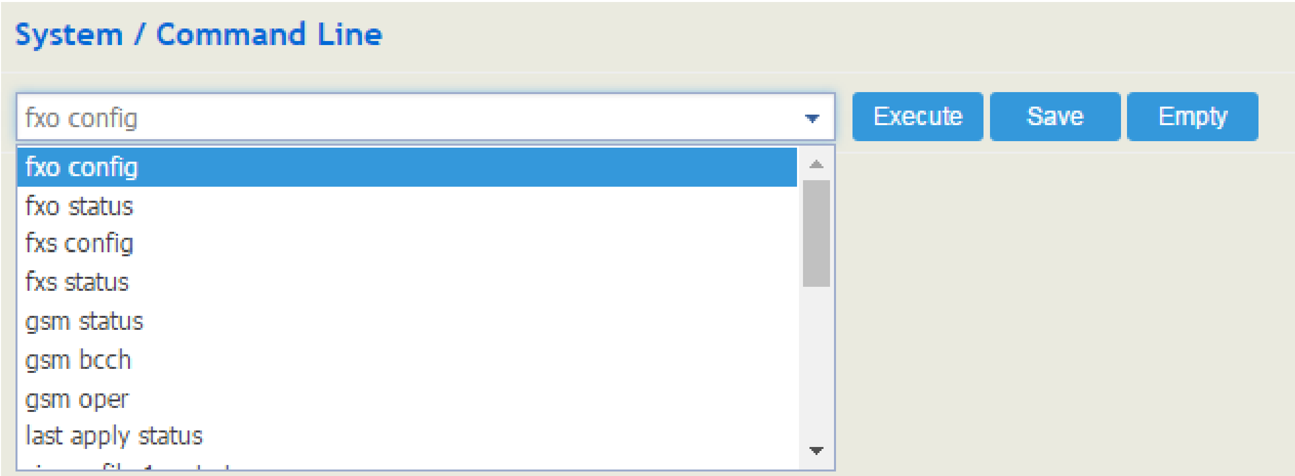
Also on **System → Voice → Voice Record** you can record the IVR from your extension, just select the extension that you want to record on it then click on Start Record to start record the IVR



5.3.9 Command Line

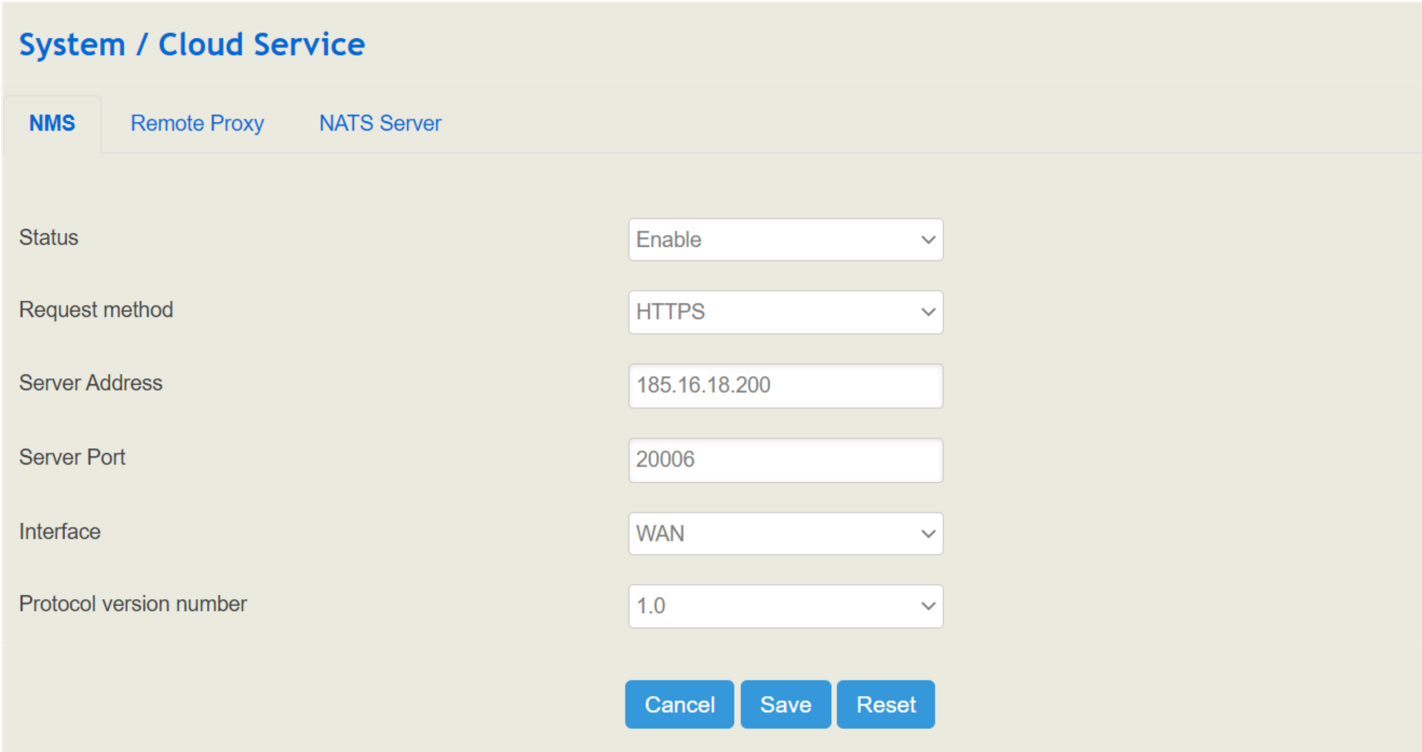
On the **System → Command Line** interface, some commonly-used command lines can be directly selected in the draw-down box, and therefore user has no need to enter command lines on Telnet. In this way, the efficiency of problem diagnostics is greatly improved.

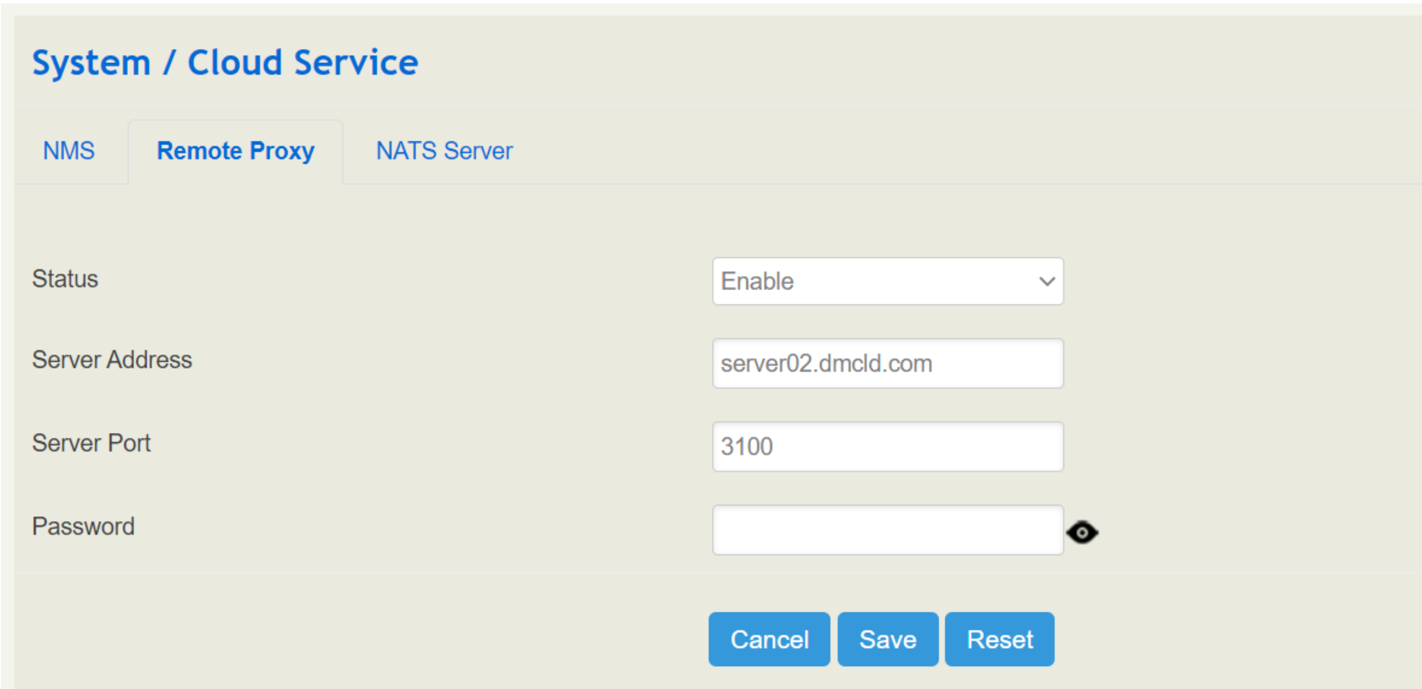
Commonly-used command lines include fxo config, fxo status, gsm status, gsm bcch, gsm oper, sip status, sip profile and so on.



5.3.10 Cloud Service

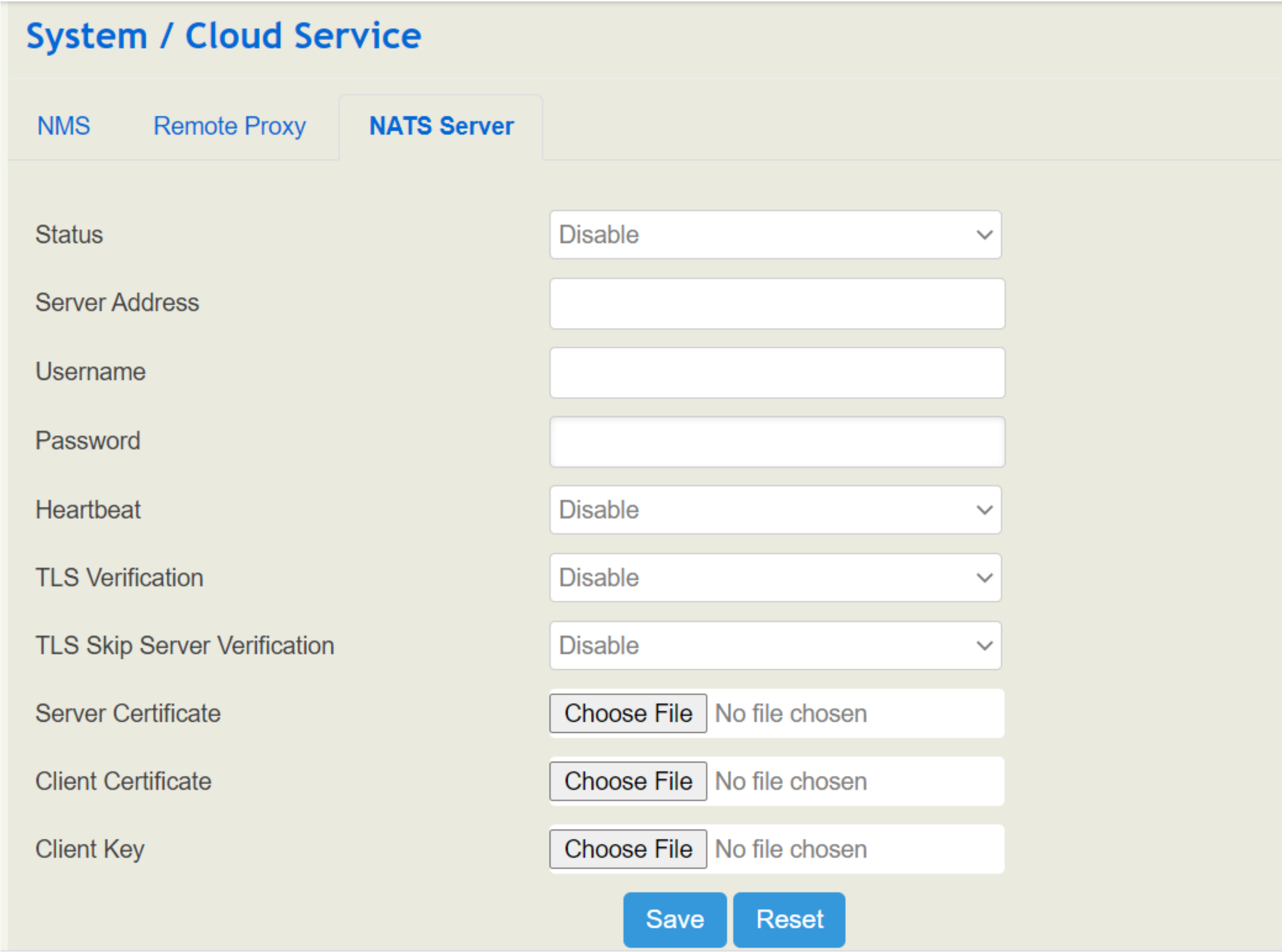
Cloud service is mainly used to centrally manage all kinds of devices. Through cloud service, you can query the status of a device, upgrade devices at batch, log in or configure a device remotely. The XonTel Plus PBX provides Cloud service. Enter the IP address, service port and password of the Cloud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





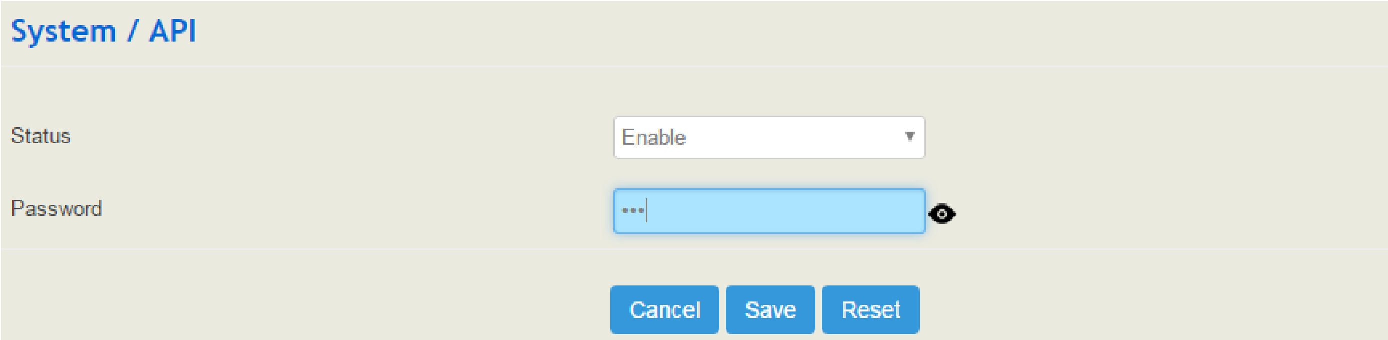
NATS Server:

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.



5.3.11 **API**

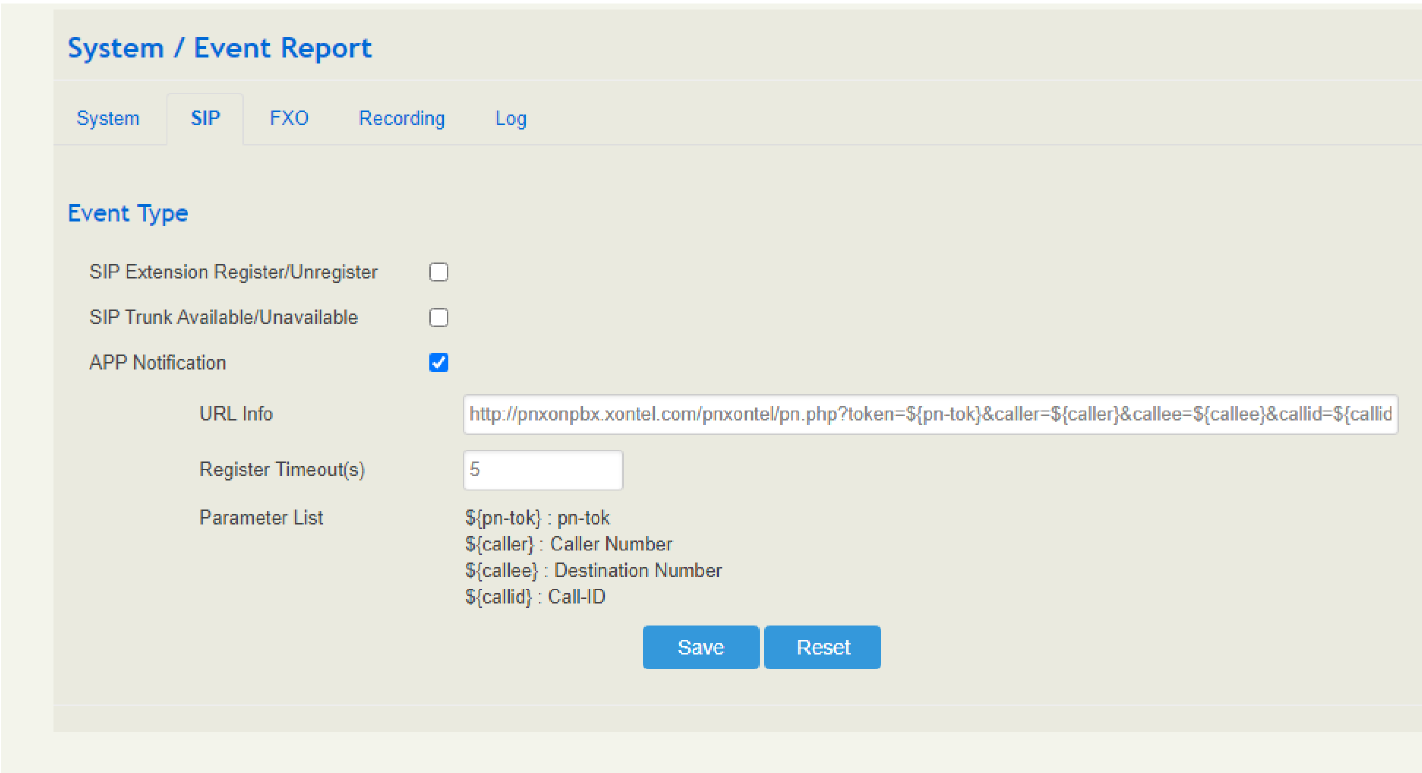
XonTel Plus provides API (Application Programming Interface) to interwork with other devices or platforms. This function enables you to centrally manage devices through command lines.



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.

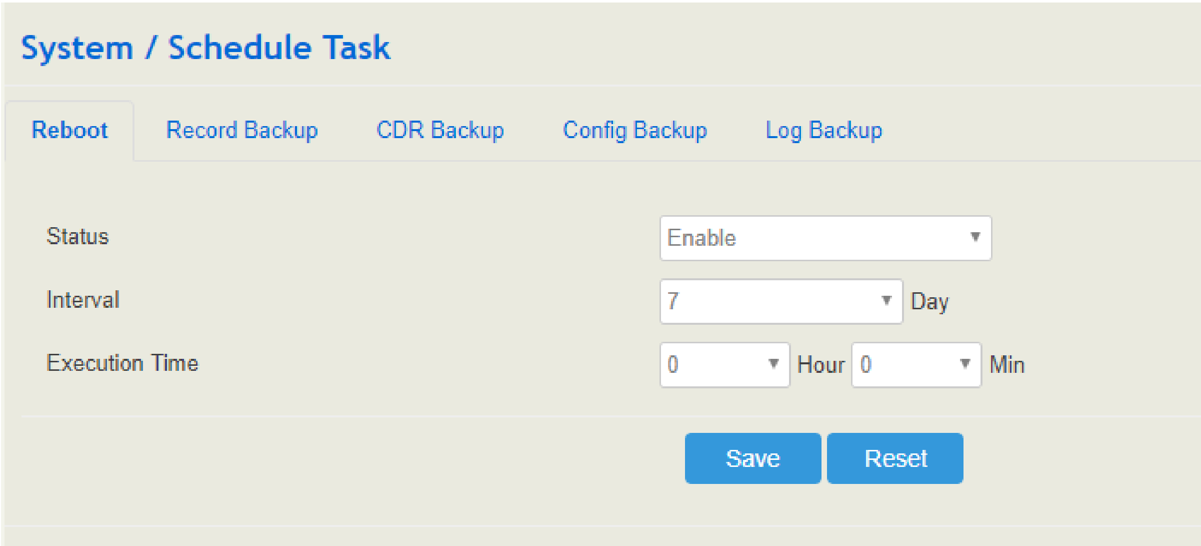
For event report through NATS, please refer to the configuration steps of NATS in the Could Server section.

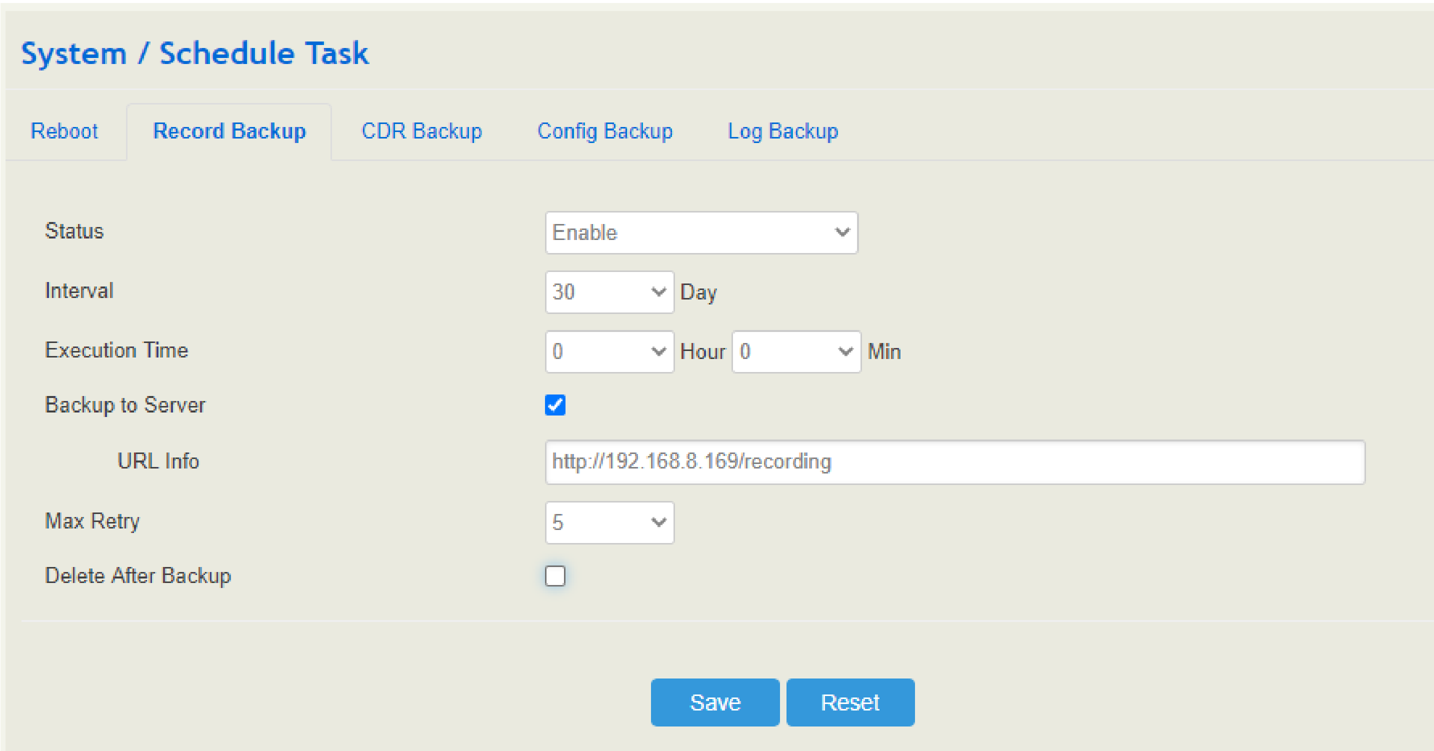


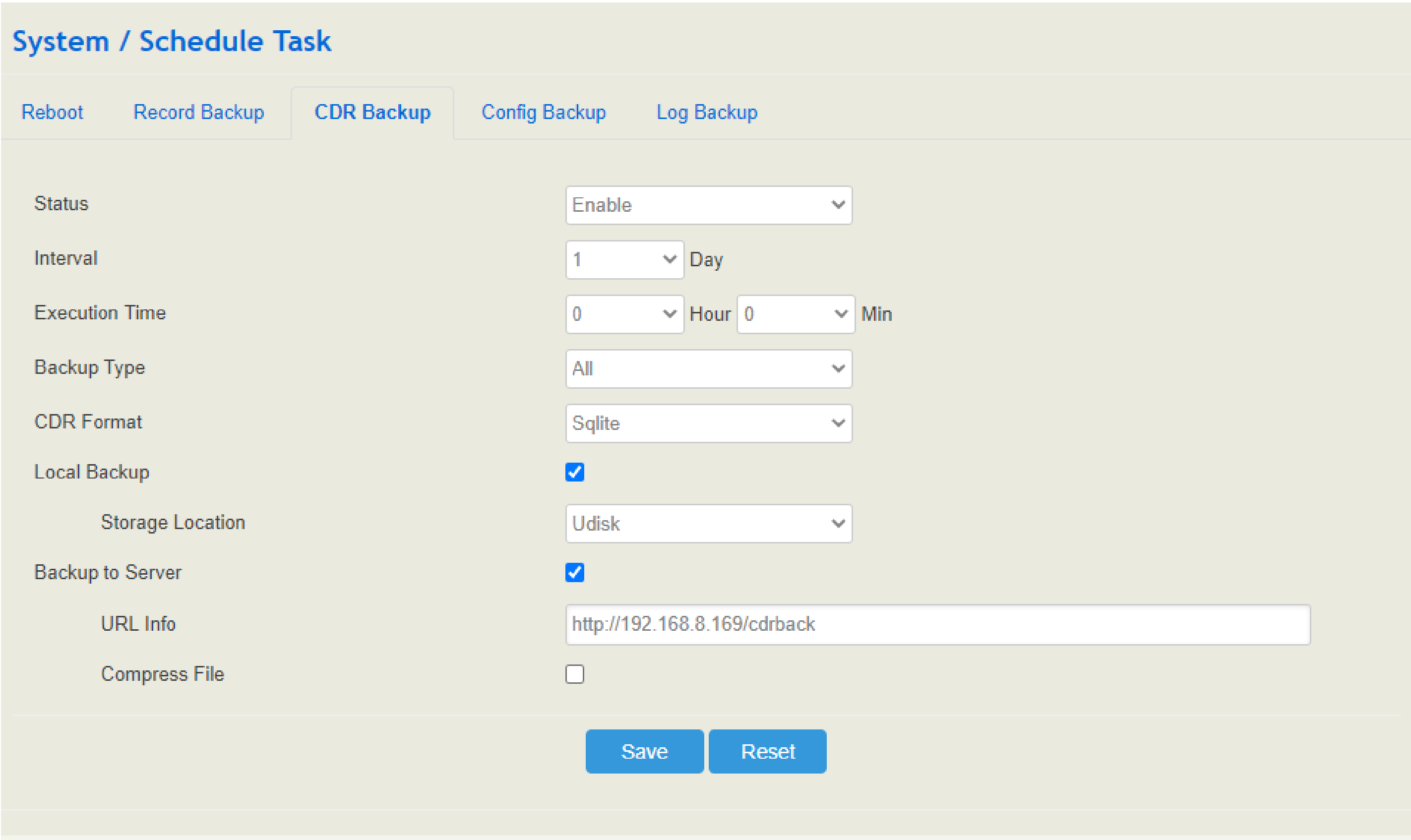
5.3.13 Schedule Task

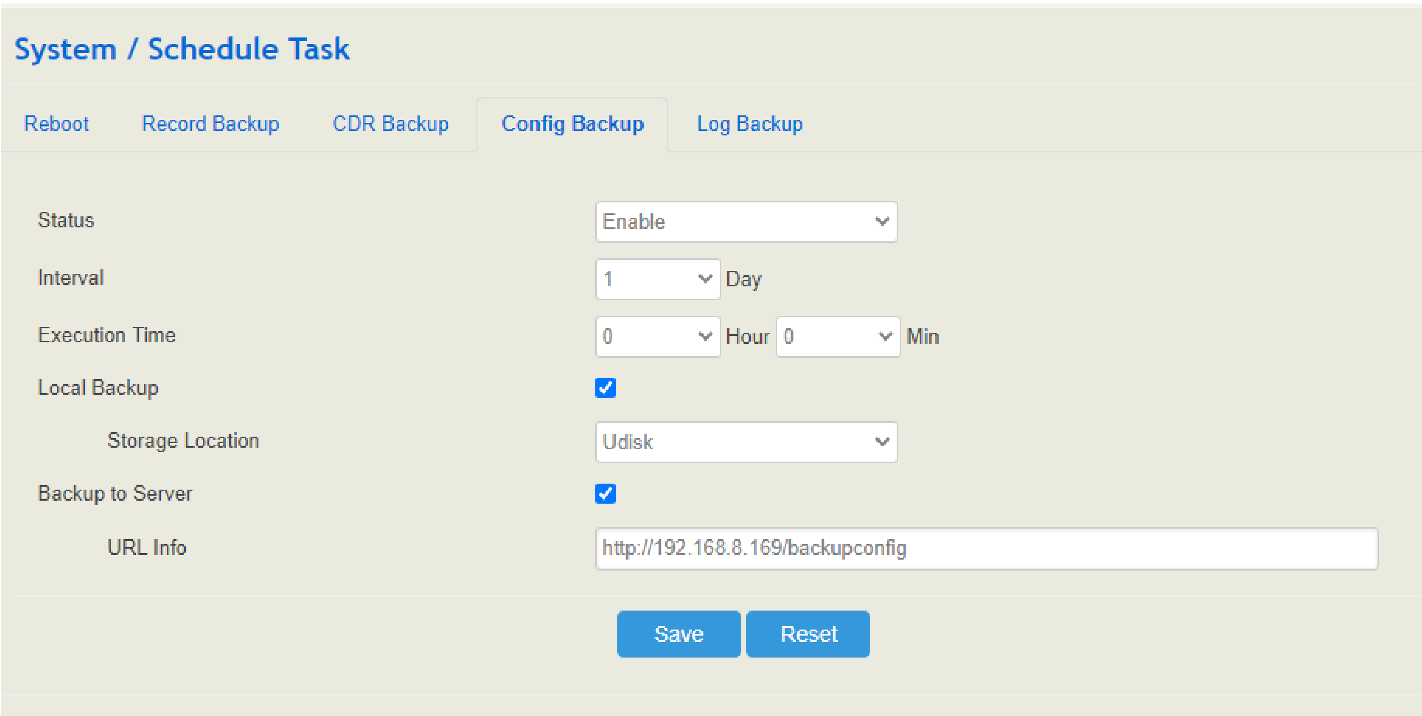
On the **System → Schedule Task** interface, you can set a scheduled time to reboot the XonTel Plus device, record backup, and back up CDRs, configuration backup or backup logs as shown in the figures below.

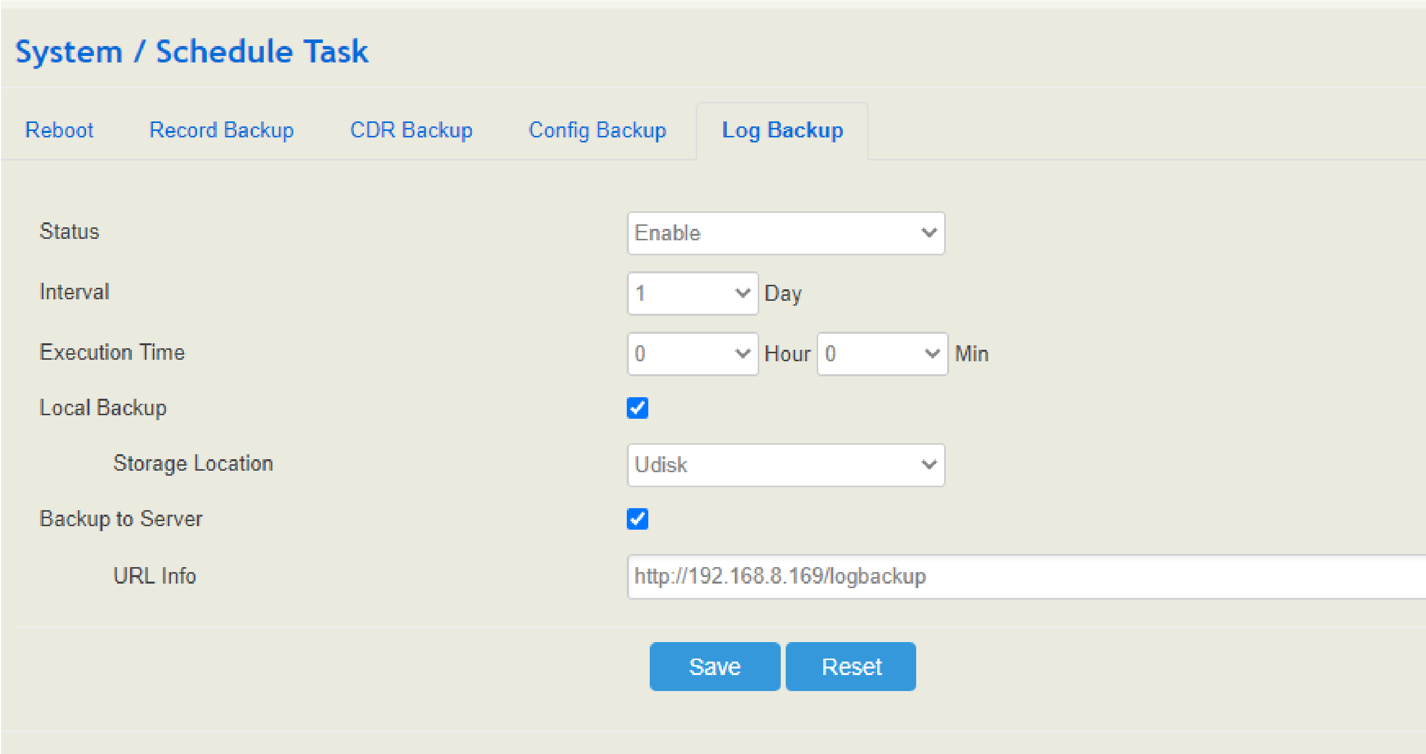
You can also make schedule backup to an http server (You can make your PC as http server by using http server application such as **hfs** software application).





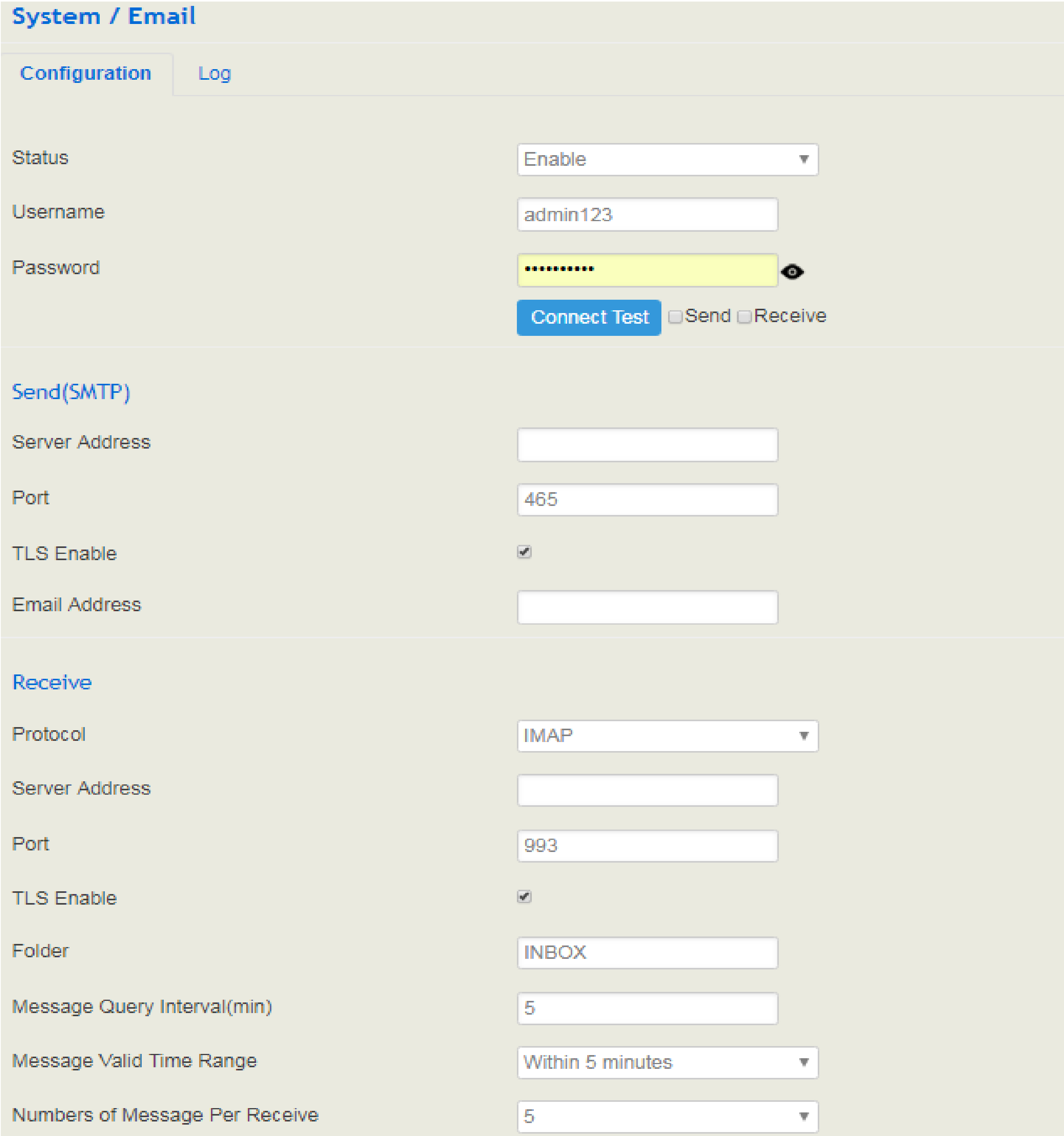

System / Schedule Task
Reboot
Record Backup
CDR Backup
Config Backup
Log Backup
Status
Enable
Interval
1
Day
Execution Time
0
Hour
0
Min
Backup Type
All
CDR Format
Sqlite
Local Backup
Storage Location
Udisk
Backup to Server
URL Info
http://192.168.8.169/cdrback
Compress File
Save
Reset


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


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

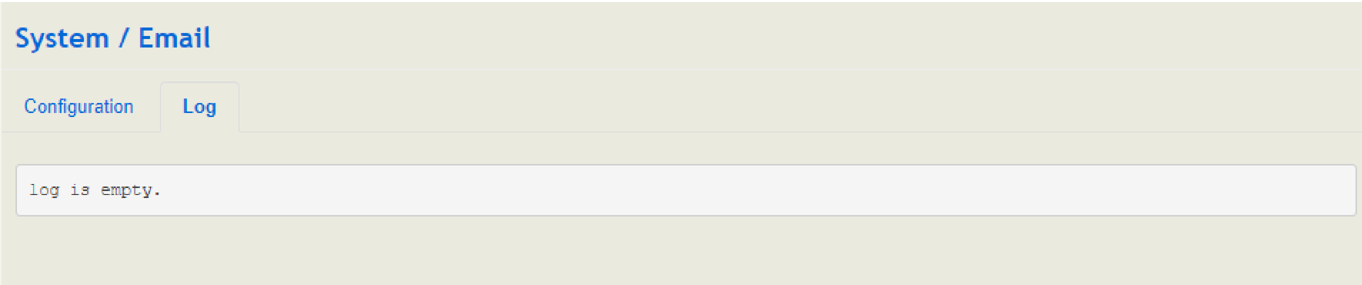
5.3.14 Email

On the **System → 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.



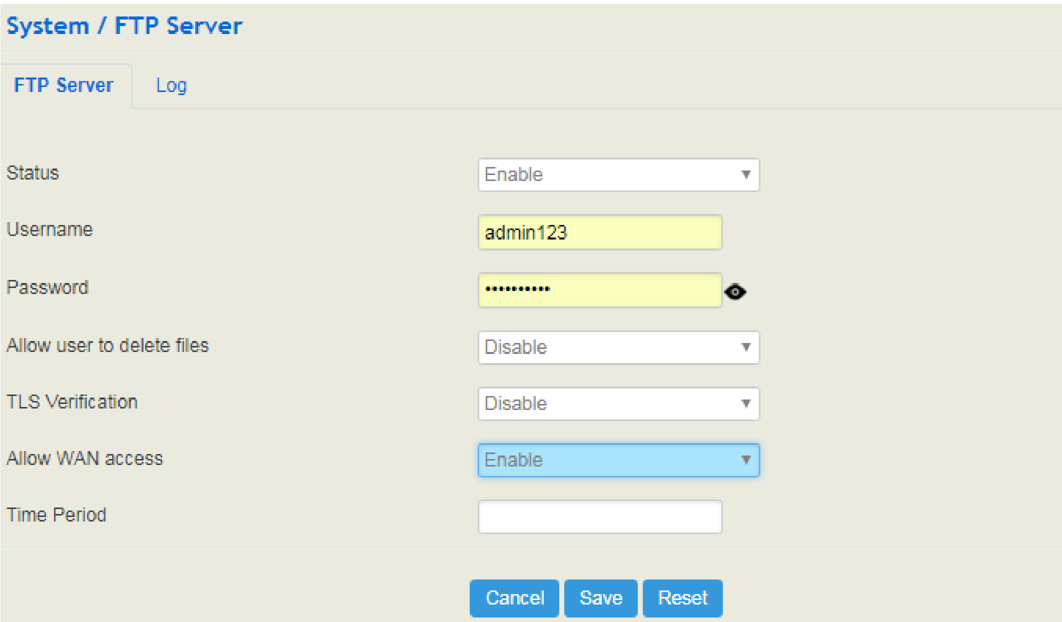
Username	Enter the address of email client
password	The password or authorization code of the email client
Server Address	The Address of the SMTP server, supported by the email client
Protocol	Choose IMAP or POP3. When POPS is selected, TLS port is 993 by default.
Message Query Interval (min)	The time 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 → Email→ Log** interface, you can check Email logs as shown below.

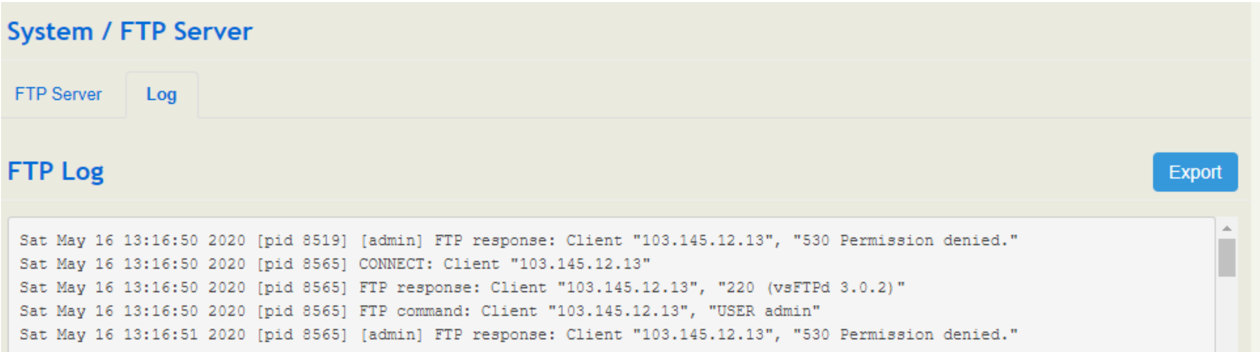


5.3.15 FTP Server

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.

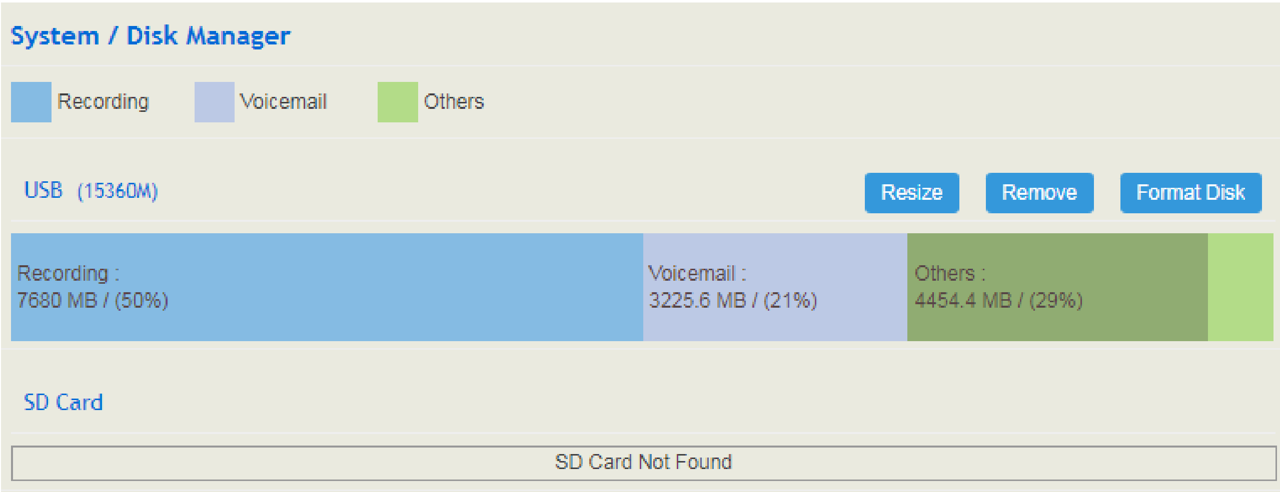


On the **System → FTP→ Log** interface, you can check FTP logs as shown below.



5.3.16 Disk Manager

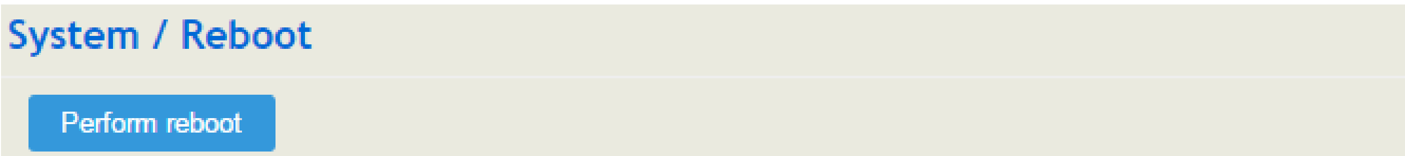
On the **System →Disk Manager** interface, you can see the memory usage of USB and SD card. USB memory are divided into three categories, including voicemail (40%), recording (50%) and Others(10%). You can also divide the proportion of each category, disconnect the USB or execute formatting on this interface.



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

5.3.17 Reboot

On the **System → 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

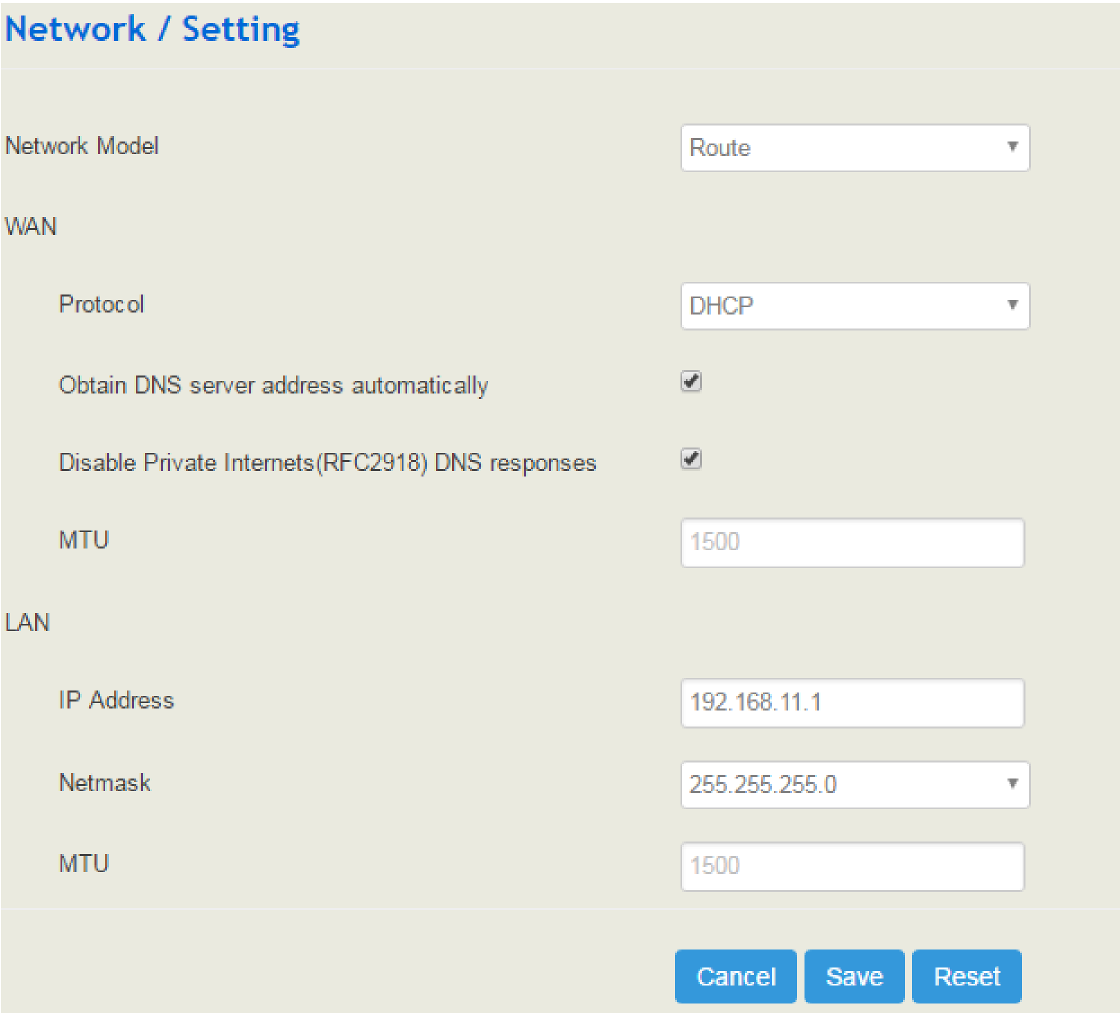
On the **Network → Setting** interface, you can set the IP address of WAN port and LAN port.

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**.

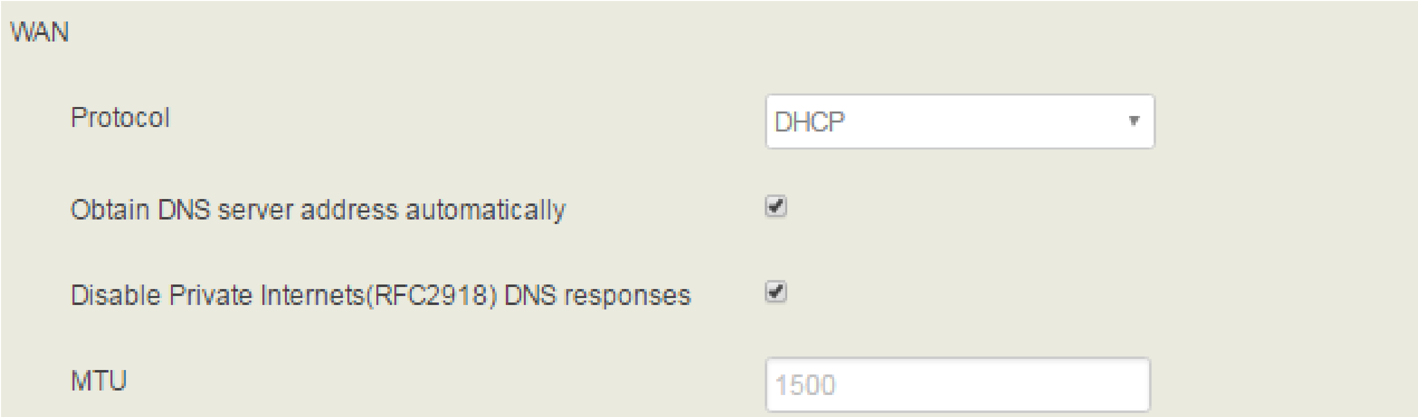
In fact, there are three kinds of IP addresses for selection for WAN port and LAN port, including Static IP address, DHCP and PPPOE.

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.



Set WAN IP as DHCP IP



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

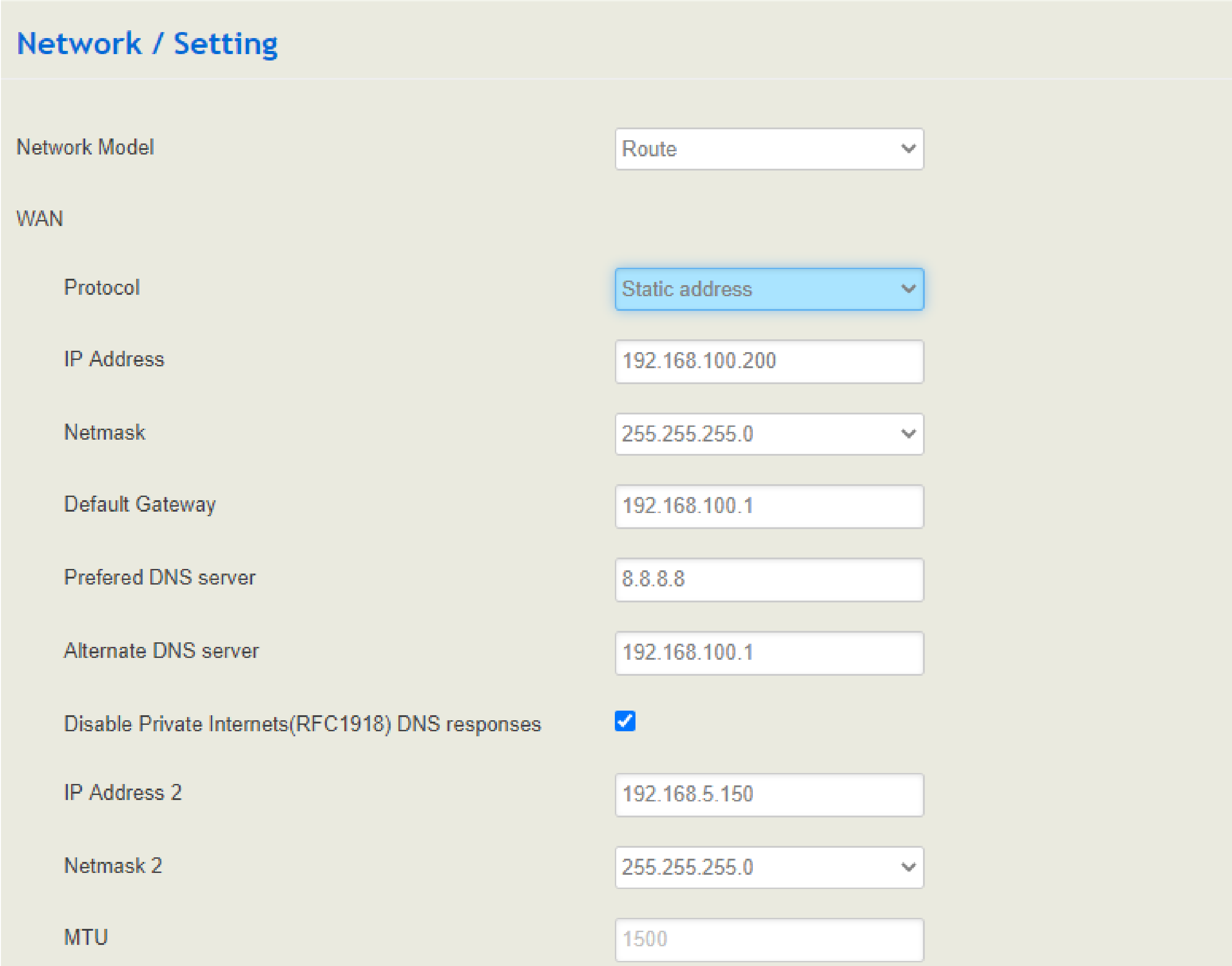
Static IP Address:

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.

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



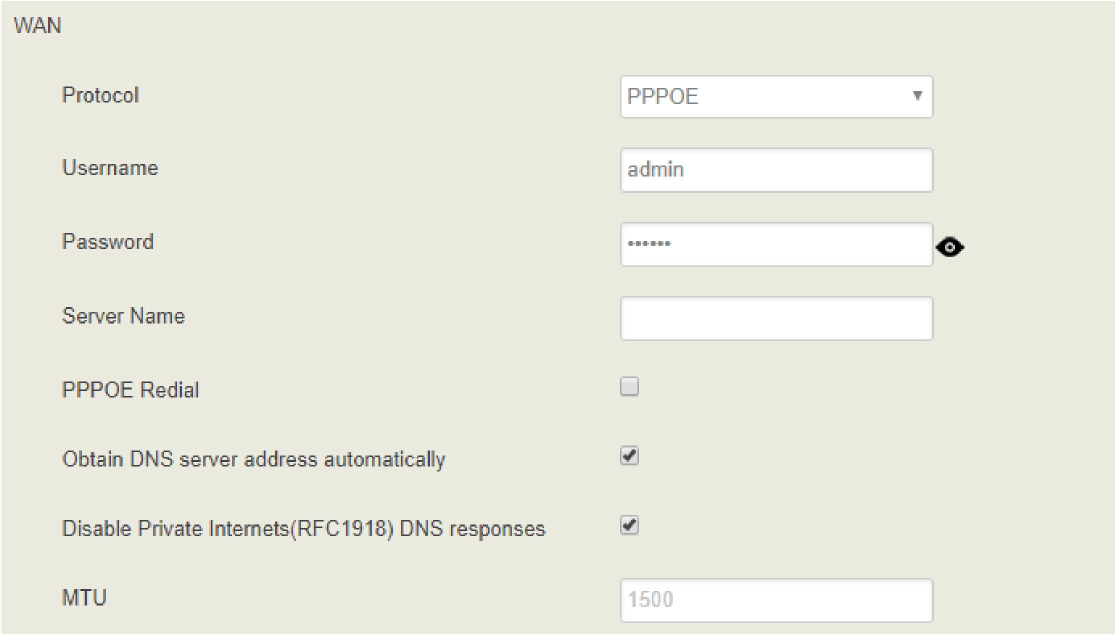
PPPoE:

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.

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

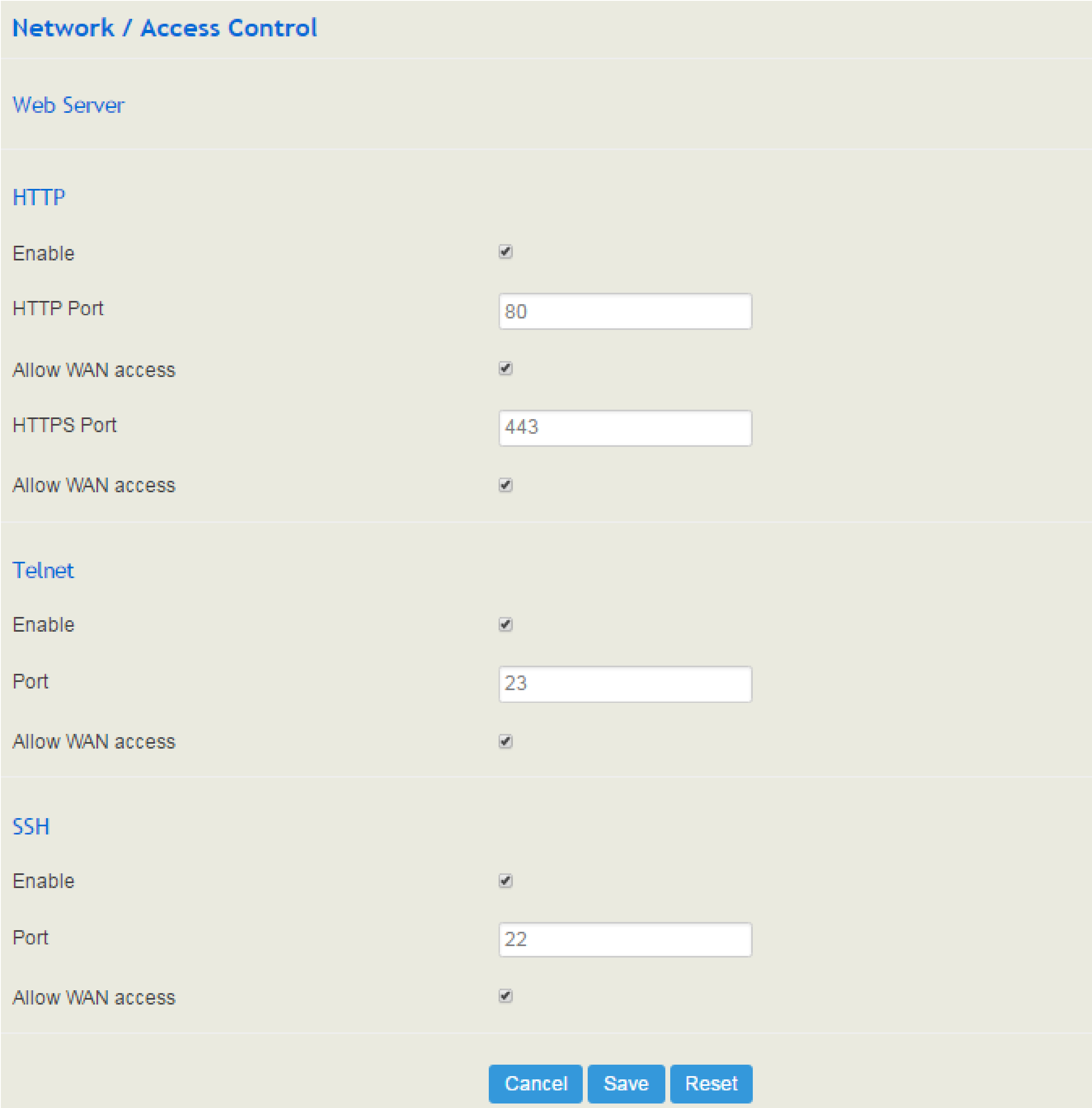
- Username: the account name of PPPoE
- Password: the password of PPPoE
- Server Name: the name of the server where PPPoE is placed

Set WAN IP as PPPoE IP



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 → Access Control** interface.

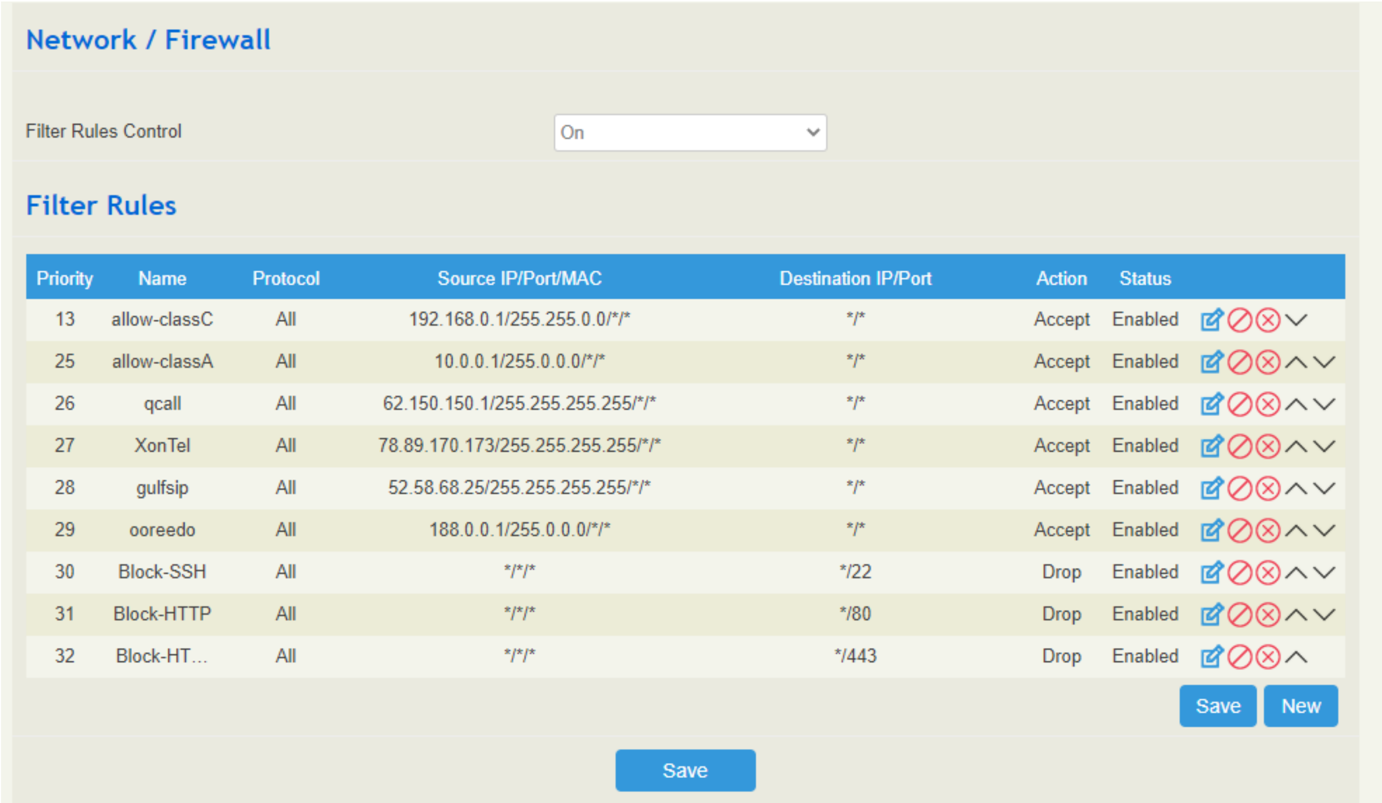


5.4.3 Firewall

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.



Priority	Name	Protocol	Source IP/Port/MAC	Destination IP/Port	Action	Status
13	allow-classC	All	192.168.0.1/255.255.0.0/*/*	*/*	Accept	Enabled
25	allow-classA	All	10.0.0.1/255.0.0.0/*/*	*/*	Accept	Enabled
26	qcall	All	62.150.150.1/255.255.255.255/*/*	*/*	Accept	Enabled
27	XonTel	All	78.89.170.173/255.255.255.255/*/*	*/*	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

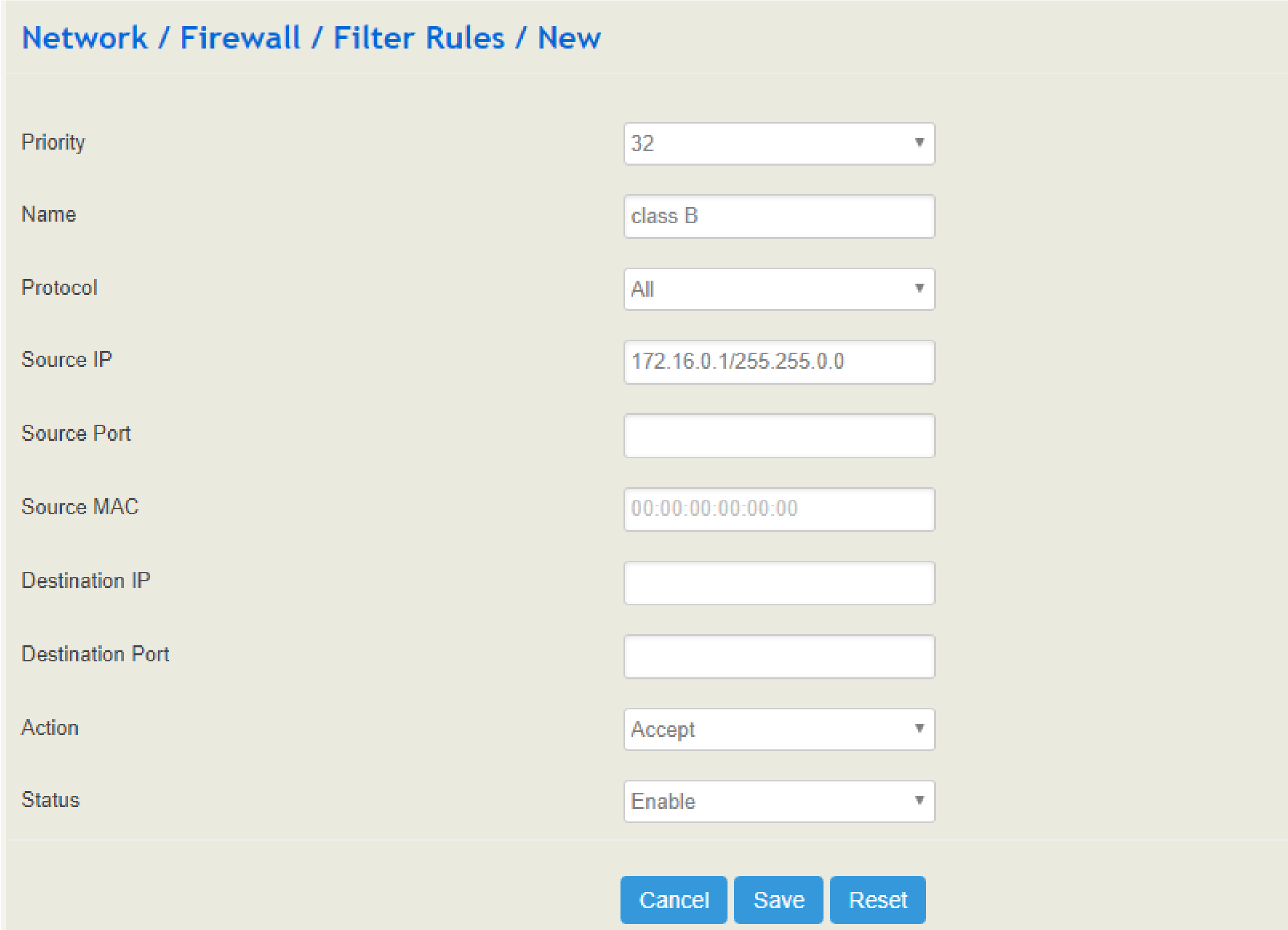
Note:

: Edit information for the corresponding filter rule.

: Delete the corresponding filter rule.

/*: Information of Source or Destination is not completely filled in.

Create Filter Rule

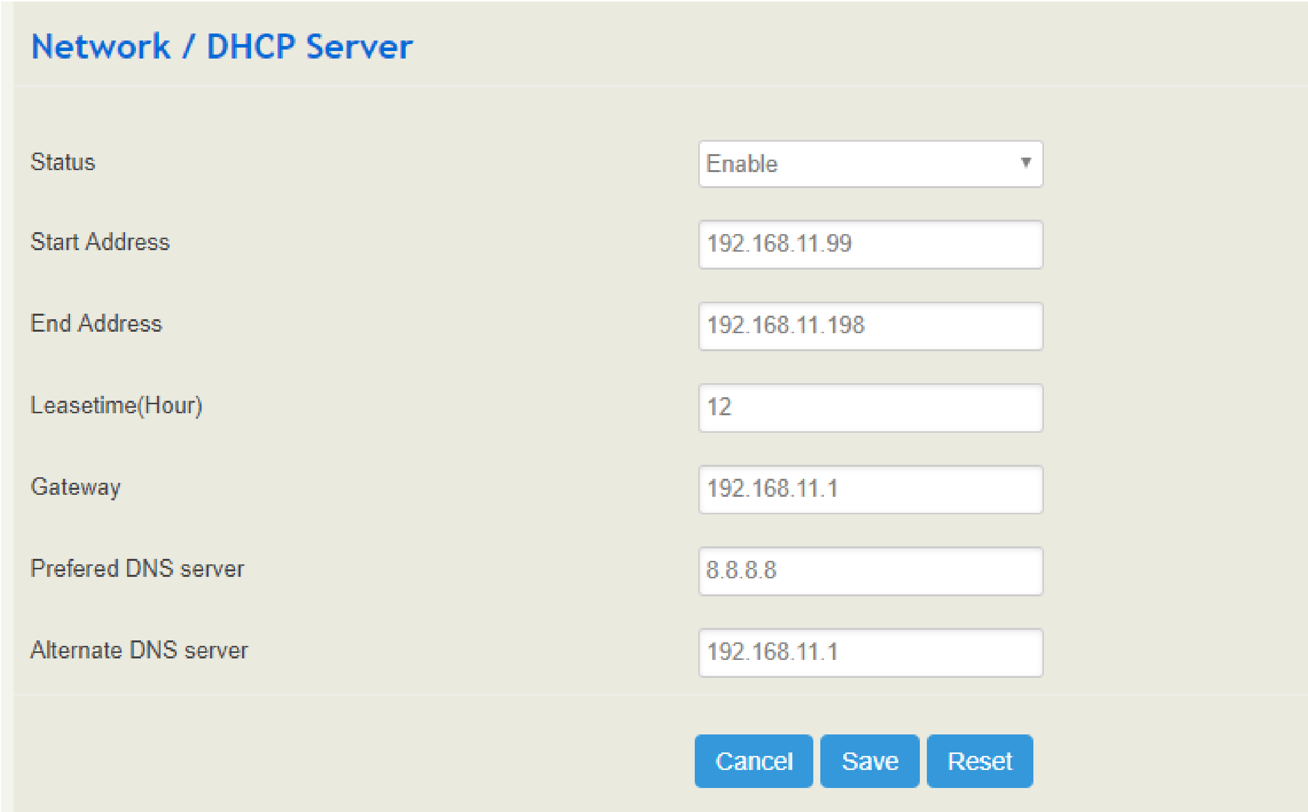


Explanation of Parameters for Filter Rule

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

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.



Explanation of Parameters for DHCP Server

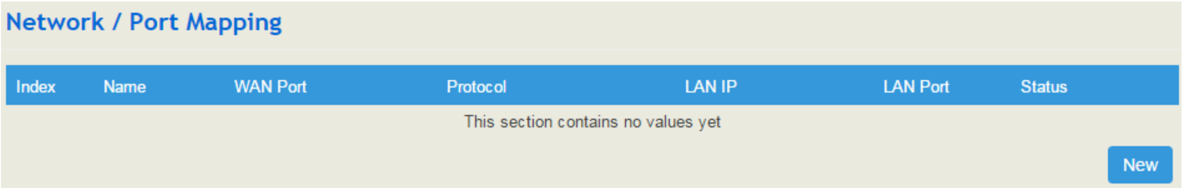
Status	Enable or disable DHCP server option
Start Address	The start IP address of the DHCP pool to be assigned
End Address	The end IP address of the DHCP pool to be assigned
Lease Time(Hour)	The validity period of the assigned IP address in hours
Gateway	The gateway of the DHCP pool to be assigned, it 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

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.

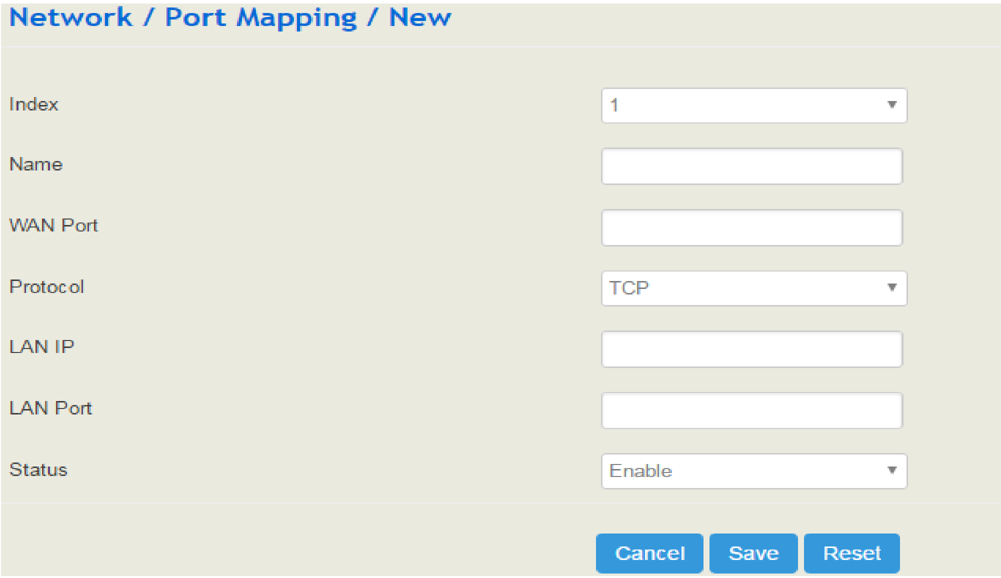
Configuration Procedures:

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



2. Click the **New** button.

3. Fill in information on the following interface.

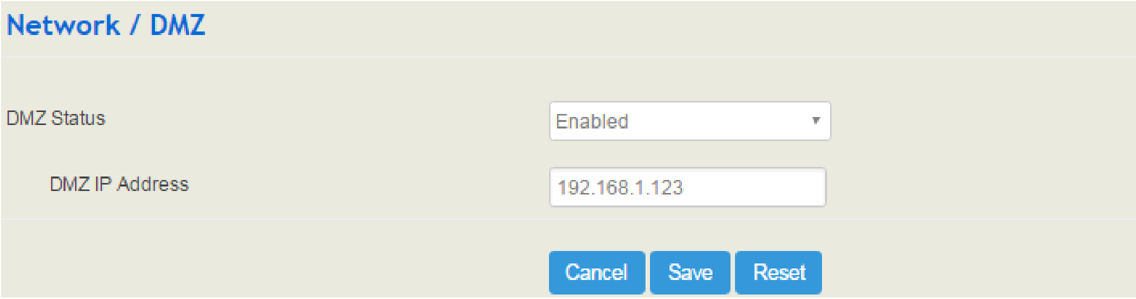


Name	The name of this port mapping
WAN Port	The port of the client in the wide-area network, which is to visit local-area network
Protocol	Choose TCP, UDP or TCP/UDP
LAN IP	The IP address of the to-be-visited client in local-area network
LAN Port	The port of the to-be-visited client in local-area network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

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

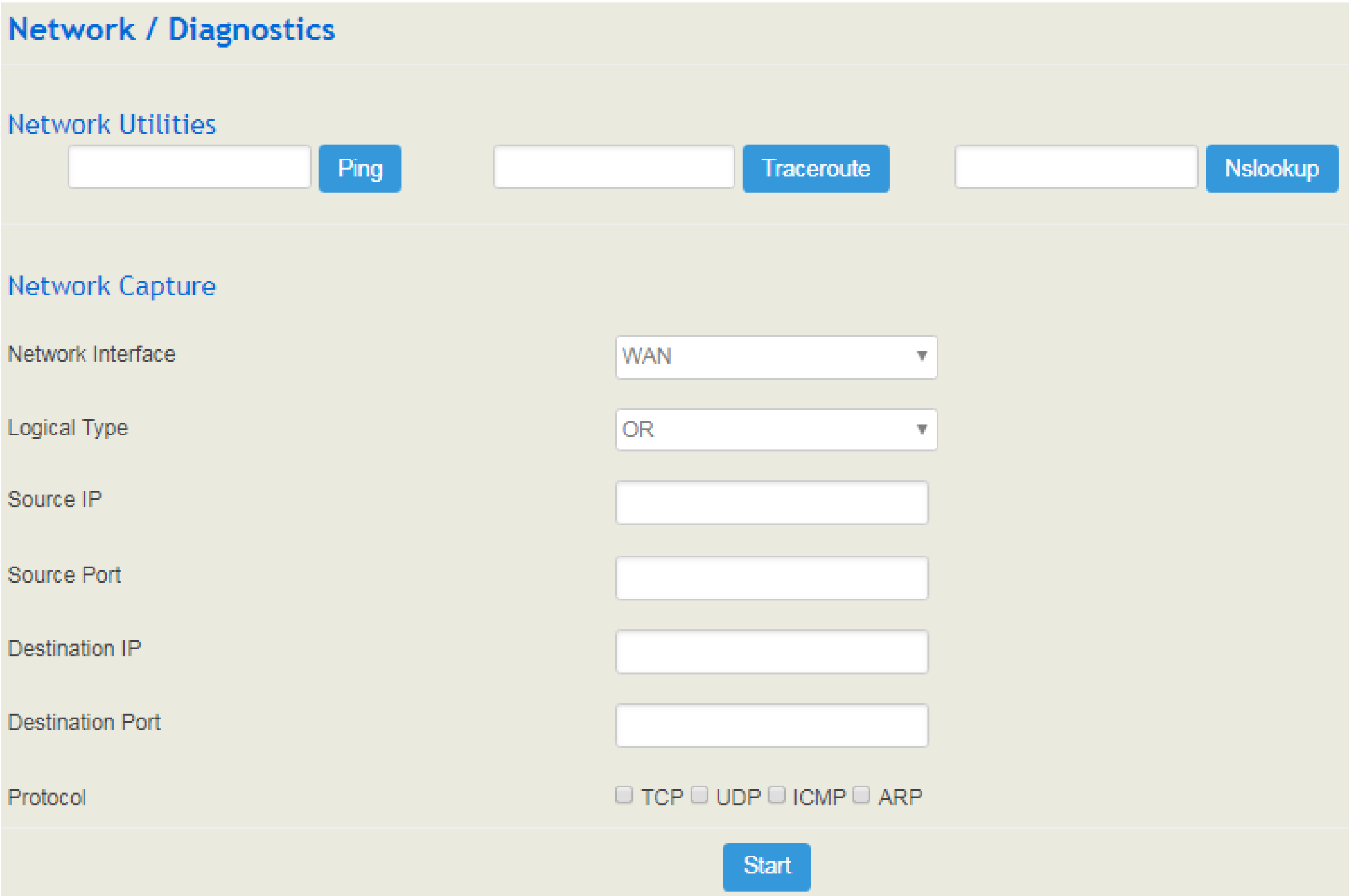
5.4.6 DMZ Setting

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**).



5.4.7 Diagnostics

On the **Network → 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.



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.

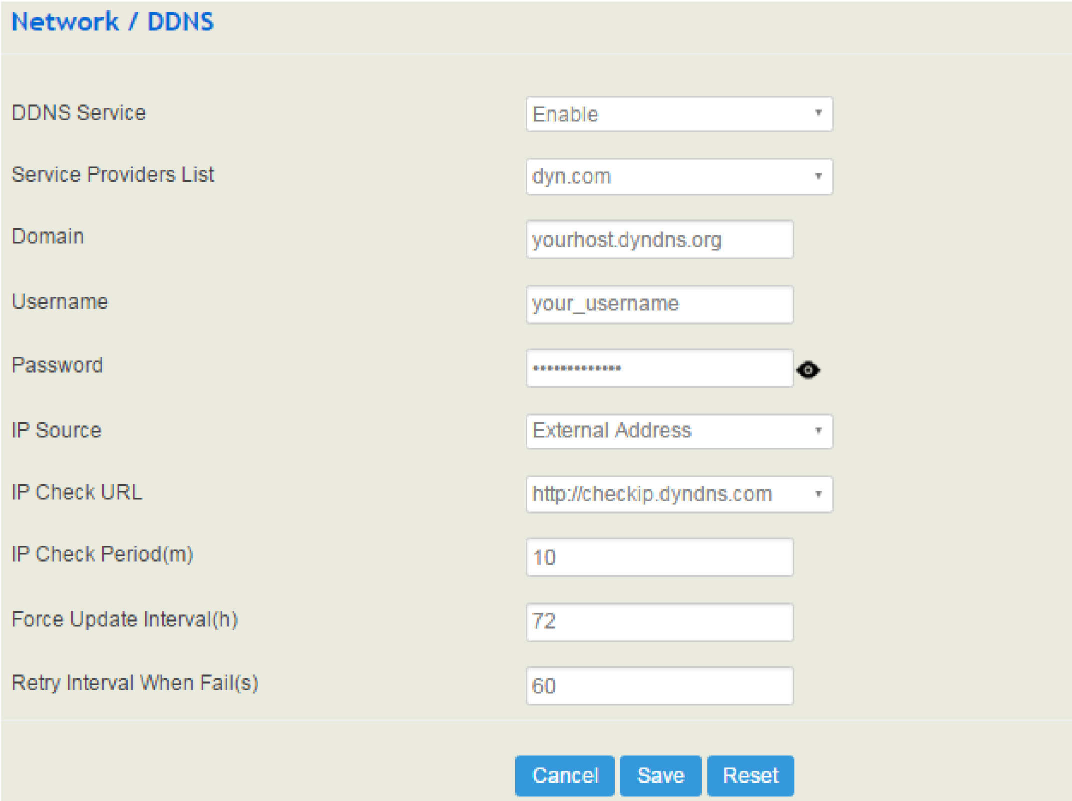
Note: If there are multiple source or destination IP addresses, please use '|' to separate them, for example, 172.16.115.12|172.16.115.15.

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 → 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.



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.

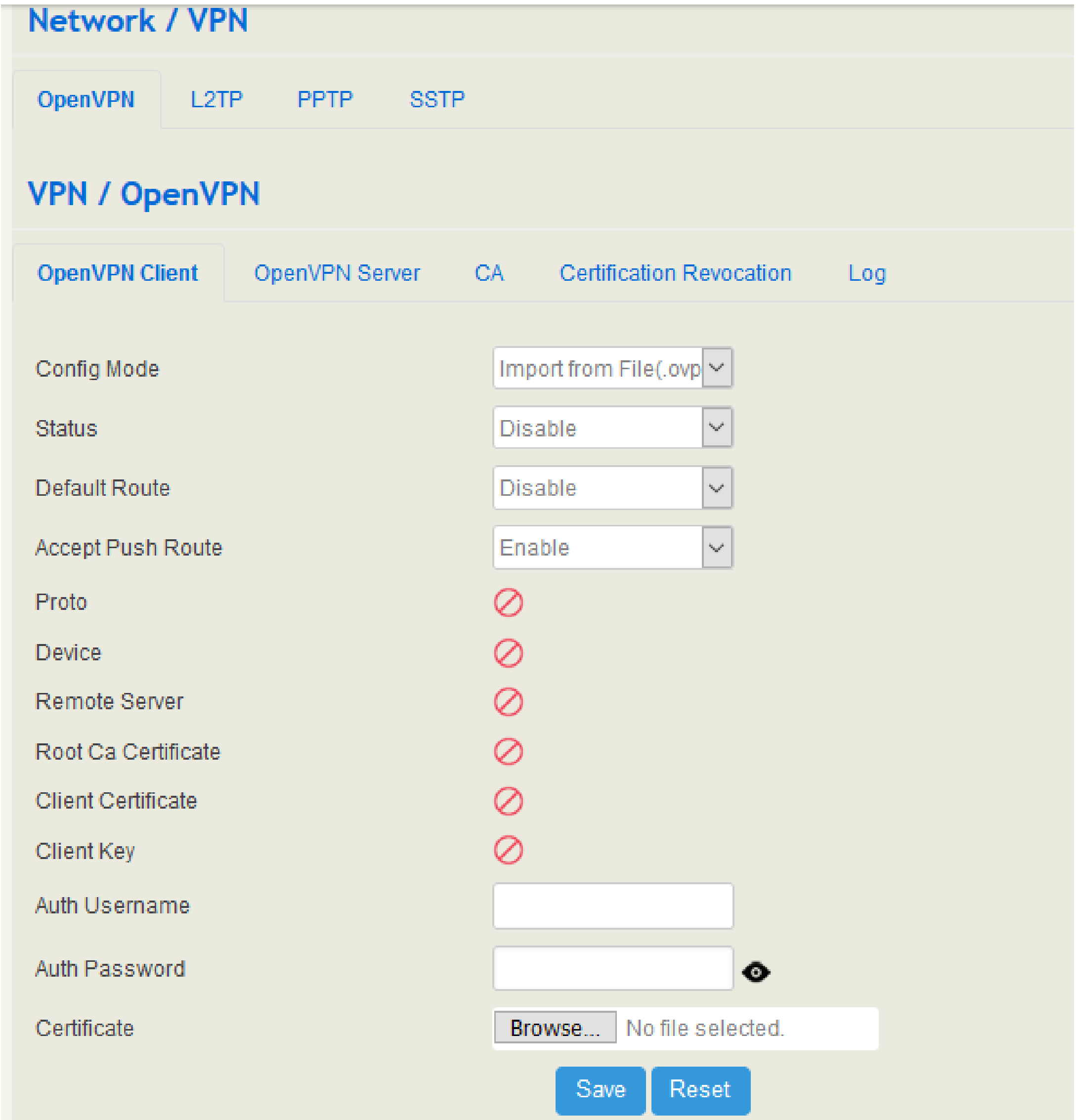
XonTel Plus supports the following VPN protocols:

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

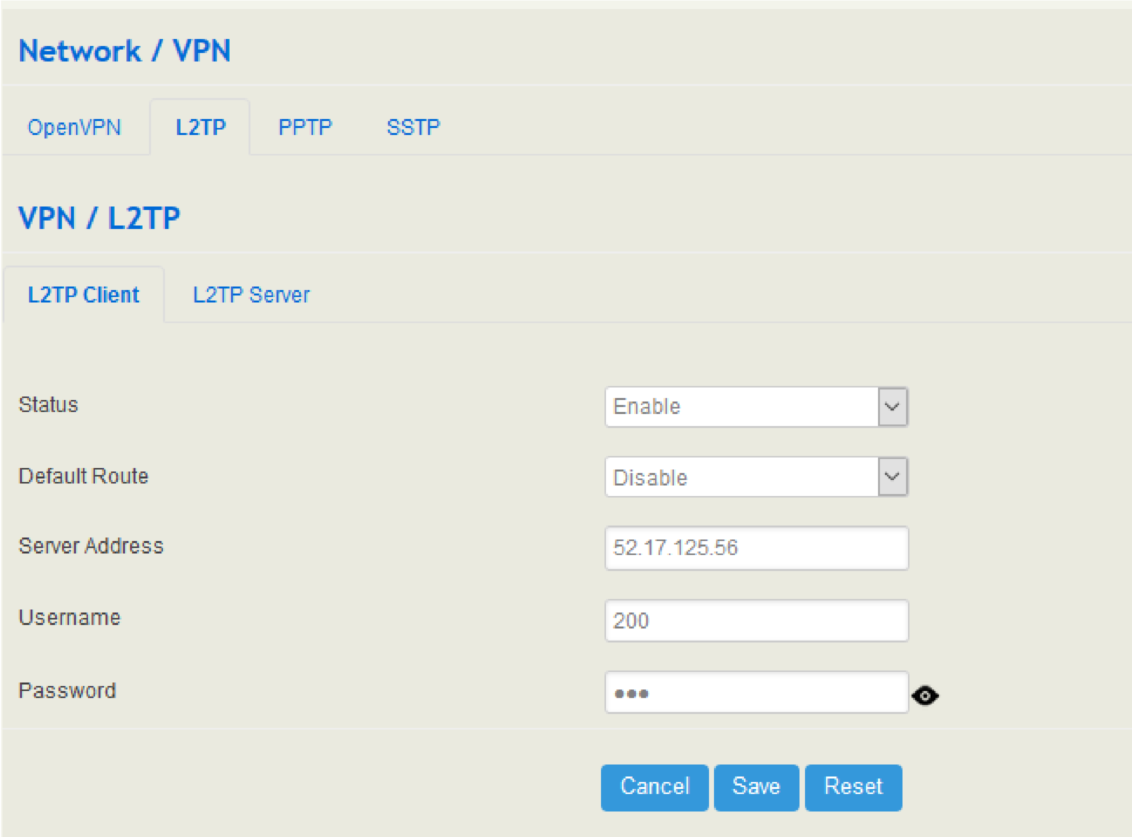
XonTel Can work as a VPN client as shown below:

A. XonTel Plus works as a OpenVPN client



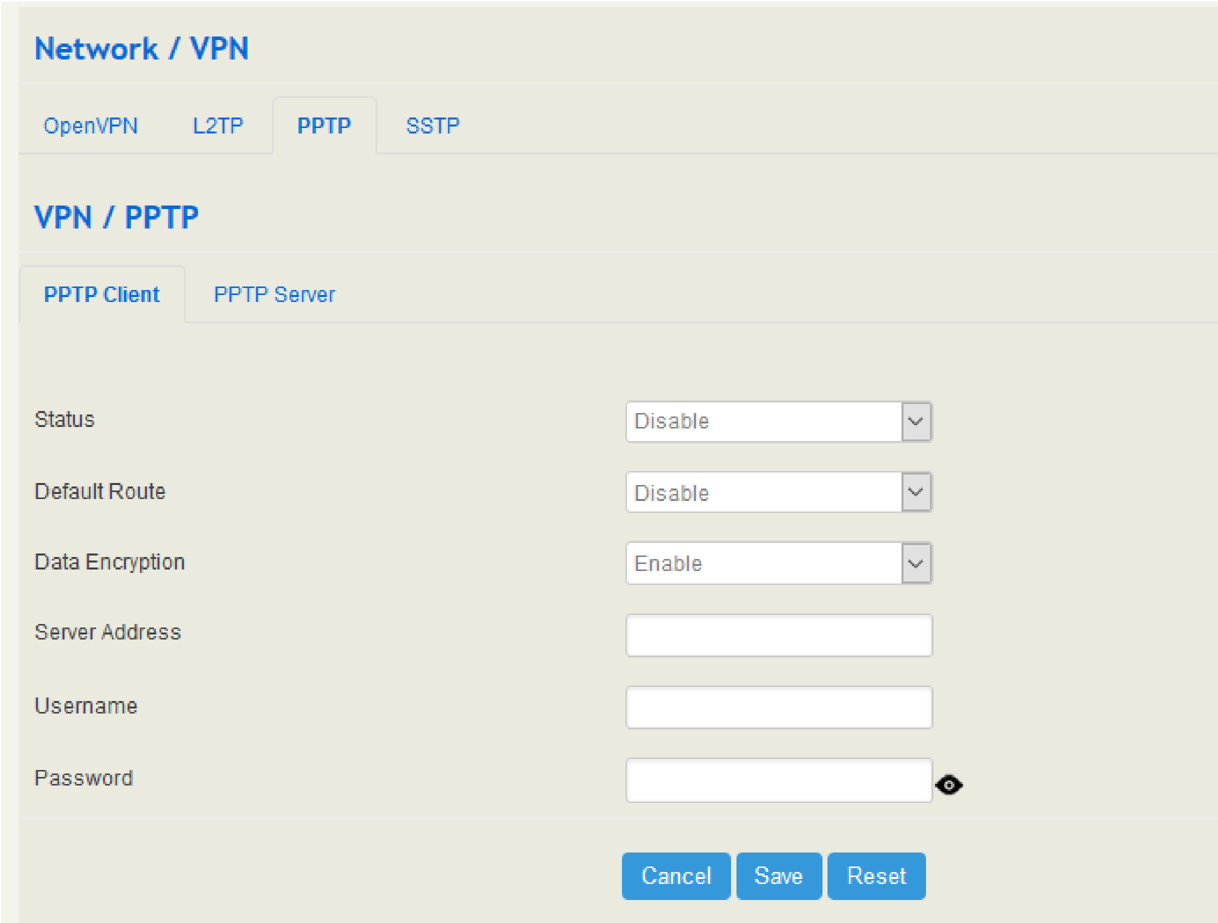
Please note that the certificate that you will upload in PBX format name must be client.ovpn.

B. XonTel Plus Works as L2TP Client



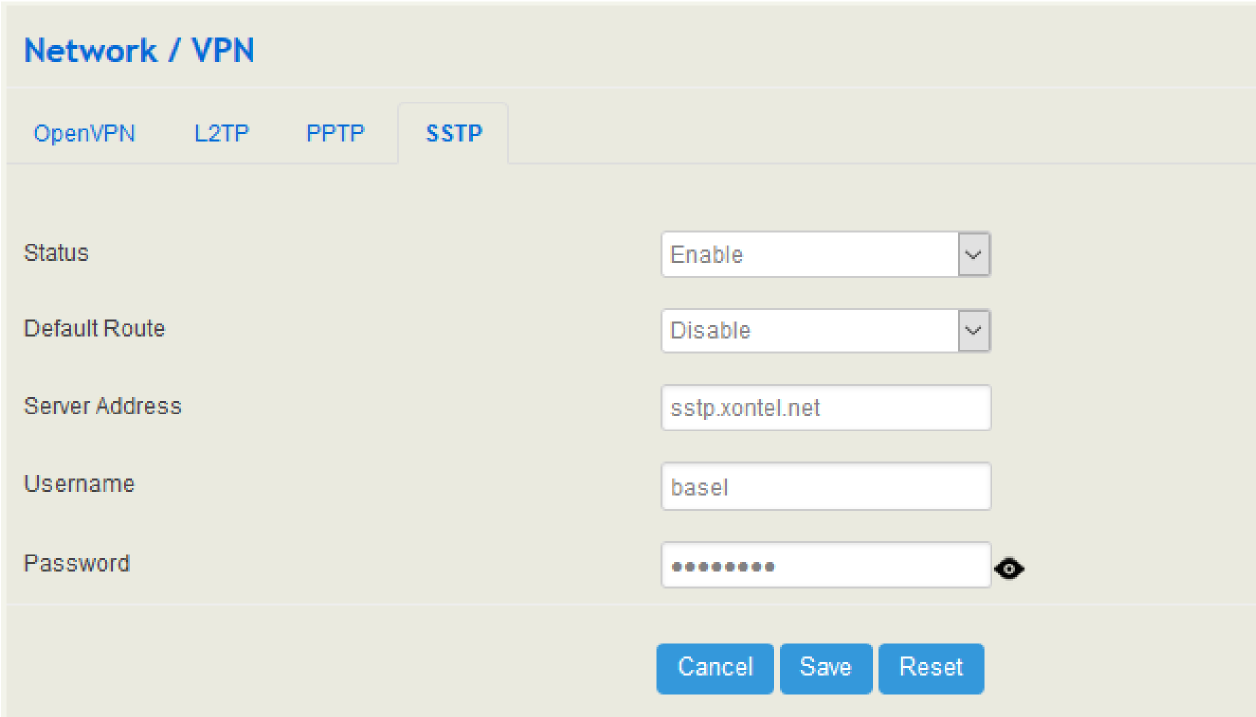
Status	Whether to enable the L2TP client function (XonTel Plus works as L2TP client)
Default Route	Whether to enable default route; If default route is enabled, data are transmitted between L2TP client and L2TP server through VPN route; if it is not enabled, data are transmitted between L2TP client and 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

C. XonTel Plus Works as PPTP Client



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

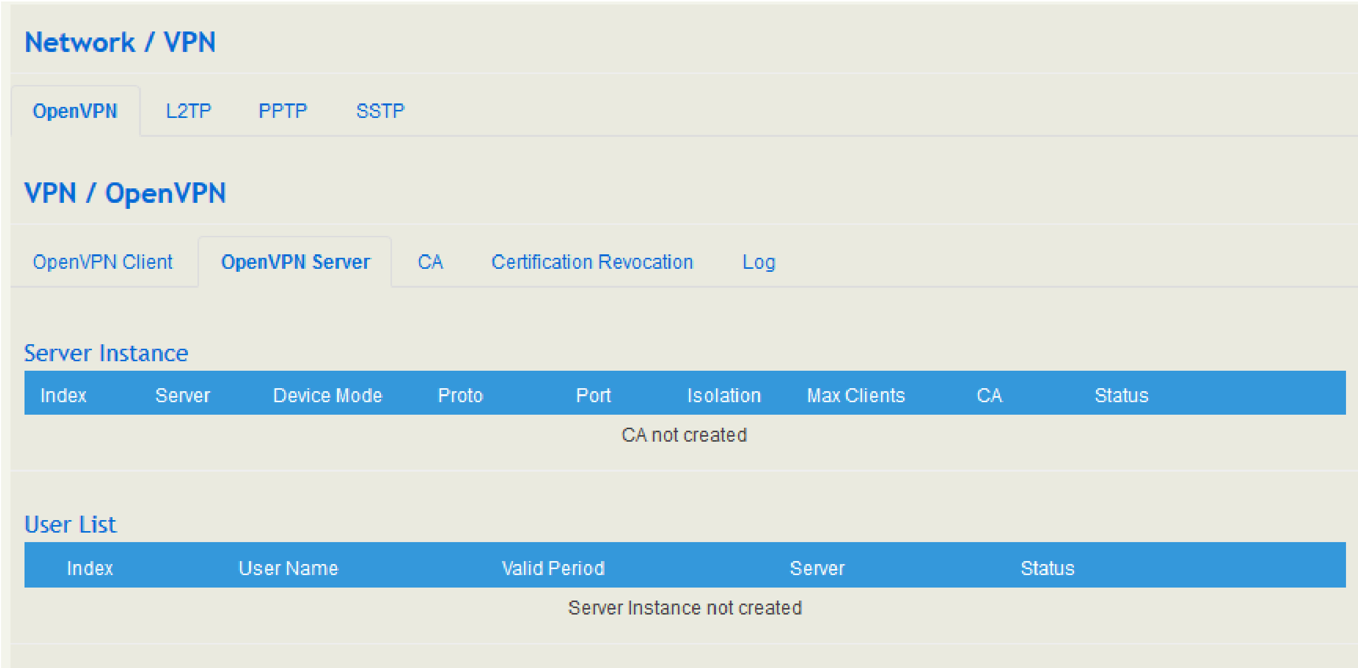
D. XonTel Plus Works as SSTP Client

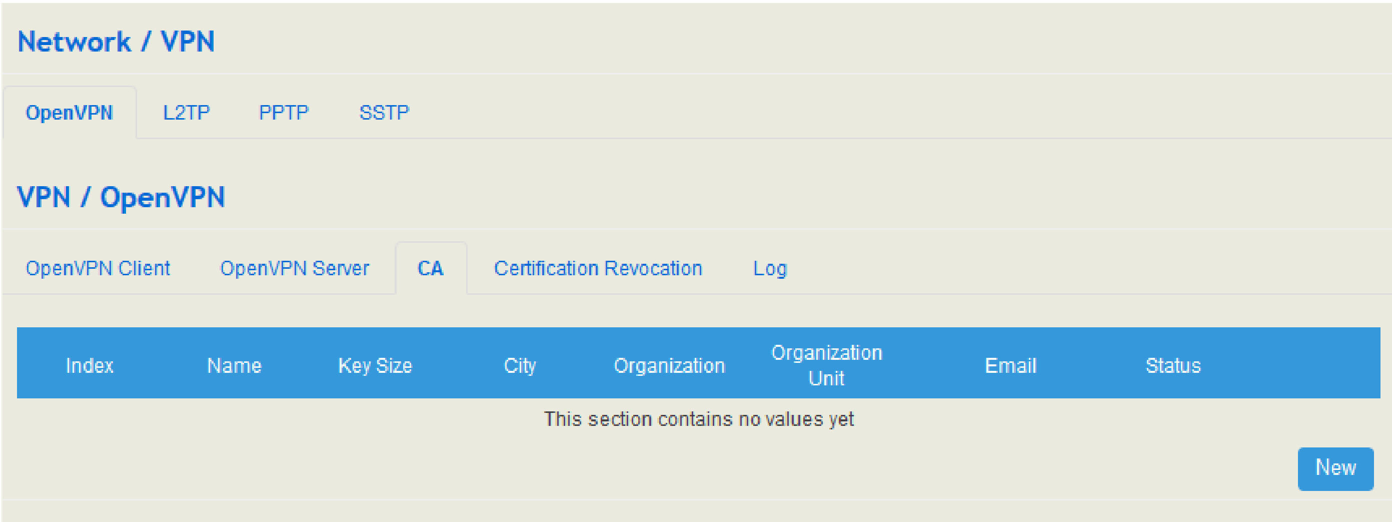


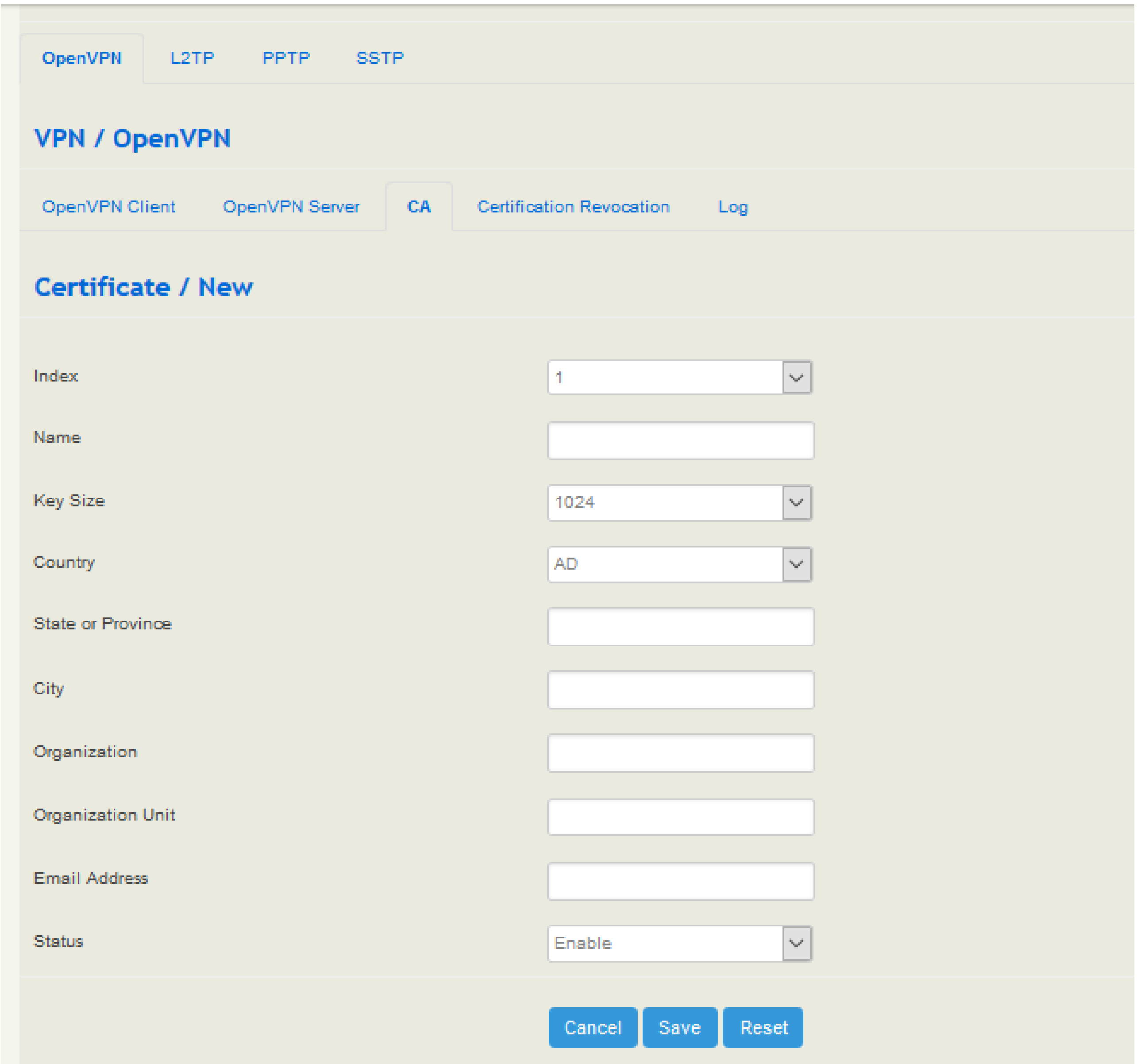
Status	Whether to enable the SSTP client function (XonTel Plus works as SSTP client)
Default Route	Whether to enable default route; If default route is enabled, data are transmitted between SSTP client and SSTP server through VPN route; if it is not enabled, data are transmitted between SSTP client and 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

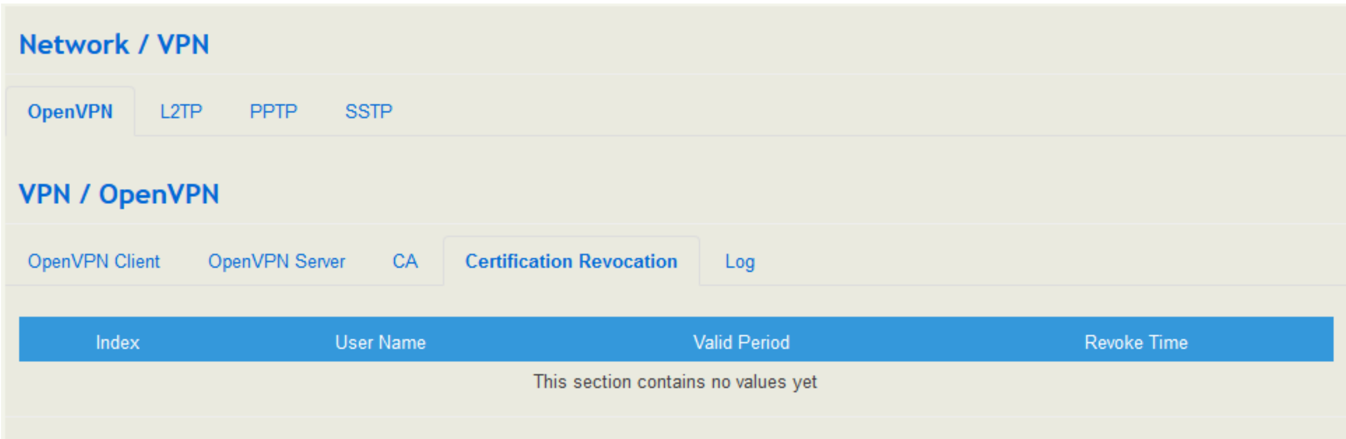
XonTel Plus can work as a VPN Server as shown below:

A. XonTel Plus works as a OpenVPN Server

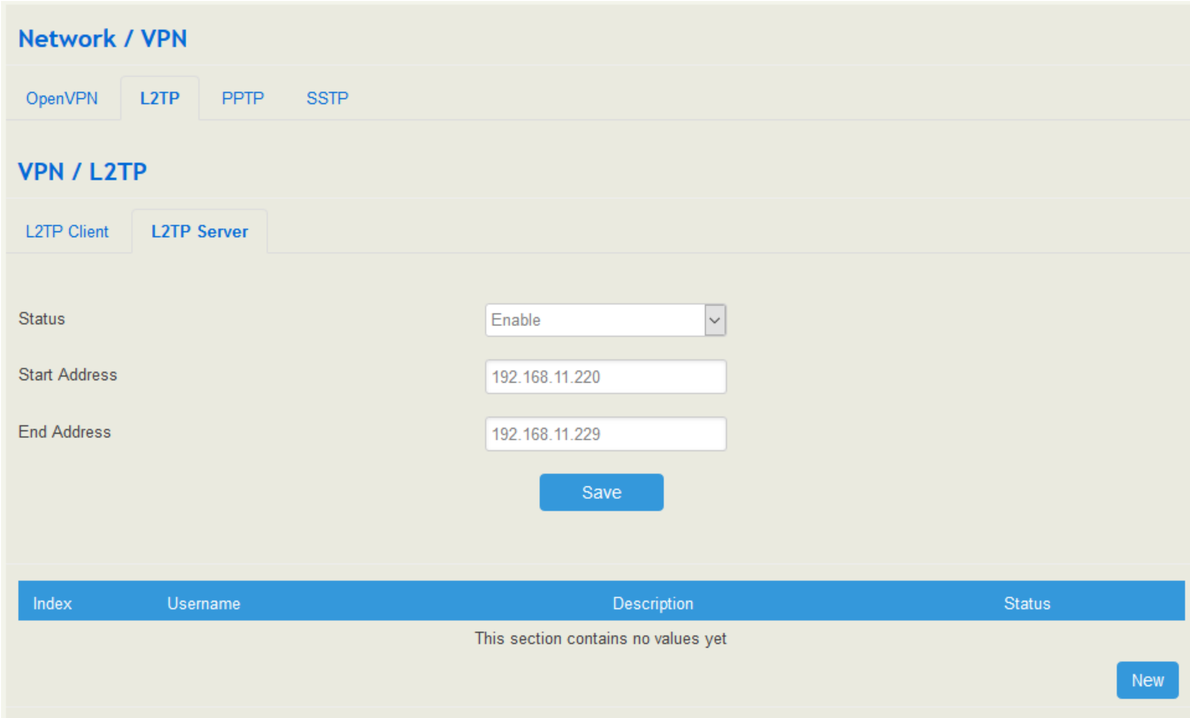


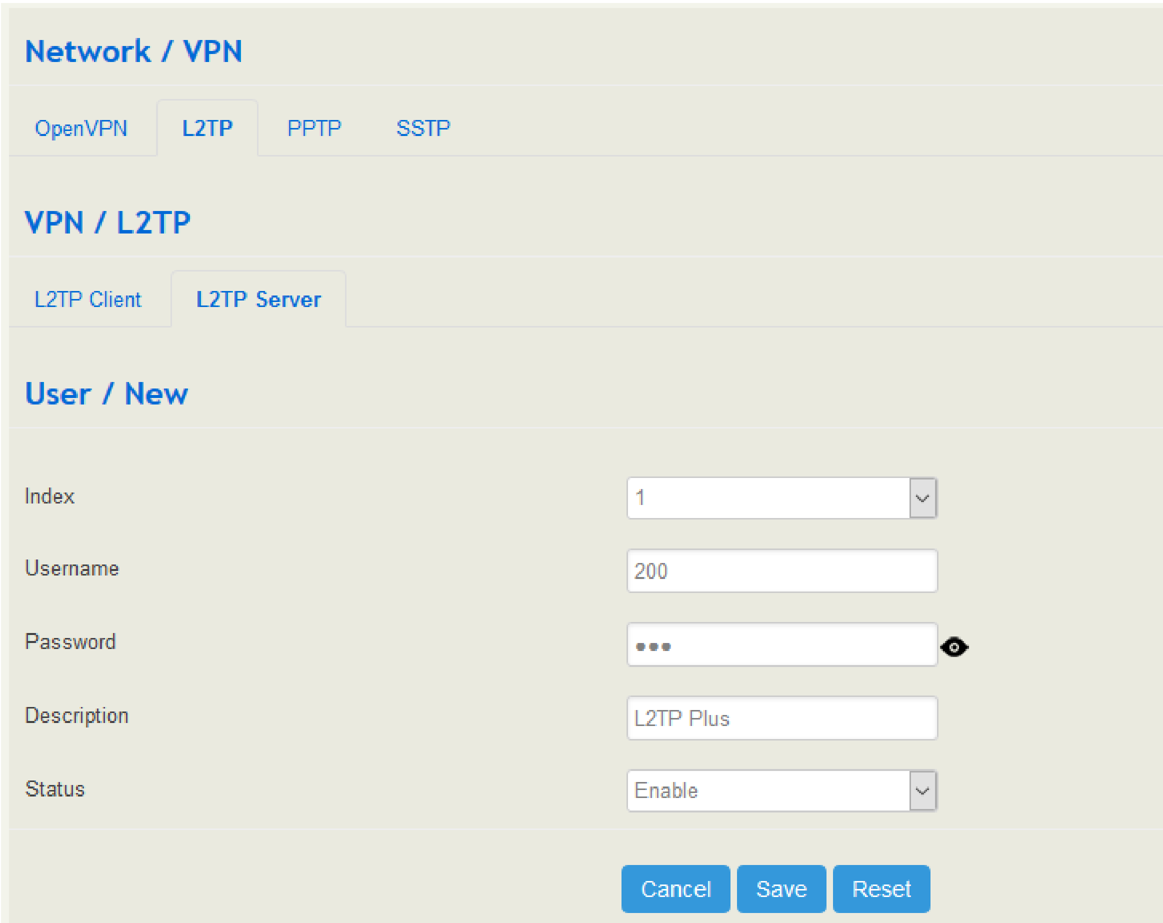



OpenVPN
L2TP
PPTP
SSTP
VPN / OpenVPN
OpenVPN Client
OpenVPN Server
CA
Certification Revocation
Log
Certificate / New
Index
1
Name
Key Size
1024
Country
AD
State or Province
City
Organization
Organization Unit
Email Address
Status
Enable
Cancel
Save
Reset

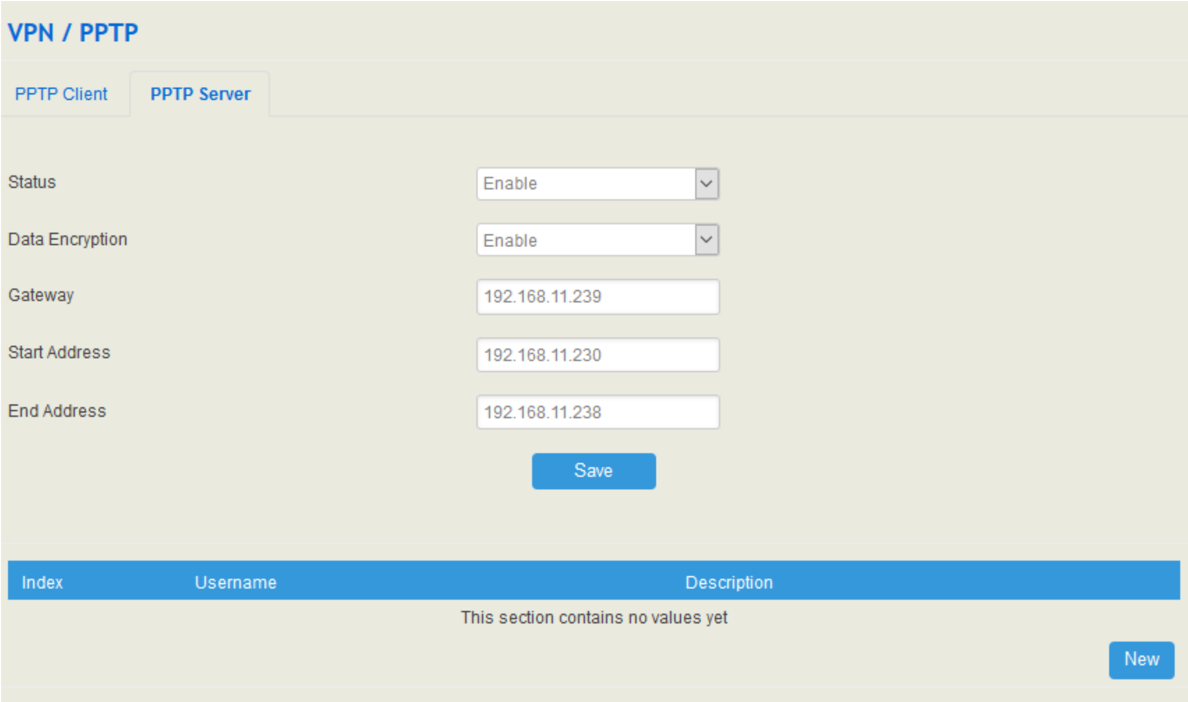

Network / VPN
OpenVPN
L2TP
PPTP
SSTP
VPN / OpenVPN
OpenVPN Client
OpenVPN Server
CA
Certification Revocation
Log
Index
User Name
Valid Period
Revoke Time
This section contains no values yet

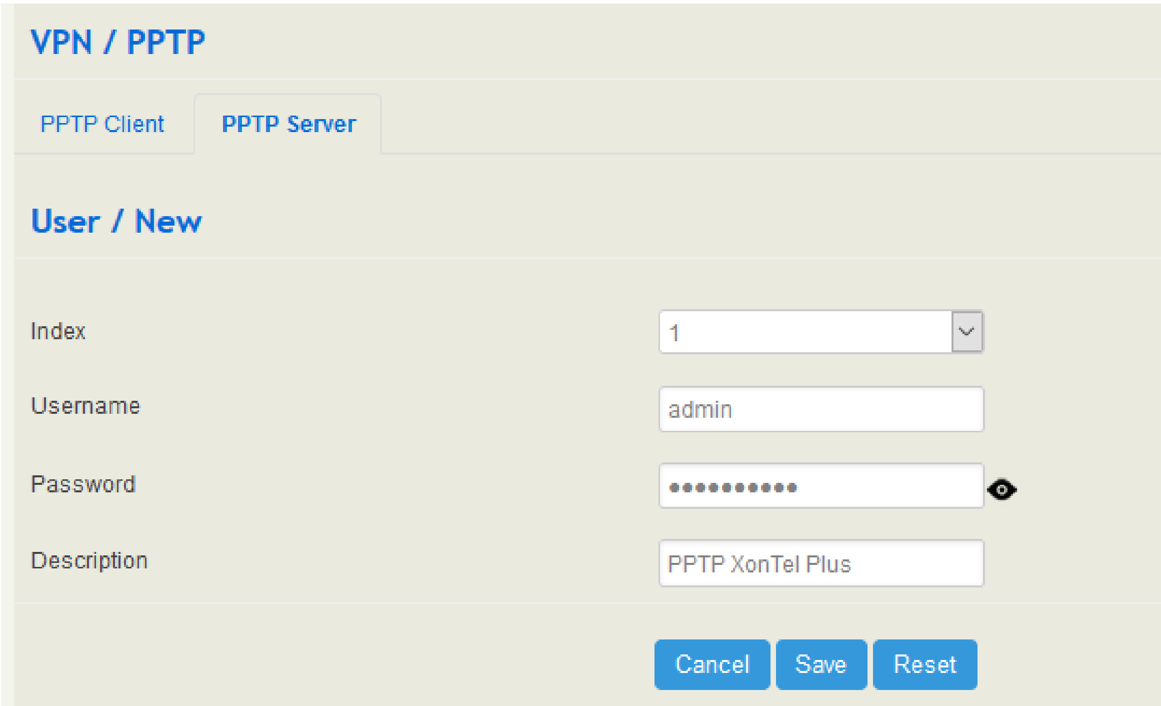
B. XonTel Plus works as a L2TP Server





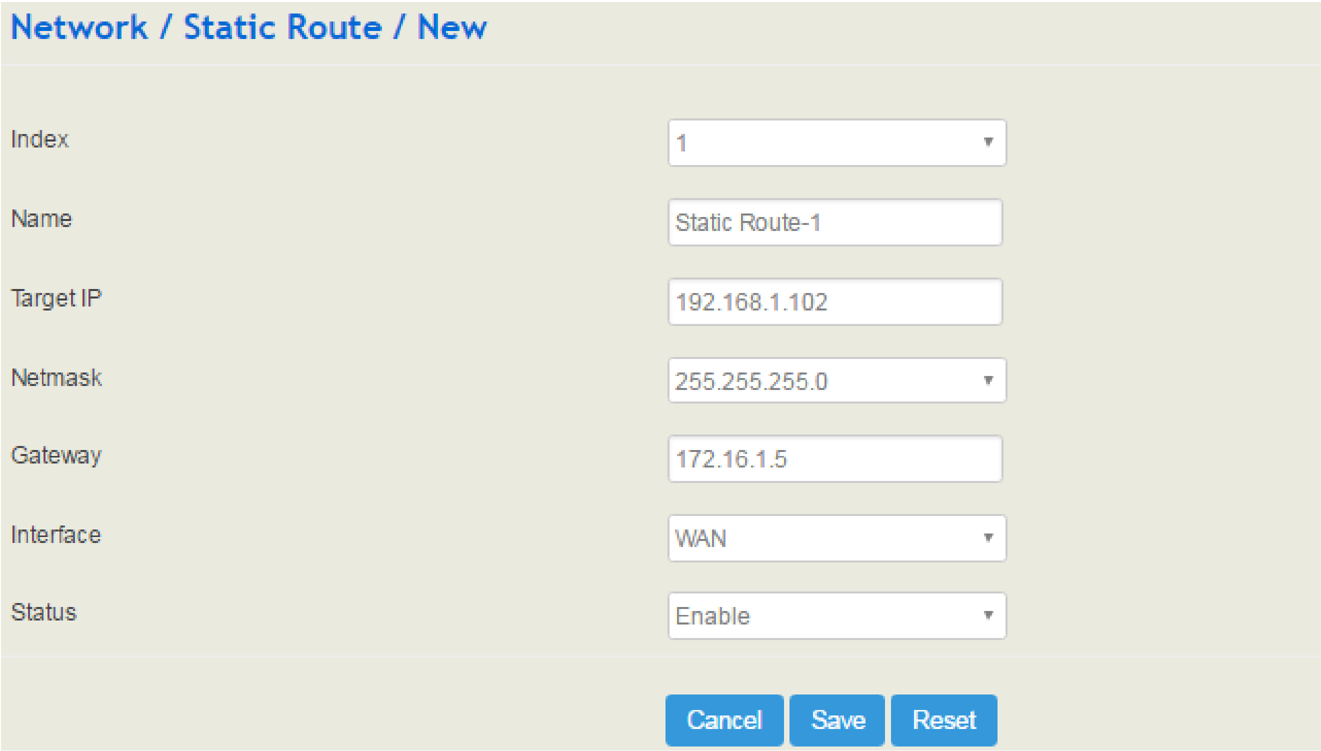
C. XonTel Plus works as a PPTP Server





5.4.10 Static Route

On the **Network → Static Route** interface, you can configure static routes for the network.

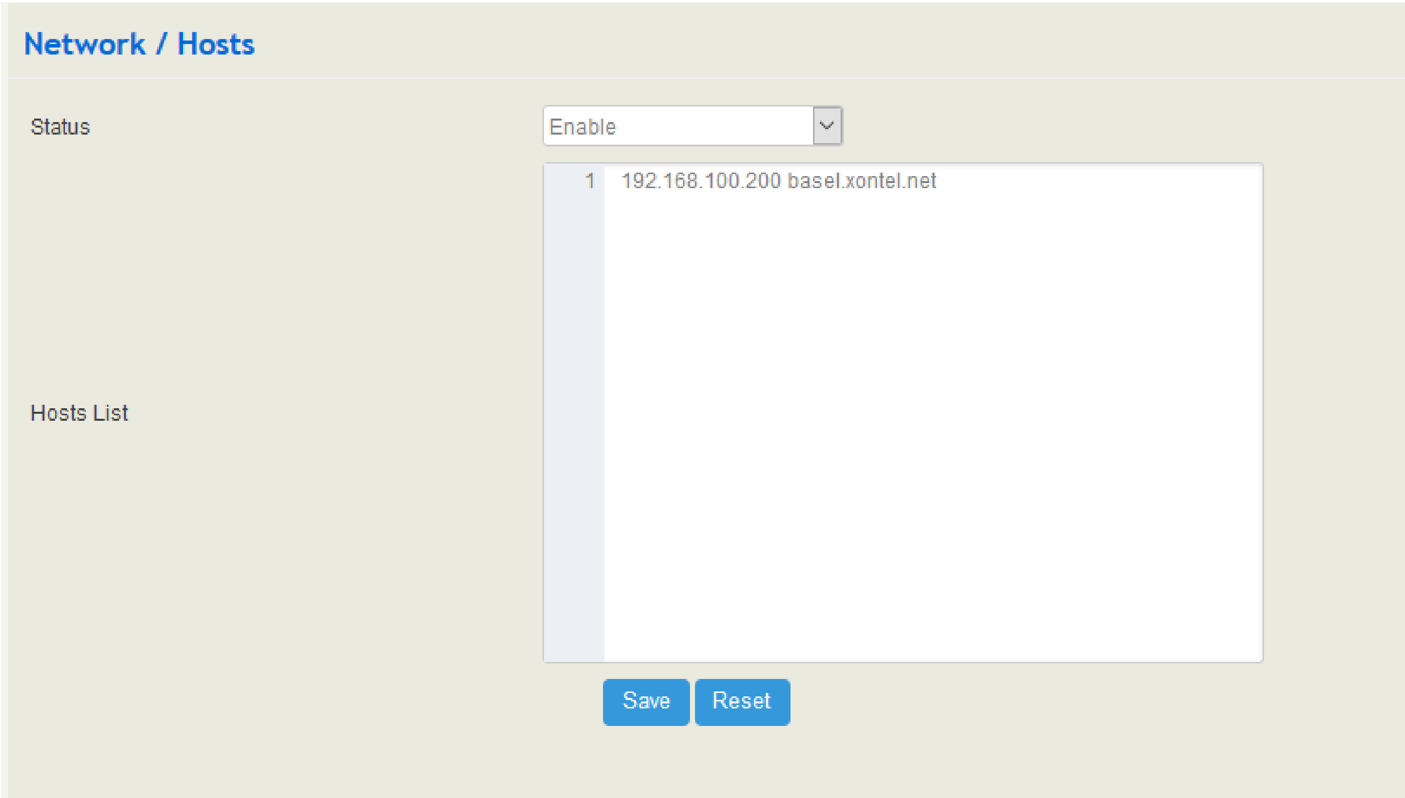


Name	The name of the static route
Target IP	The destination host of the static route
Netmask	The netmask of the static route, default: 255.255.255.0
Gateway	The gateway address of the static route
Interface	The outbound interface of the static route, namely WAN port or LAN port
Status	The static route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

5.4.11 Hosts

On the **Network → 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.

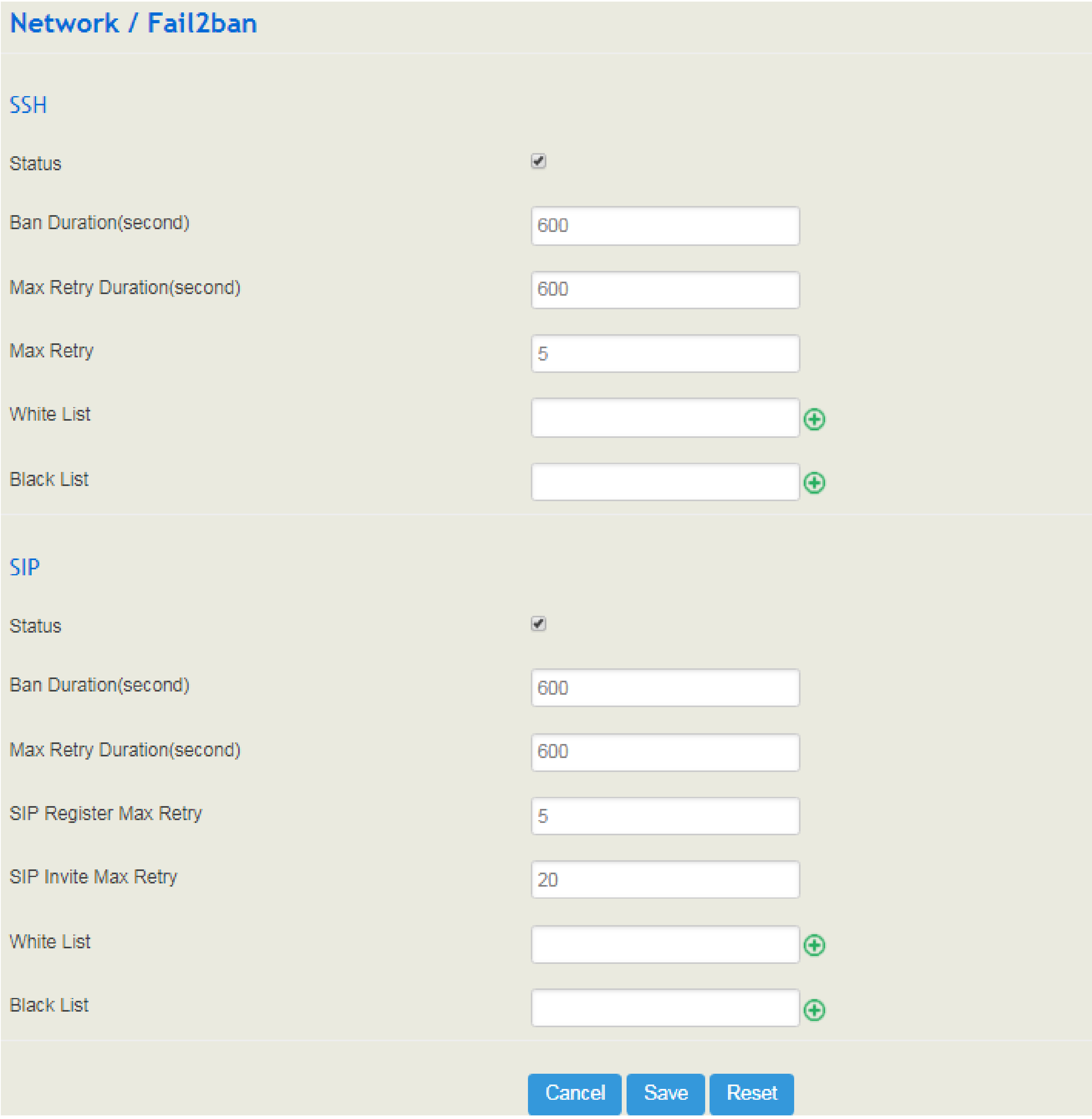
The hosts file contains the mapping relationship between IP address and hostname//domain name. And the mapping relationship allows quick and convenient access to the host.



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.



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 → Fail2ban** interface to check whether the IP address is banned or not as shown below.

Status / Fail2ban

Current Ban List

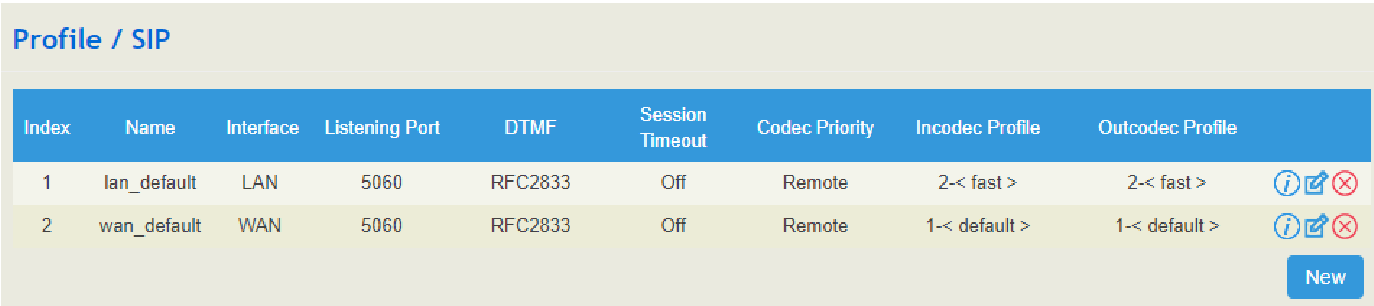
Index	IP	Ban time	Release time	Type	Action
0	45.143.220.95	2020/05/17 16:25:09	2028/08/03 16:25:09	SIP REGIS...	
1	185.53.88.171	2020/05/09 06:38:27	2028/07/26 06:38:27	SIP INVITE	
2	62.173.147.235	2020/05/09 20:42:40	2028/07/26 20:42:40	SIP INVITE	
3	45.143.220.62	2020/04/28 15:14:46	2028/07/15 15:14:46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

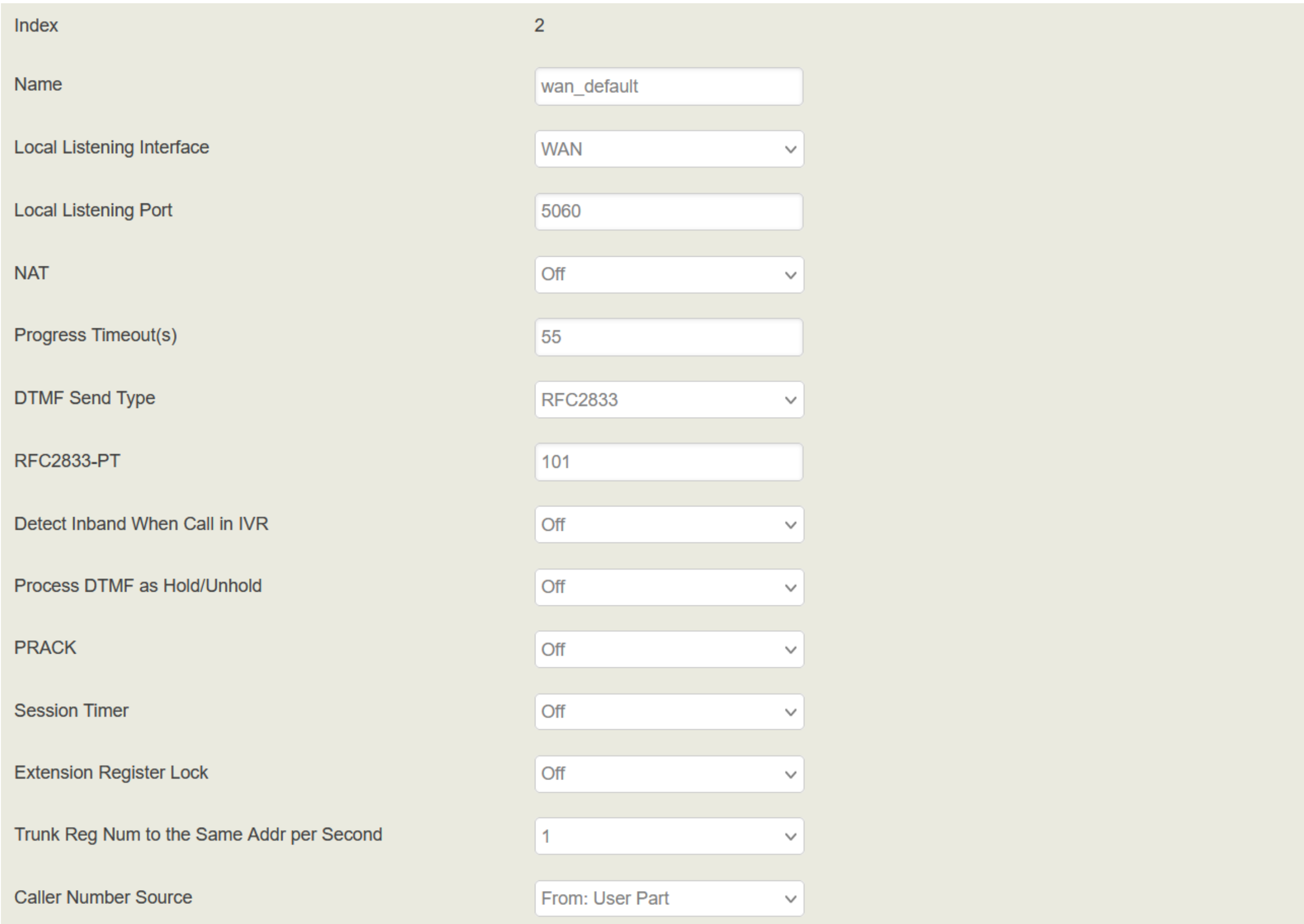
5.5 Profile

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 → 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.





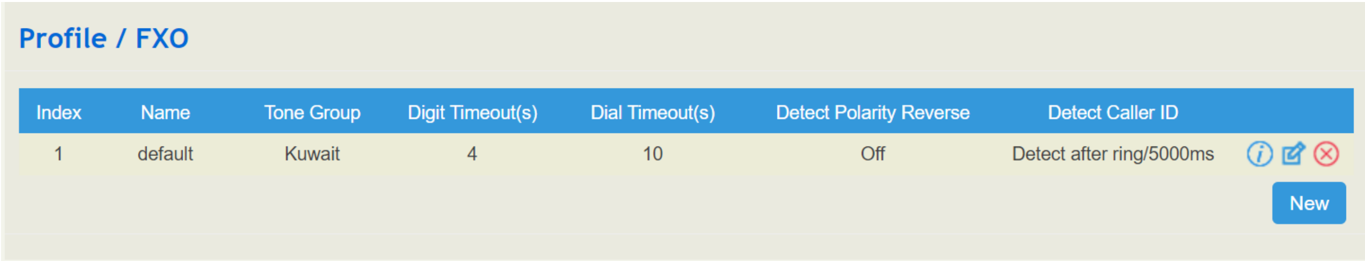
Called Number Source	To: User Part
Inbound Codec Negotiation Priority	Remote
Inbound Codec Profile	1-< default >
Outbound Codec Profile	1-< default >
CNG(Comfort Noise Generator)	On
Bypass Media(SIP to SIP)	Off
Proxy Media(SIP to SIP)	Off
Detect Extension is Online	Off
Ignore ACK	Off
BLF	Off
CID Header	P-Asserted-Identity
PickUp Caller Refresh Method	Off
Allow Unknown Call	Off
Inbound Source Filter	0.0.0.0/0
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
Local Listening Interface	The local listening interface of this SIP profile. It can be WAN port, LAN port, Open VPN, L2TP, PPTP and SSTP. 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
Caller Number Source	**From: User Part**: to obtain the caller number from the user part contained in the ‘From’ field. **From: Display Name**: to obtain the caller number from the display name contained in the ‘From’ field. **To: User Part**: to obtain the caller number from the user part contained in the ‘To’ field. **Contact: User Part**: to obtain the caller number from the user part contained in the ‘Contact’ field.
Called Number Source	**From: User Part**: to obtain the called number from the user part contained in the ‘From’ field. **From: Display Name**: to obtain the called number from the display name contained in the ‘From’ field. **To: User Part**: to obtain the called number from the user part contained in the ‘To’ field. **Contact: User Part**: to obtain the called number from the user part contained in the ‘Contact’ field.

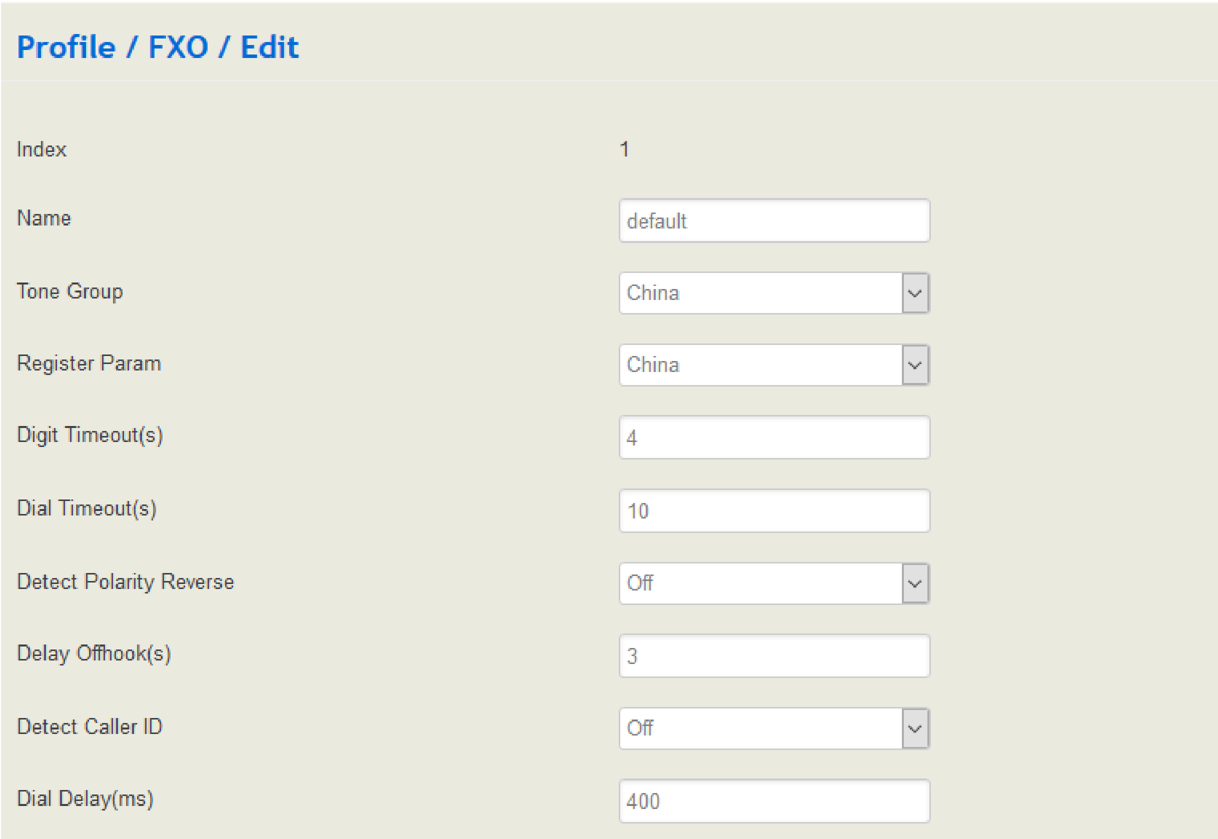
Inbound Codec Negotiation Priority	To take the remote device or the local device as priority for inbound codec negotiation Assume local device supports PCMA, PCMU, G.729 and G.723, while the remote device supports G.723 and G.729 If remote device is taken as codec negotiation priority, G.723 will be 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
CID Header	• Off: Disable CID header. • Remote-Party-ID: whether to send Remote-Party-ID or not. • P-Asserted-Identify: whether to send P-Asserted-Identify or not.
Pickup Caller Refresh Method	The extension has two ways to get the real CID when make call pickup. • Off: Disable Pickup Caller Refresh 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

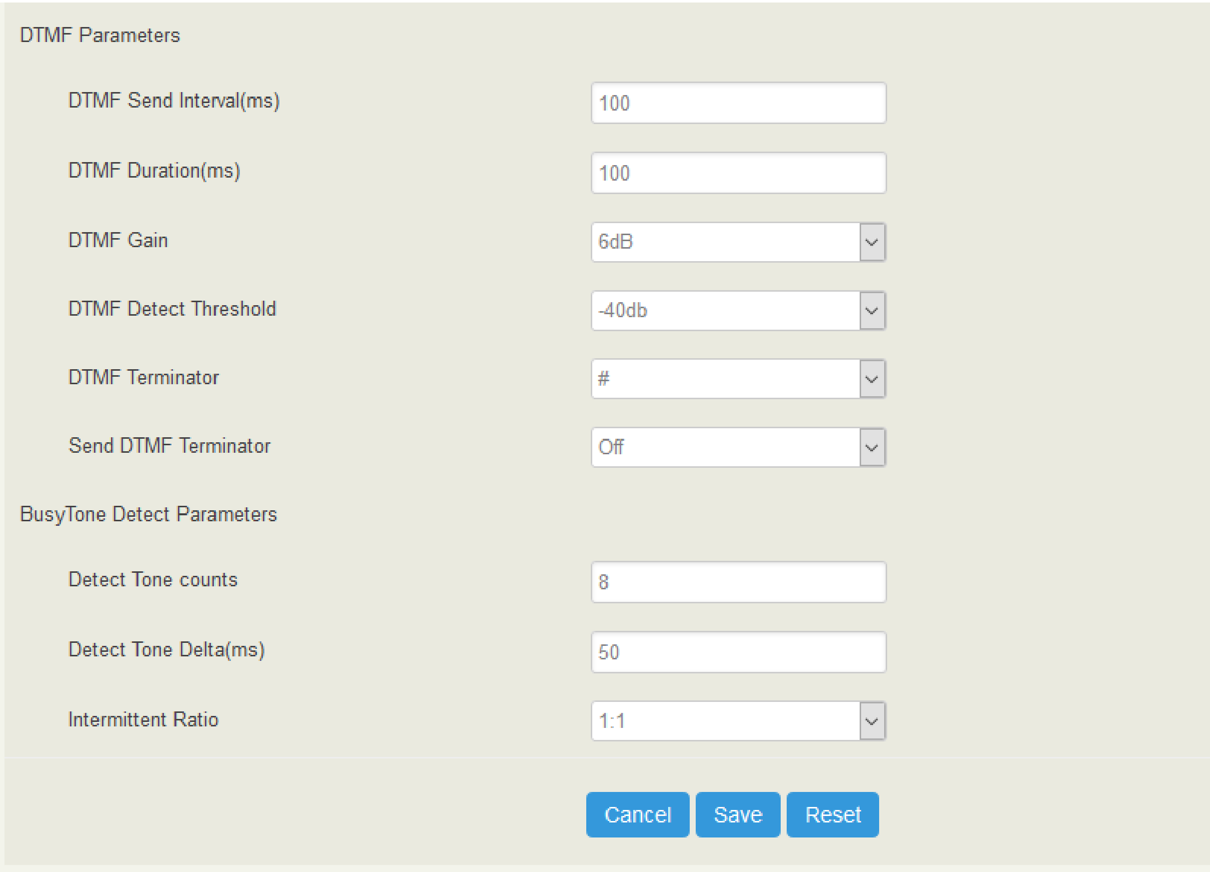
5.5.2 FXO

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.



Click and corresponding configuration interface will pop up.



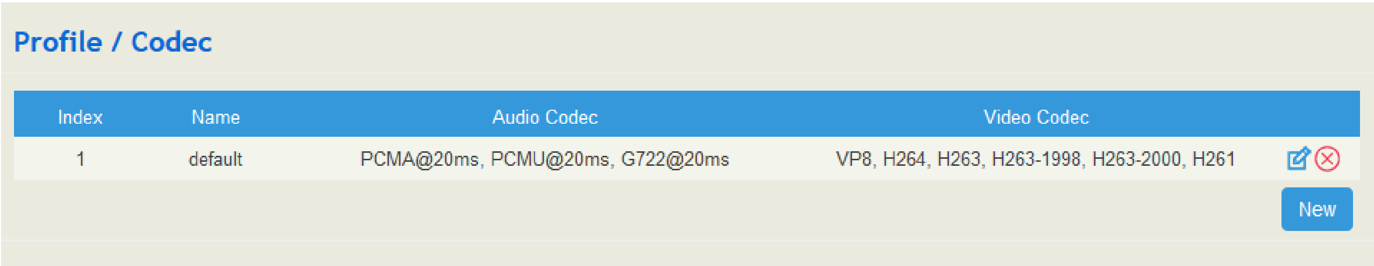


Name	The name of this FXO profile
Tone Group	The national standard of dialing tone, busy tone and ring tone; default value is China
Digit Timeout (s)	The timeout value for dialing a digit of a telephone number; When the time of dialing a digit exceeds this value, the system will think the dialing has 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

	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 Interval(ms)	The minimum interval between the sending of two DTMF tone 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

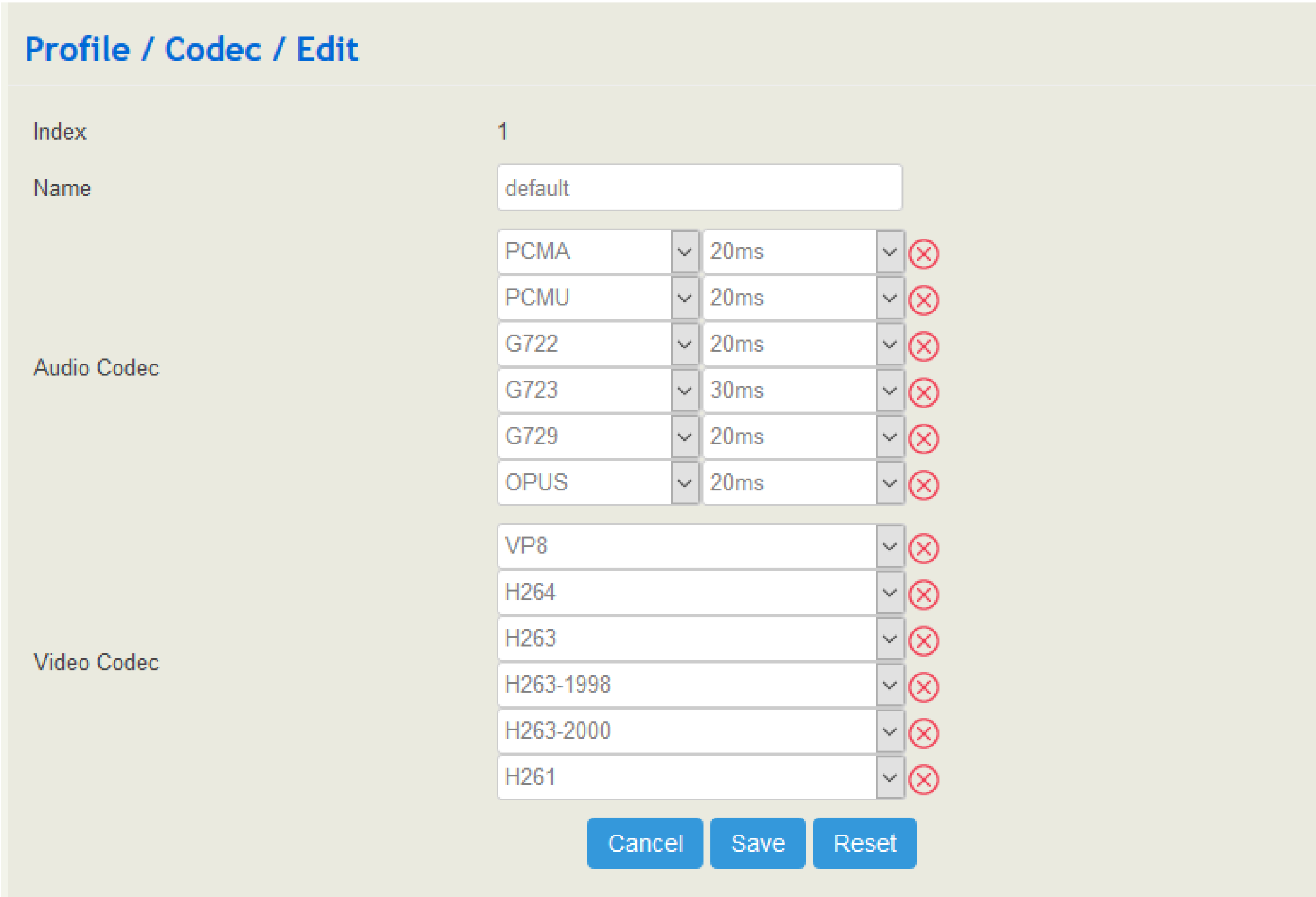
XonTel Plus supports six audio codec modes, including G729, G723, G722 PCMU PCMA and OPUS. XonTel Plus also supports six video codec modes, including VP8, H264, H263, H261, H263-1998 and H263-2000. You can adjust the priority of these modes according to you needs.



: Edit codec profile.

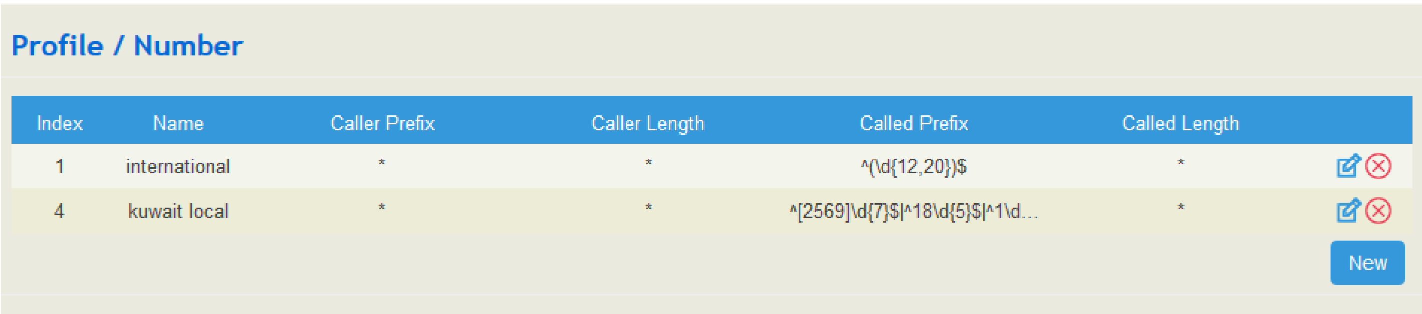
: Delete the corresponding codec profile or a codec mode.

New: Create a new codec profile



5.5.4 Number

On the **Profile →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.

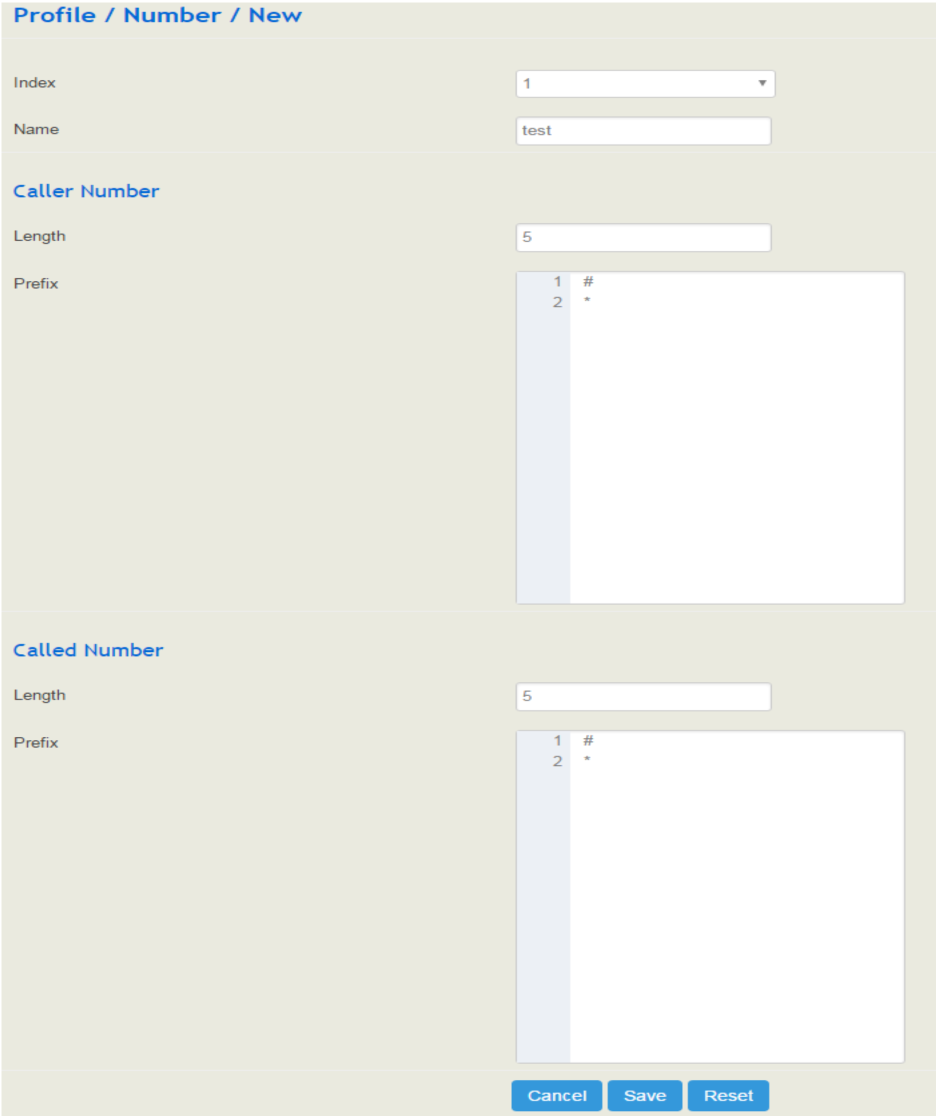


Index	Name	Caller Prefix	Caller Length	Called Prefix	Called Length
1	international	*	*	^(\d{12,20})$	*
4	kuwait local	*	*	^[2569]\d{7}$\|^18\d{5}$|^1\d...	*

: Edit number profile.

: Delete the corresponding number profile

Click New and you will see the following interface:



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.

\	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, abbbc, 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.
\S	Mark any character that is not a blank character

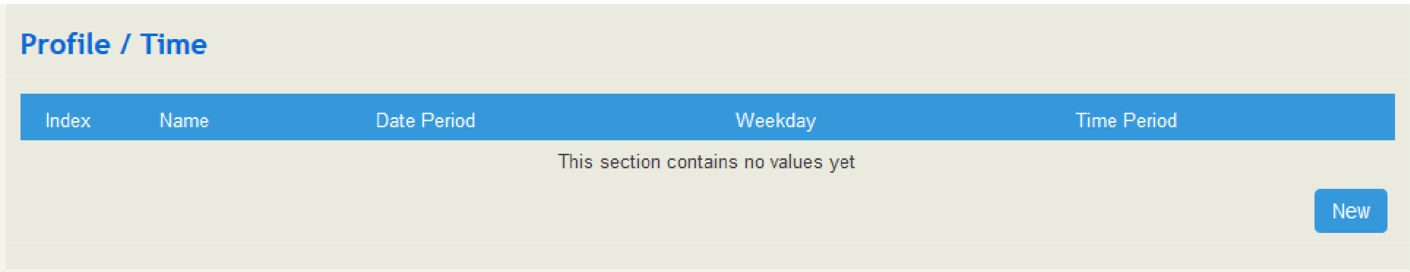
Examples:

^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.

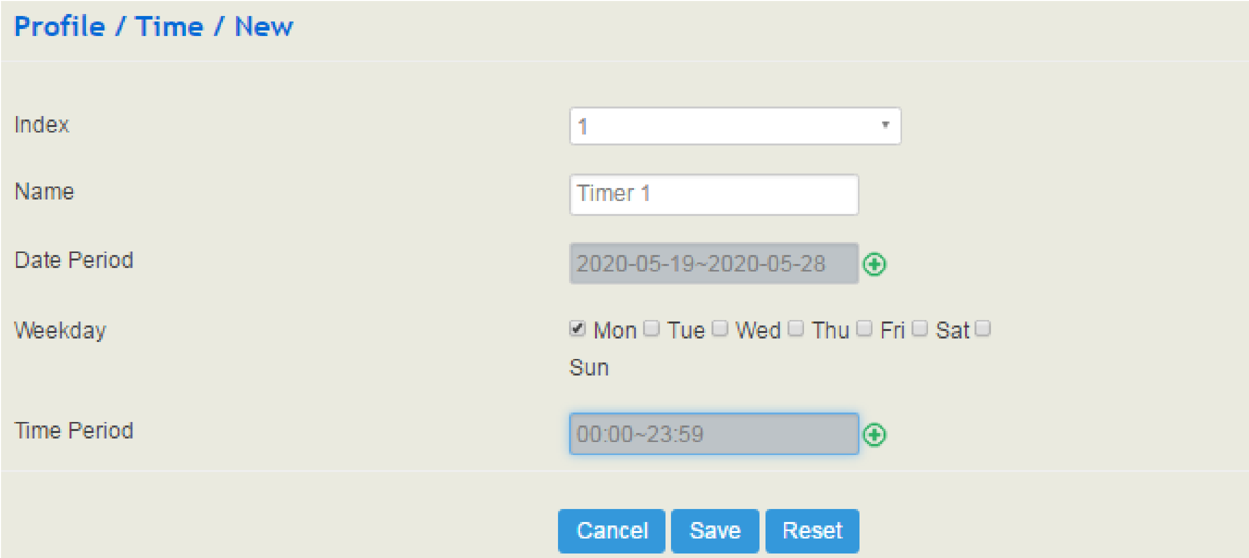
Note: the matching of number prefix also supports some digits that are not conform to the format of regular expression. For example, 0755 matches the numbers starting with 0755, and 0755|8899|0110 matches the numbers starting with 0755, 8899 or 0110.

5.5.5 Time

On the **Profile → 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.



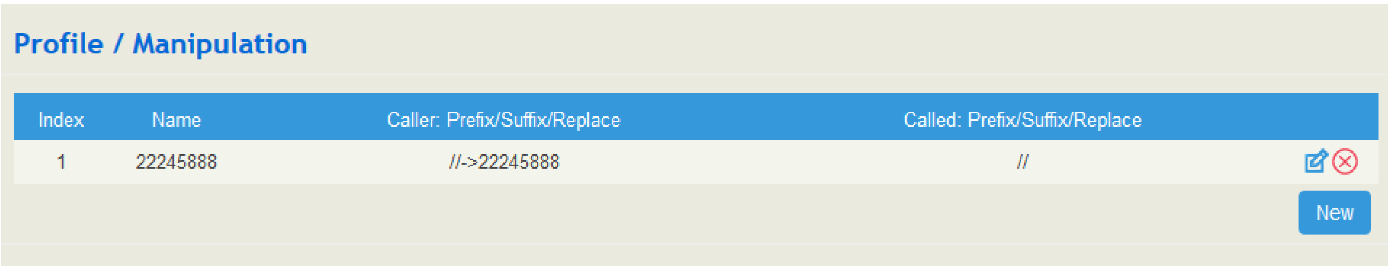
Click the **New** button, and you will see the following interface:



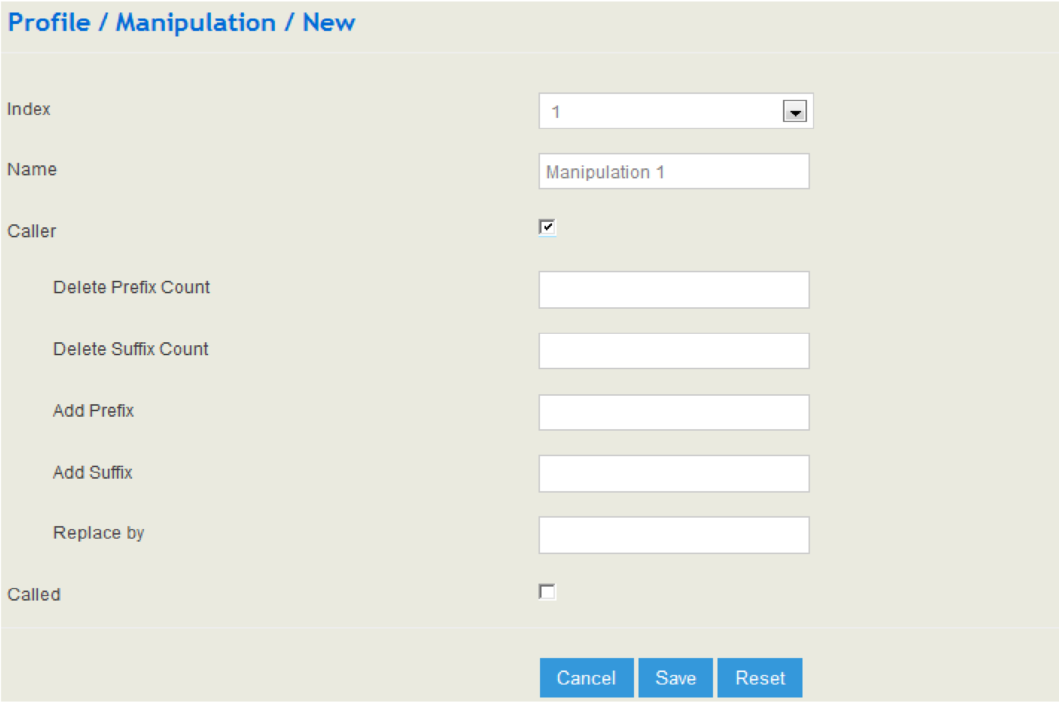
Name	The name of the number profile
Date Period	Configure the starting date and ending date of a period : Add a date period : Delete a date period
Weekday	Choose the desired week days
Time Period	Choose the desired starting time and ending time of the day

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.



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

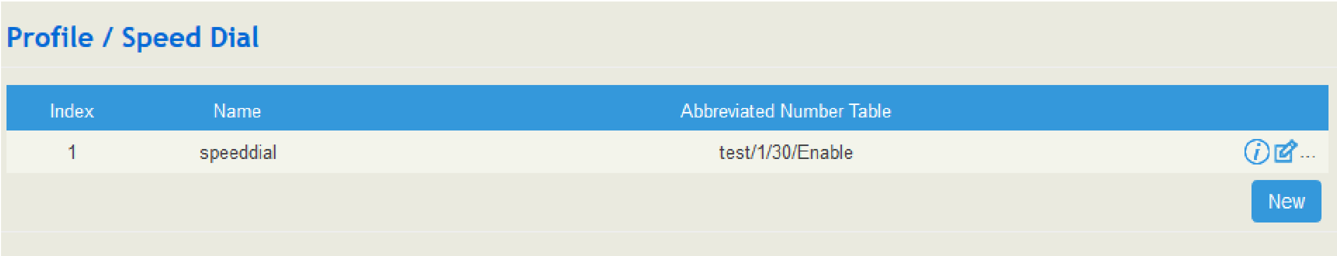


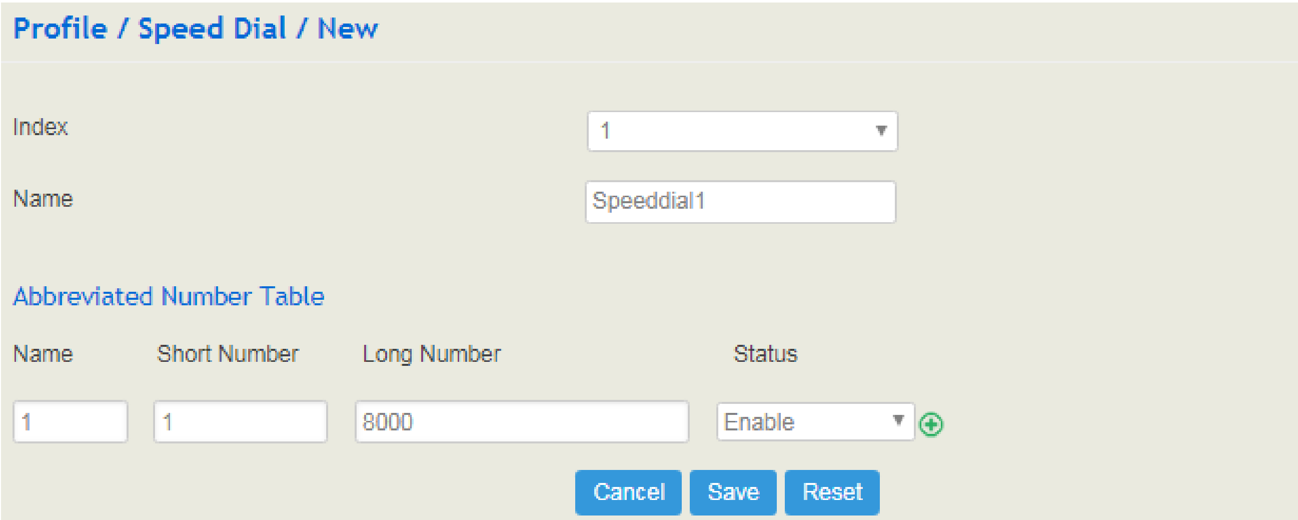
Name	The name of this manipulation profile
Delete Prefix Count	The number of digits that are deleted from the left of the caller number or calling number
Delete Suffix Count	The number of digits that are deleted from the right of the caller number or calling number
Add Prefix	The prefix added to the 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
	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.

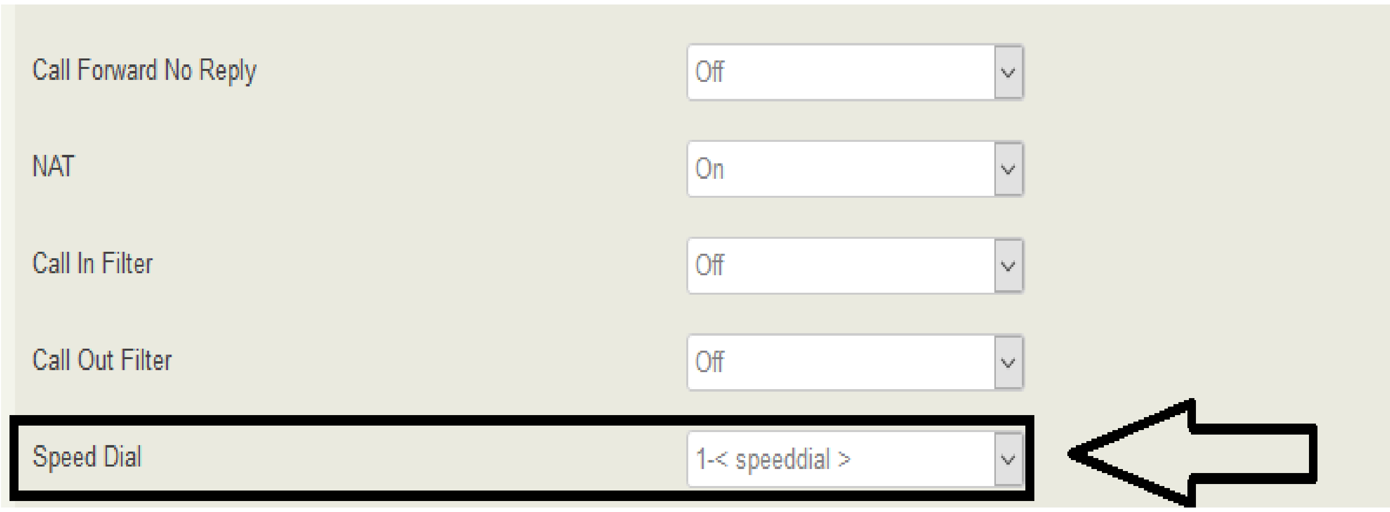
5.5.7 Speed Dial

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.



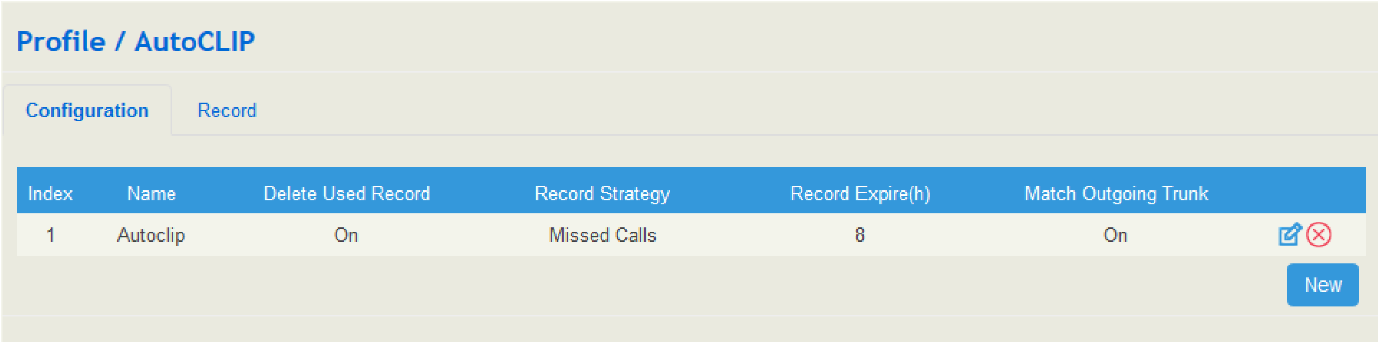


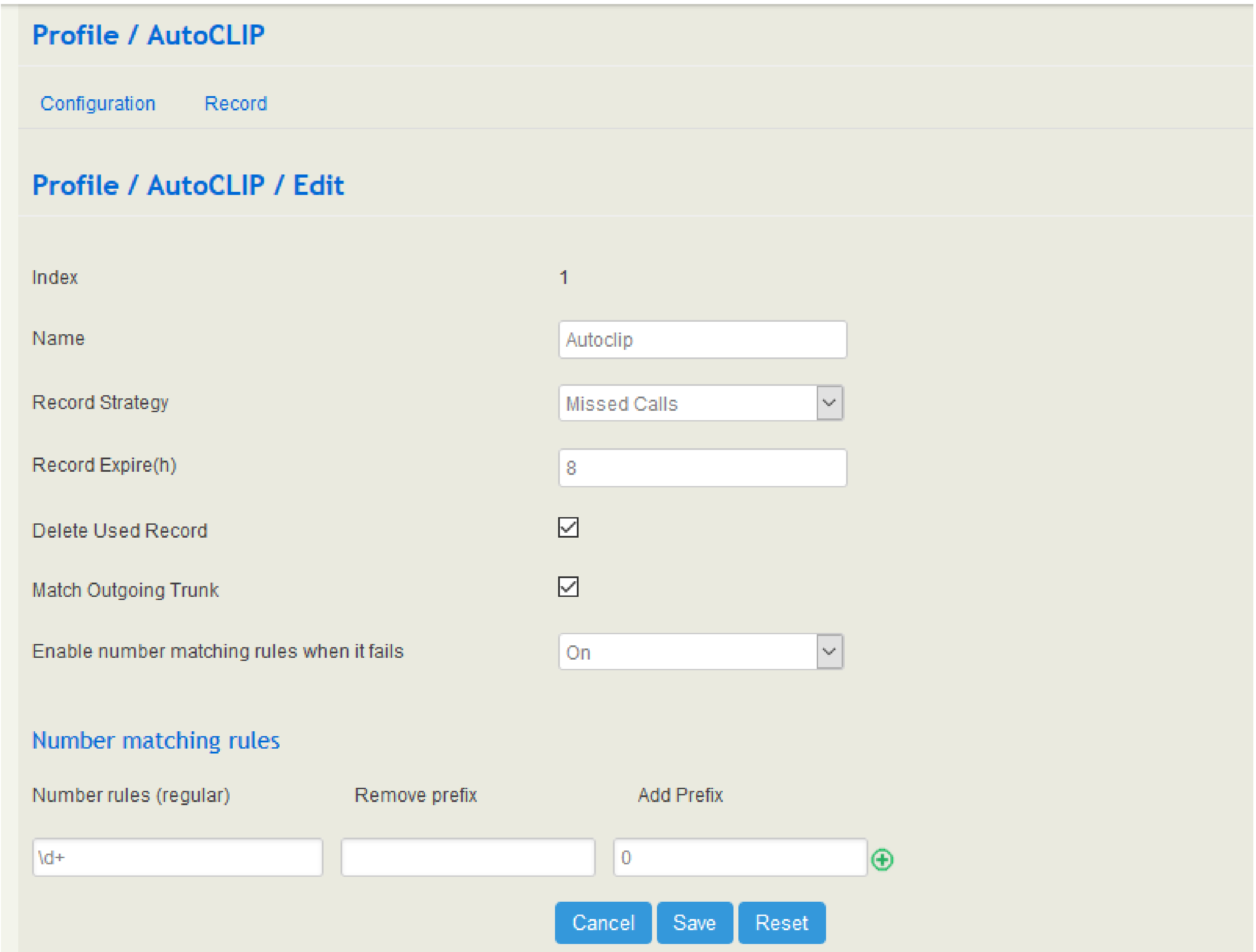
After that you can enable the Speed Dial profile in the extension settings as shown below.



5.5.8 AutoCLIP

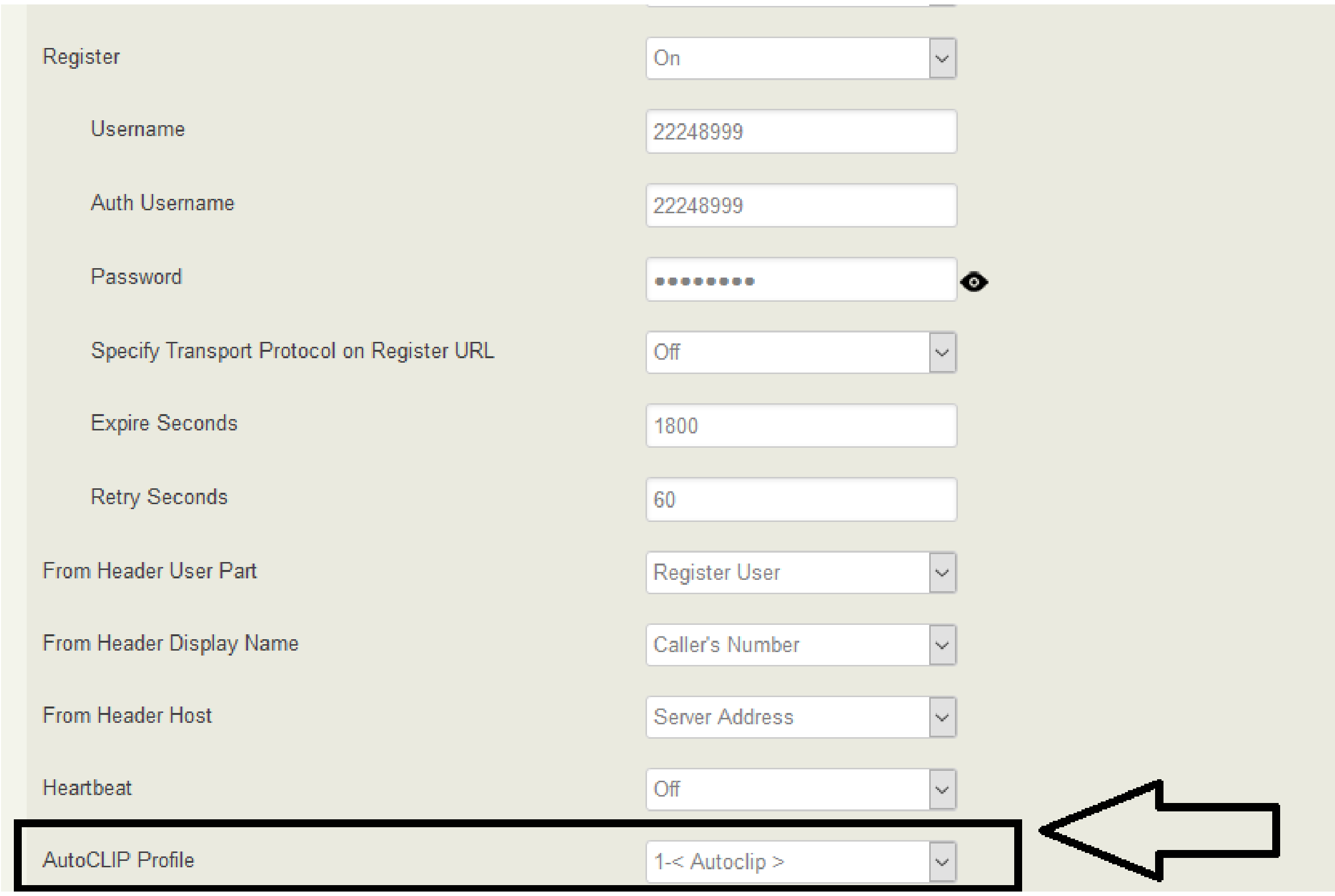
AutoCLIP is mainly used to SIP trunks and FXO trunks. AutoCLIP helps record the outgoing and incoming calls of a trunk.



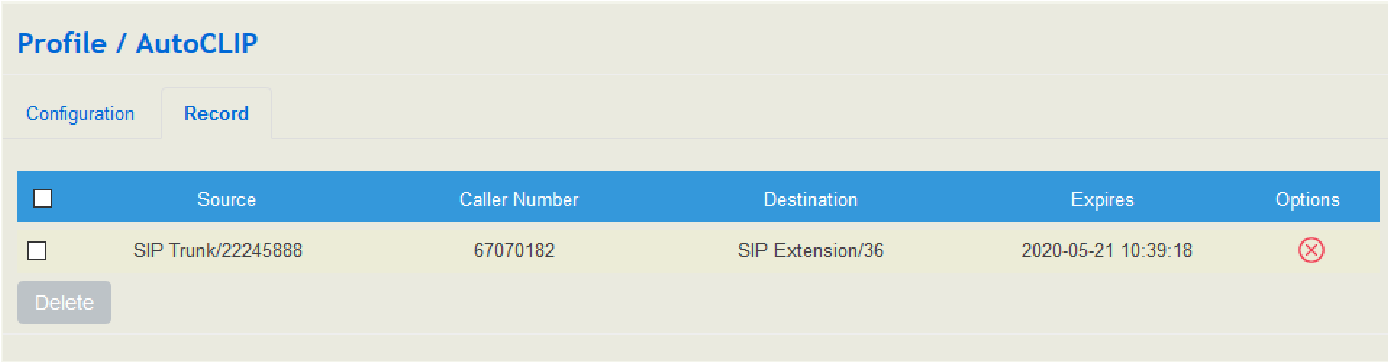


Index	The index of AutoCLIP profile
Name	The name of AutoCLIP profile
Record Strategy	You can choose missed calls or all calls. If missed calls is selected, XonTel Plus will record the missed calls of the trunk. If all calls are selected, all the calls going through the trunk will be recorded
Record 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 **0505443281** out, but **0505443281** call back with caller number is **505443281**, 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



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

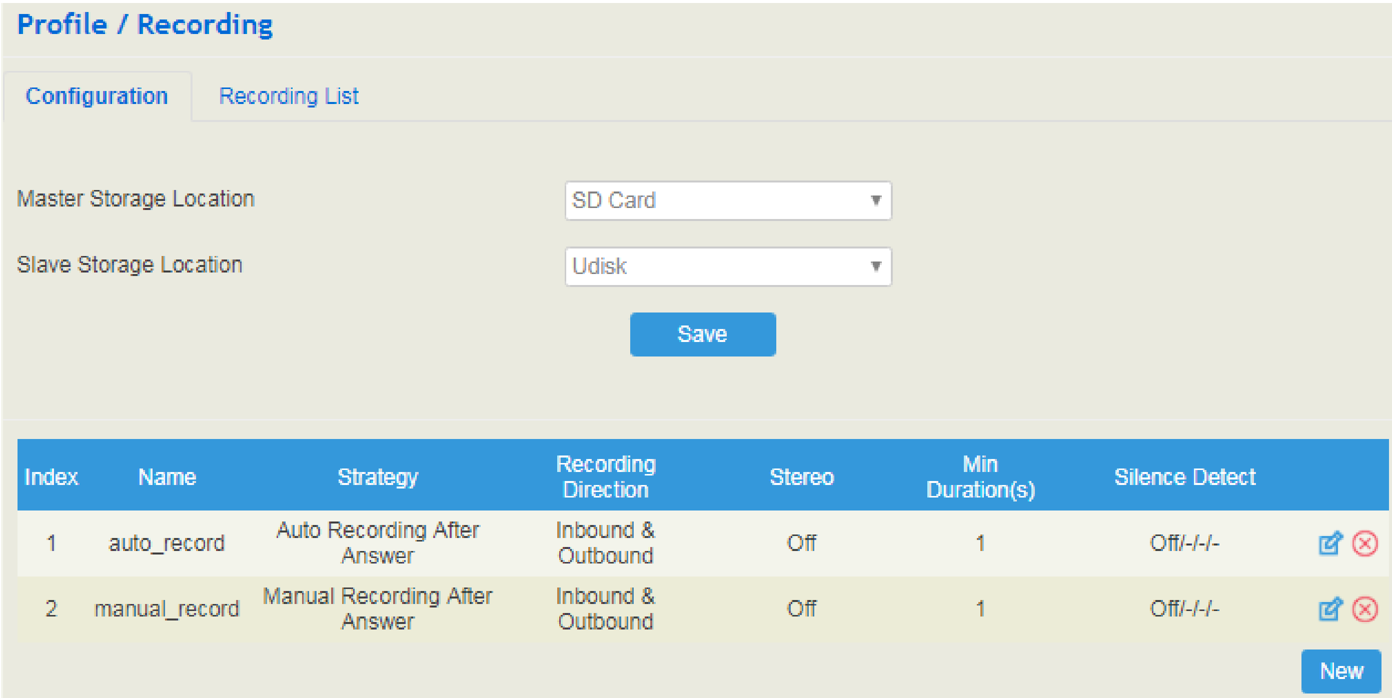


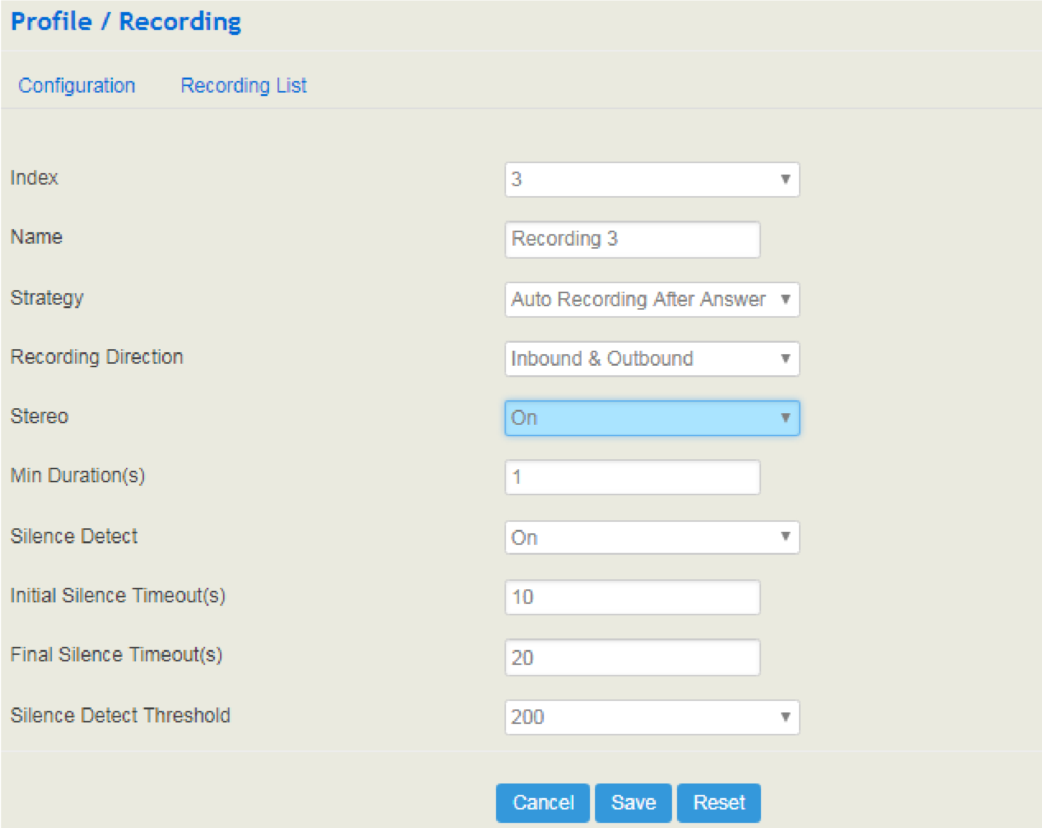
5.5.9 Recording

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.

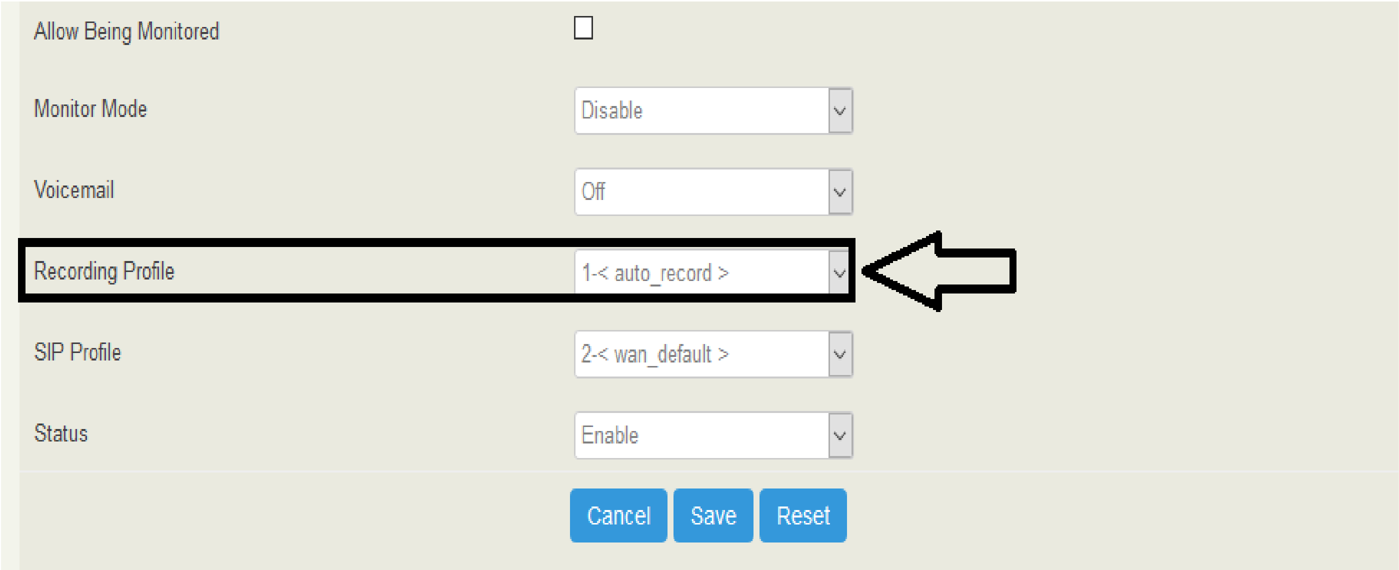




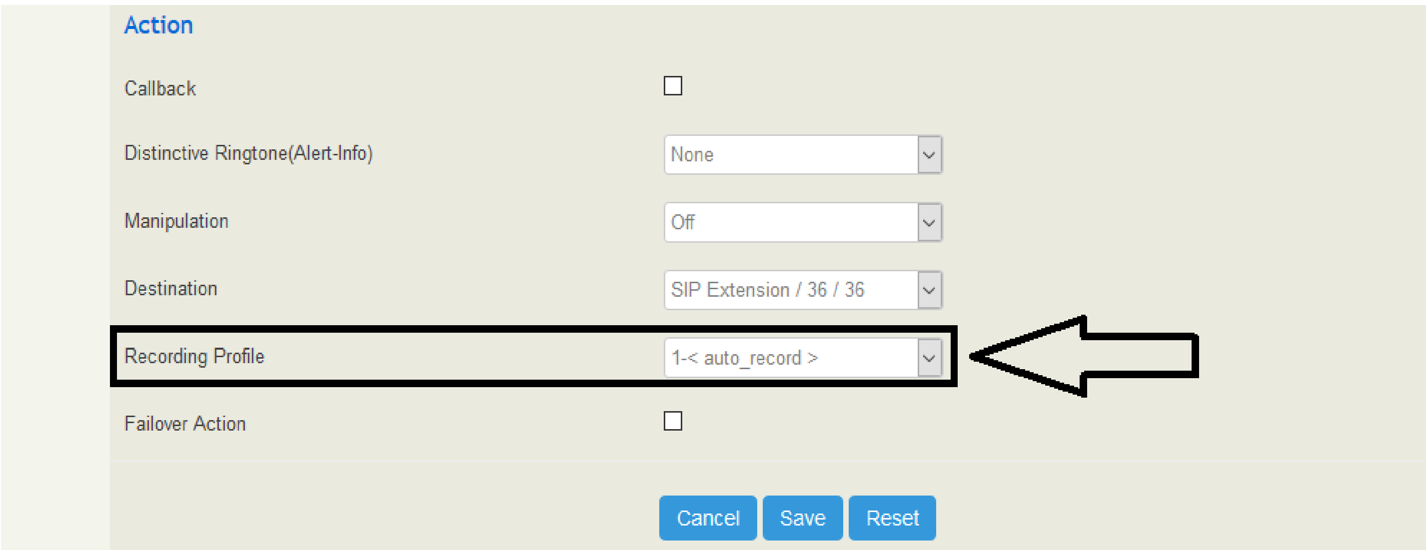
Index	The index of the recording profile. Range: 1-32
Name	The name of the recording profile, used to identify the recording profile
Strategy	Auto Recording after Answer: start recording after the callee pick up the phone. Ban Recording: ether caller or callee enables his function, and then the call in both directions will not be recorded. Manual Recording after Answer: press *1 to start recording after the callee answers the call.
Recording Direction	Inbound & Outbound: If this recording profile is added to SIP extension, both inbound and outbound calls will be recorded. Inbound: If this recording profile is added to SIP extension, only inbound calls will be recorded. Outbound: If this recording profile is added to SIP extension, only outbound calls will be recorded. Note: If this recording profile is added to routing, this parameter is invalid and all calls going through the routing will be recorded.
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.

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

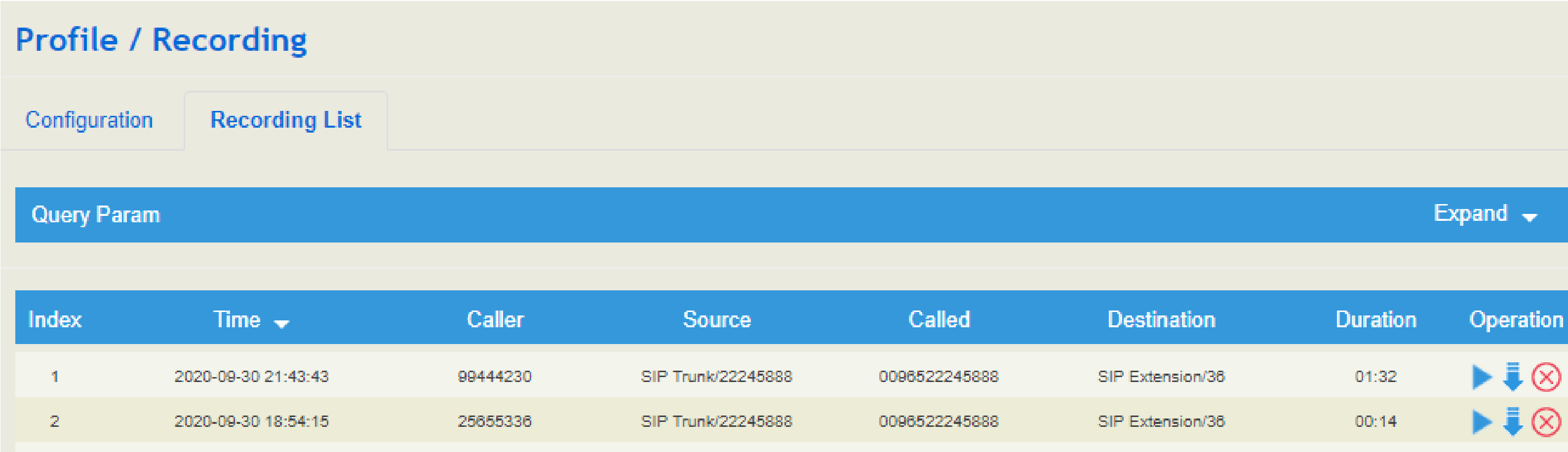
Enable calls recording for the extension.



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

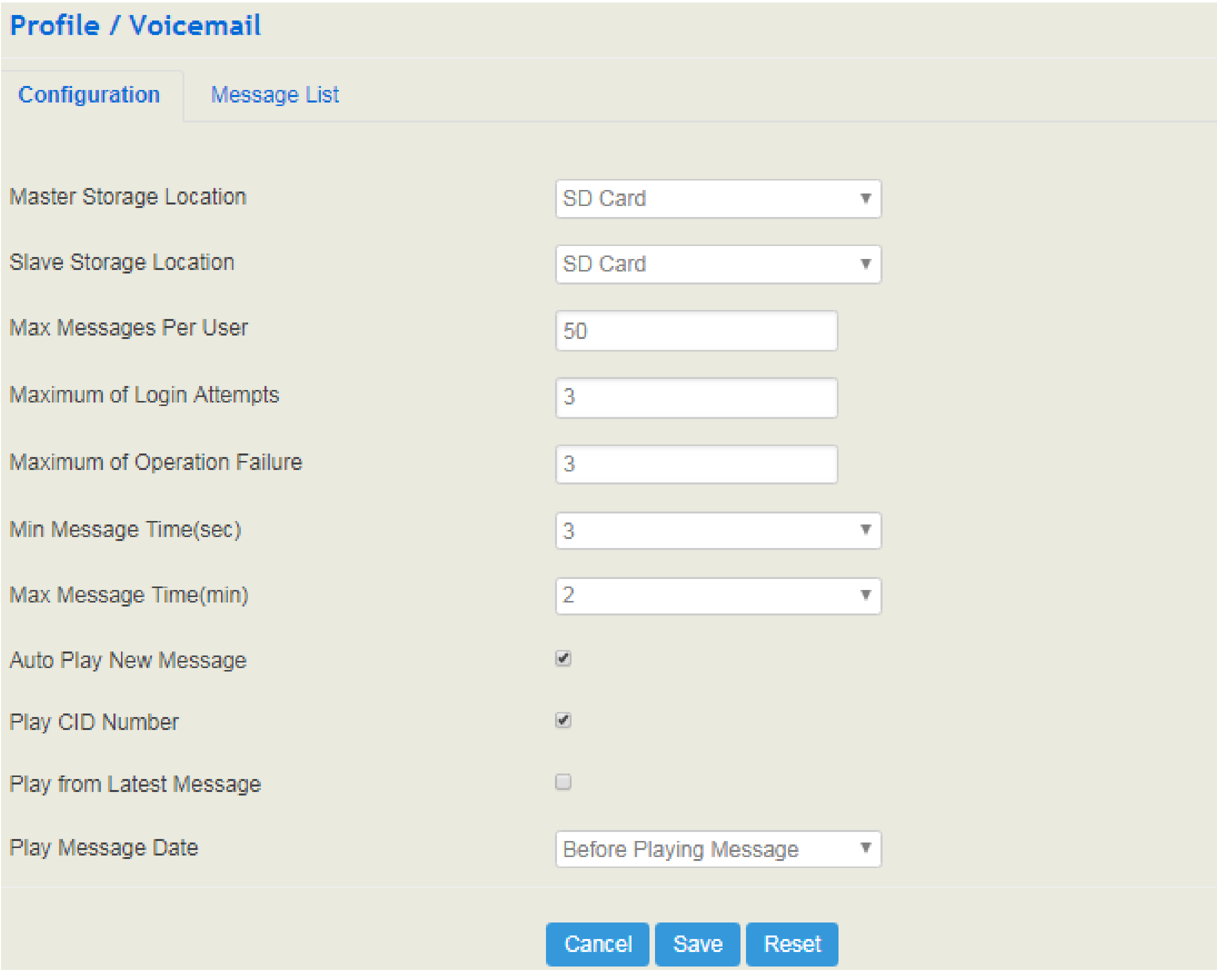


You can click **Recording List** in the recording profile to view the recording files which show the caller/called number, recording duration and so on. You can also play, download or delete the recording files on this interface.



5.5.10 Voicemail

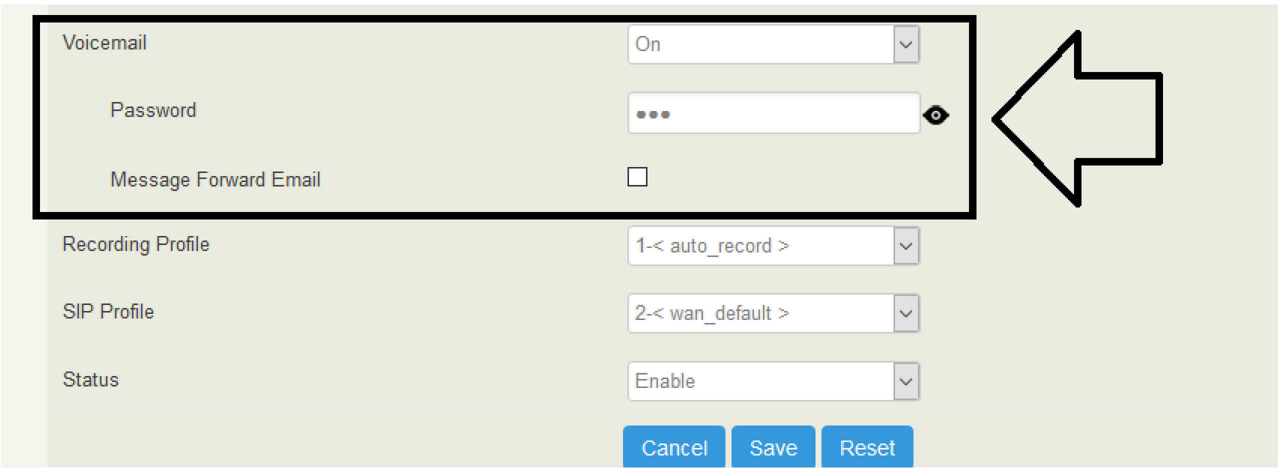
On the **Profile→ Voicemail** interface, you can configure the location, number and duration of a voicemail.



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:

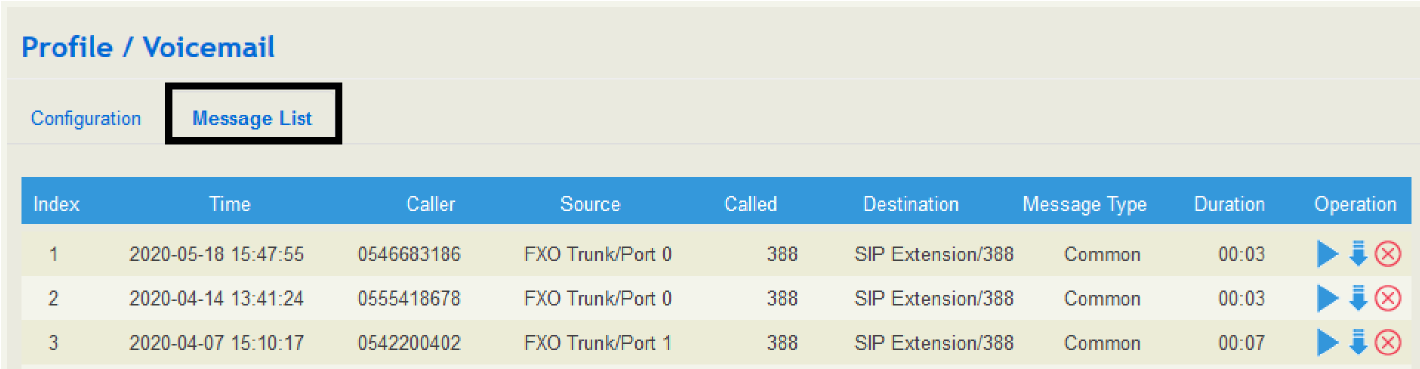
Go to the **Extension → SIP** interface, click **New** to create new SIP extension and enable the voicemail function for it, and then calls that times out will enter into voicemail.



<u>Note</u>

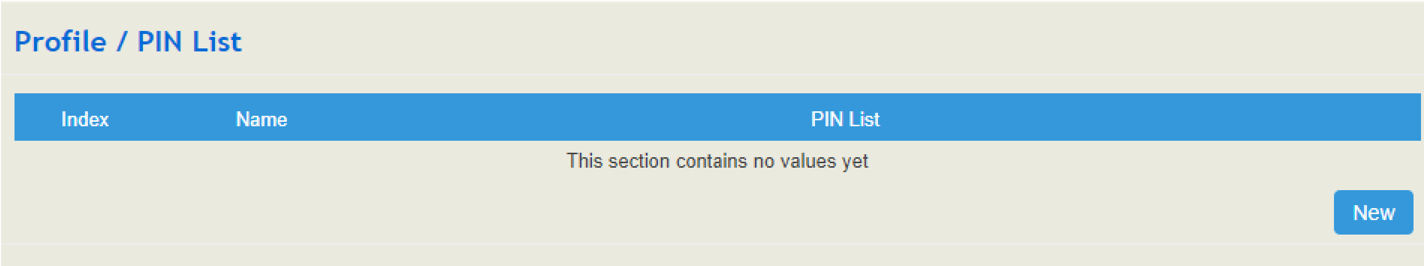
To use **Message Forward Email (voicemail to Email)** option, please configure PBX **Email settings**.

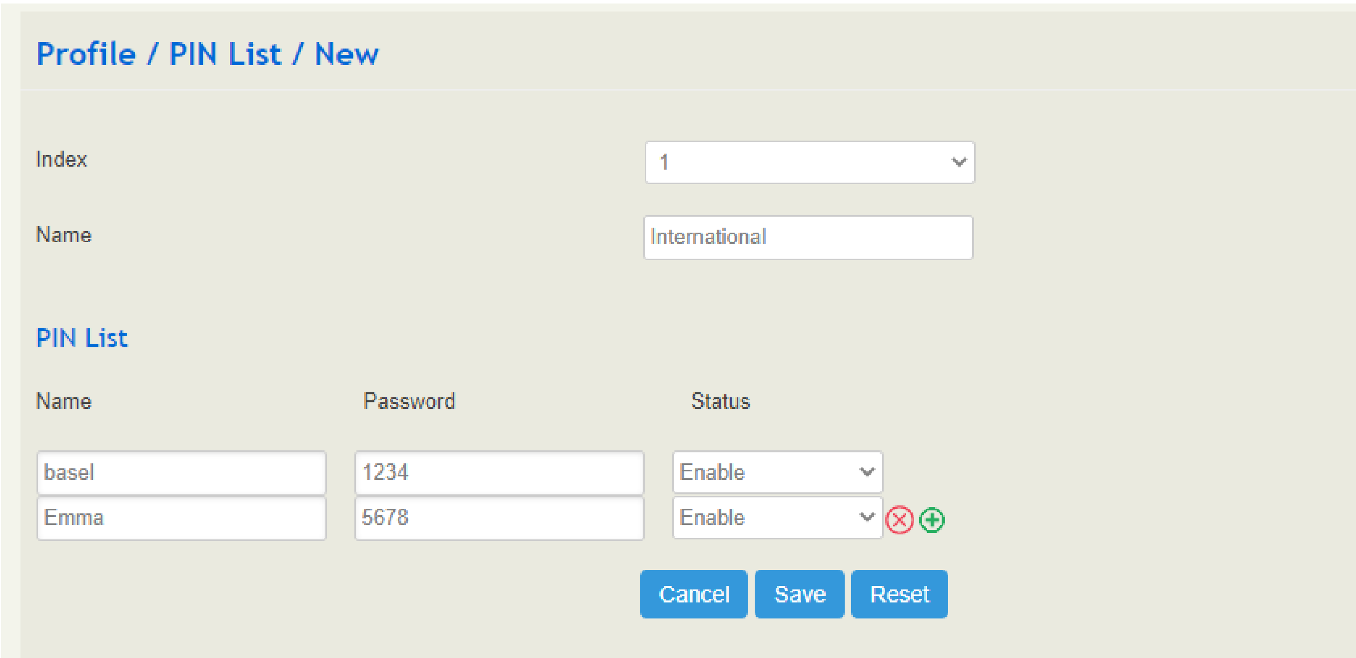
You can click **Message List** in the voicemail profile to view the voicemail files which show the caller/called number, message duration and so on. You can also play, download or delete the message files on this interface.



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.

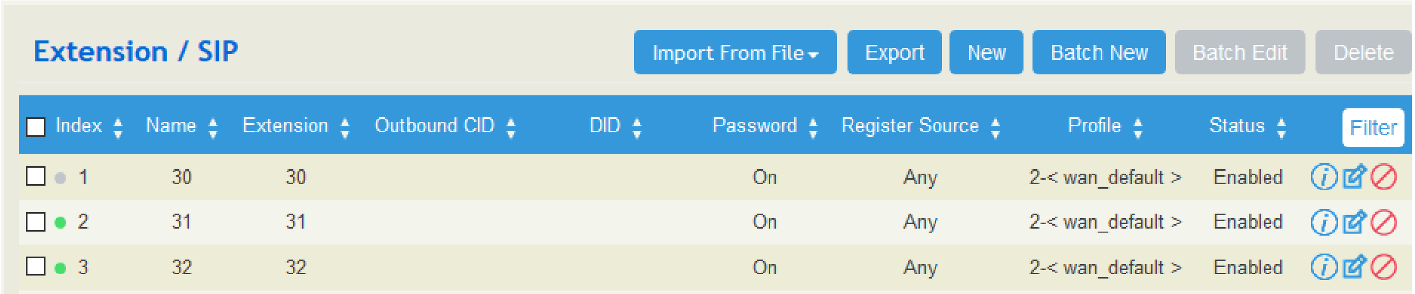


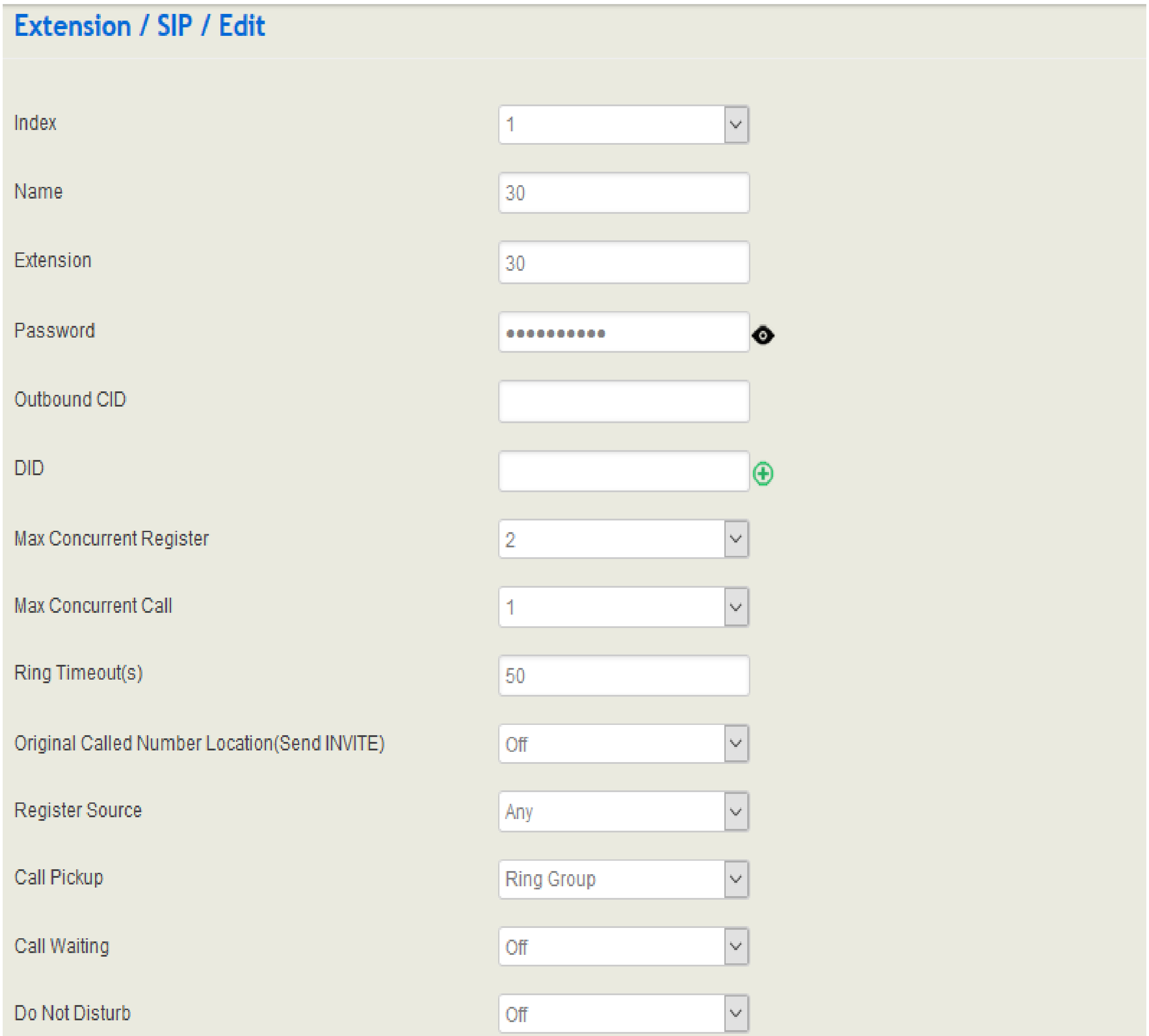


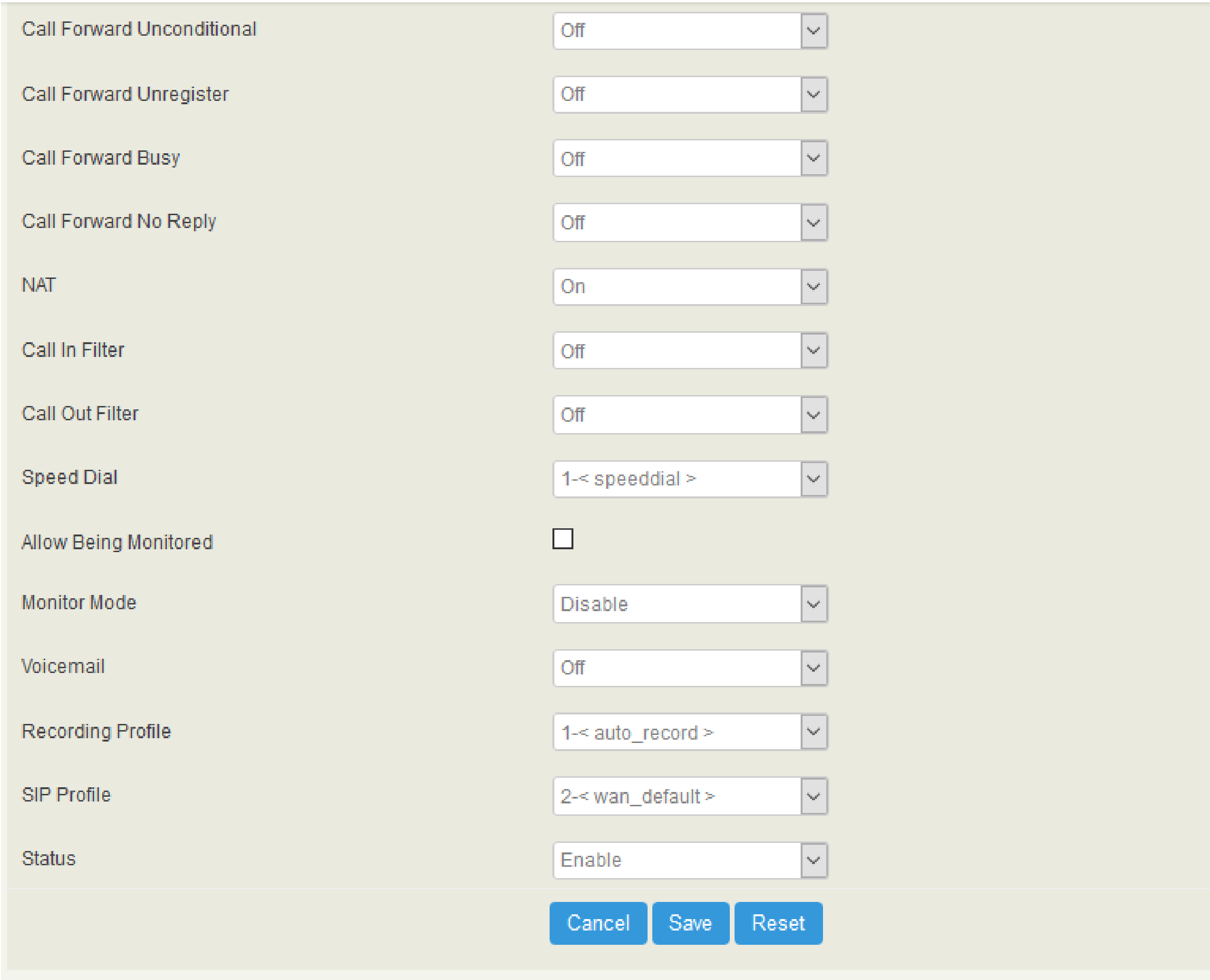
5.6 Extension

5.6.1 SIP

On the **Extension → 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).







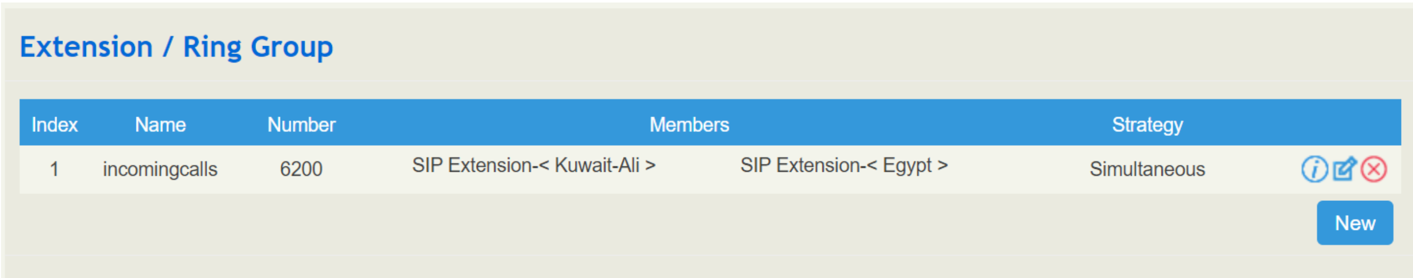
Name	The name of this SIP extension
Extension	The SIP account of the extension registered in XonTel Plus by a SIP client
Password	The password of the SIP account registered in XonTel Plus by a SIP client
Outbound DID	Outbound Direct Inward Dialing. Overrides the caller ID when dialing out a trunk. Leave 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. **SIP forking** 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.

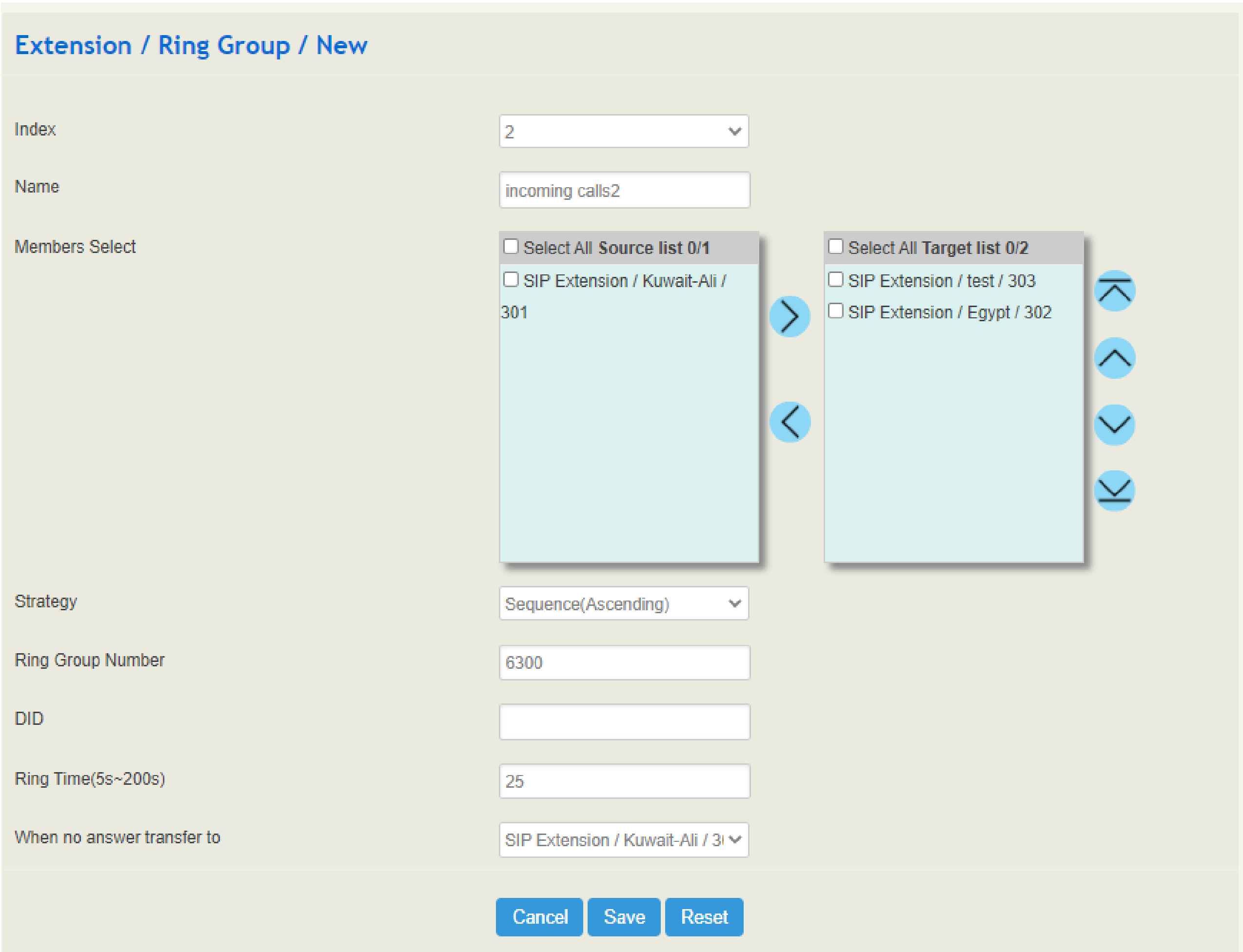
Register Source	If 'Any' is chosen, all SIP clients are allowed to register the SIP account of this extension; if 'Specified' is chosen, only the SIP client with the specified IP address or network segment is allowed to register the SIP account of this extension.
Call Waiting	If a calling party places 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.
Call Pickup	Allows extension to answer another extension incoming call. **Ring Group**: Extension can pick up the call that is ringing at the other extension **that is in the same ring group only**. **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.
Monitor Mode	Decide how you will monitor another extension's current call. □ **Disable**: you will not be allowed to monitor other's call. □ **Listen Mode**: you can only listen to the call, but can't talk (default feature code: ***222**). □ **Whisper Mode**: you can talk to the extension you're monitoring without being heard by the other party (default feature code: ***223**). □ **Barge-in Mode**: you can talk to both parties (default feature code: ***224**).
Voicemail	Check this box to enable voicemail for this extension. **Password**: 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.

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 → Ring Group** interface, you can group SIP extensions together and set strategy for choosing the SIP extension to ring under a ring group.

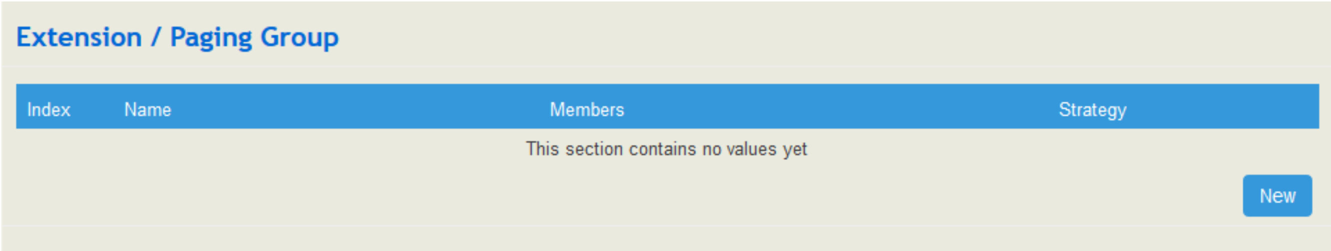


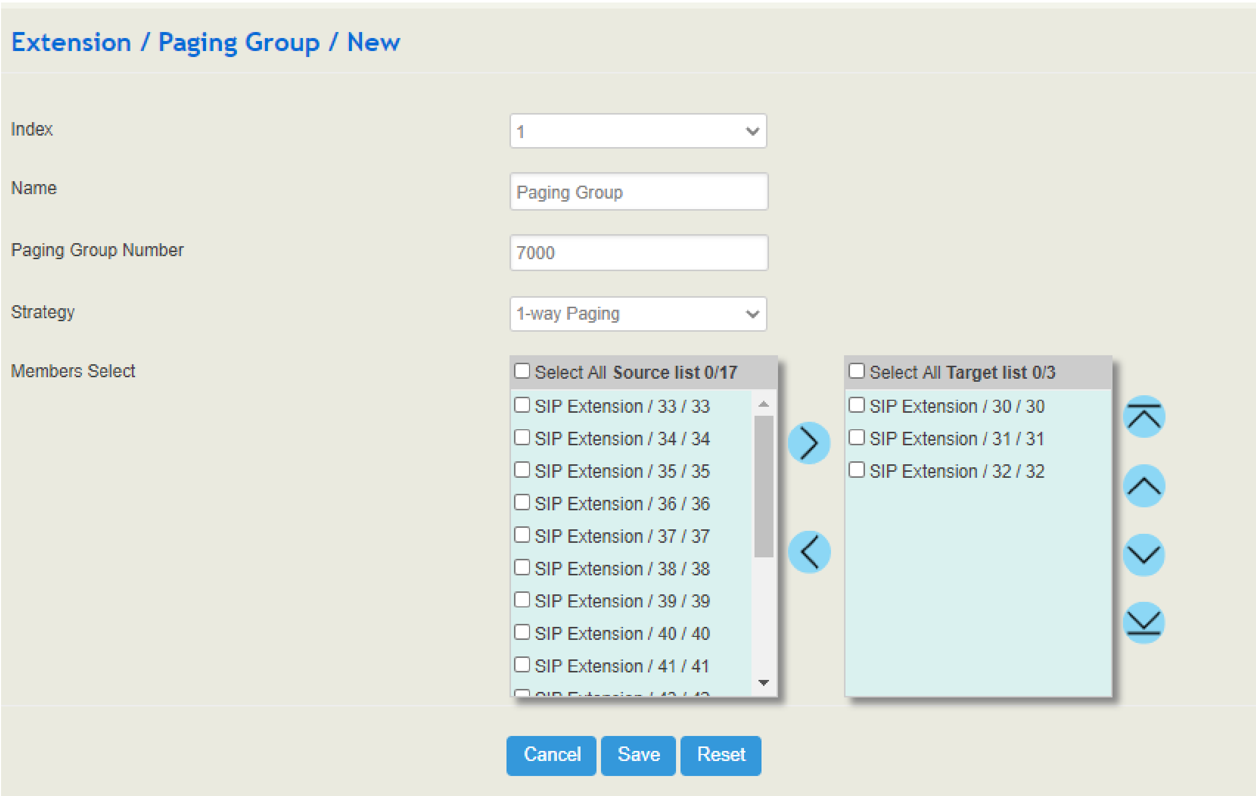


Name	The name of this ring group
Members Select	Select the SIP extension or several SIP extensions. Add extension to the ring group by adding it to the target list.
Strategy	The strategies for choosing which SIP extension to ring, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous 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

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.



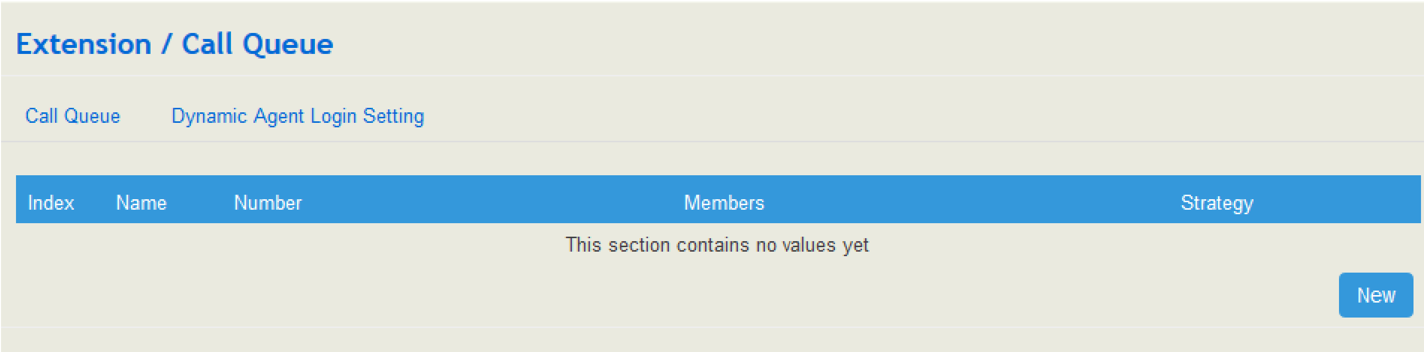


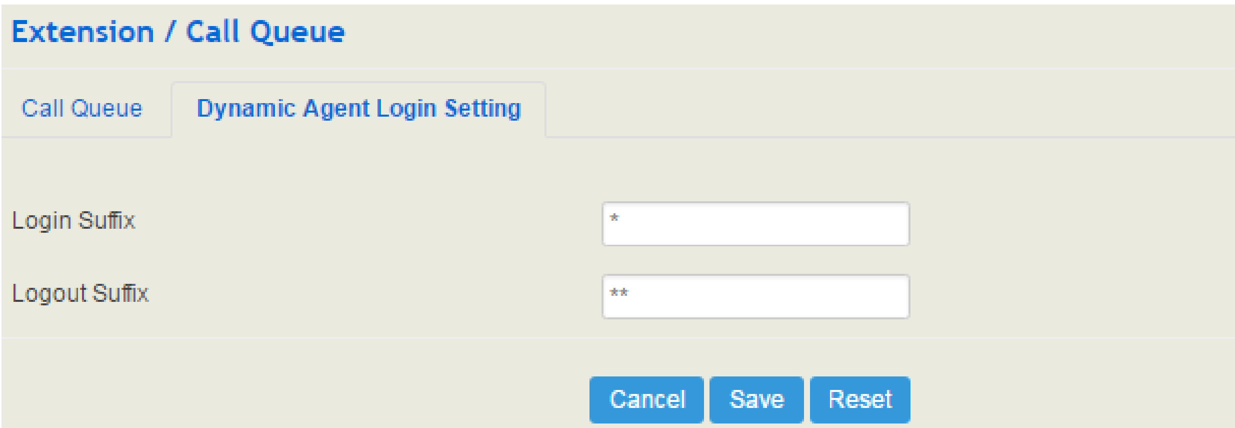
Name	The name of this paging group
Paging Group Number	The number of the paging group.
Strategy	**1-way paging**: members of the paging group only can listen to the voice of presenter and cannot answer the call. **2-way intercom**: members of the paging group can have conversation with the presenter, but members cannot talk to each other.
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.

5.6.4 Call Queue

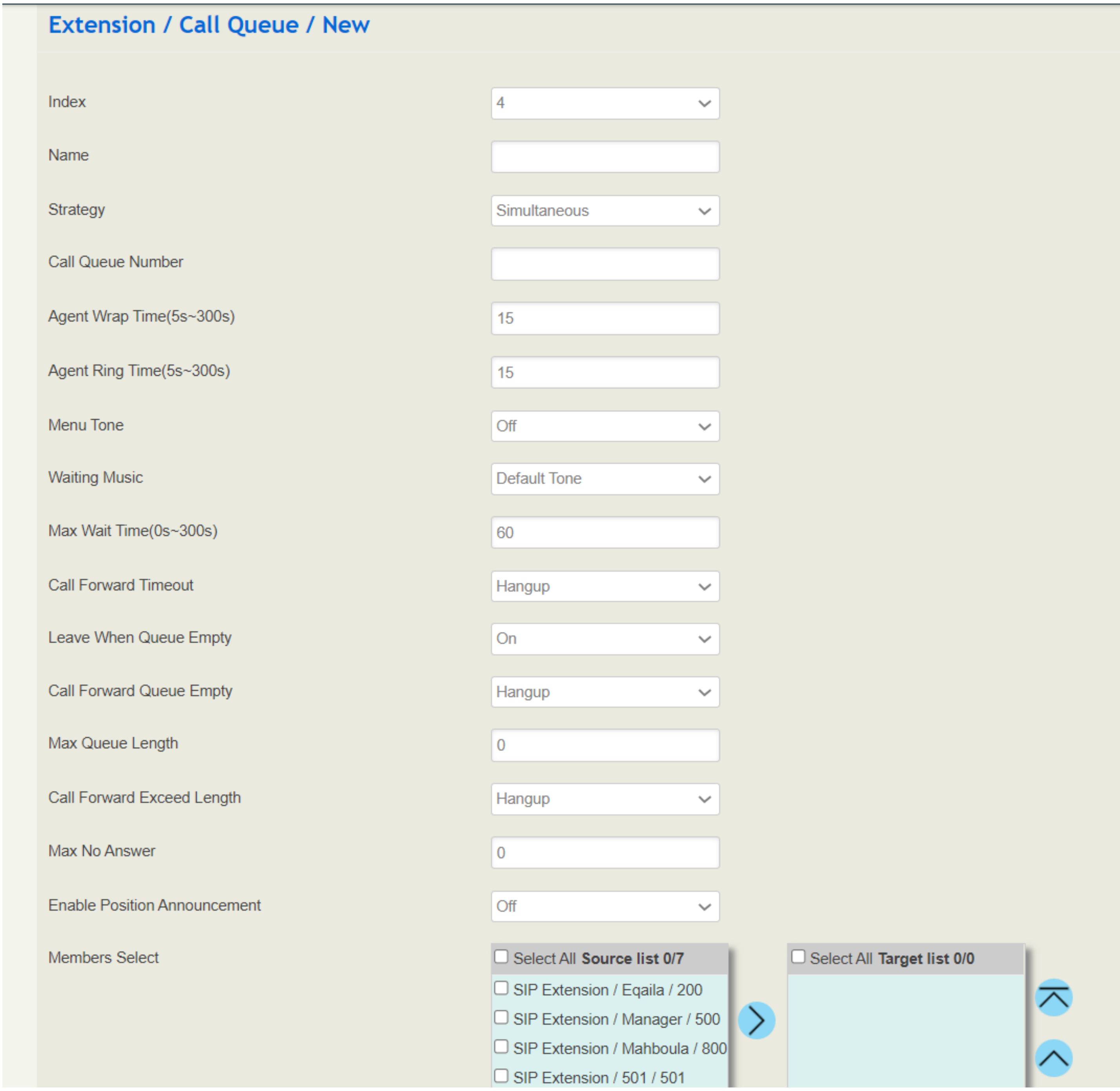
On the **Extension →Call Queue** interface, you can set a strategy of how to handle the queue, members that answer the queue, waiting music and other parameters for a call queue.

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.





- **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)
15
Agent Ring Time(5s~300s)
15
Menu Tone
Off
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
SIP Extension / Eqaila / 200
SIP Extension / Manager / 500
SIP Extension / Mahboula / 800
SIP Extension / 501 / 501
Select All Target list 0/0

Name	The name of this queue
Strategy	Simultaneous: All available agents will ring simultaneously until one answer. Liner: rings agents in the order specified in the queue configuration. Random: ring a random agent. Memory Round Robin: Round Robin with Memory, remembers where it left off in the last ring pass. Least Recent: ring the agent which was least recently called. Fewest Calls: ring the agent with the fewest completed calls.
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.

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	

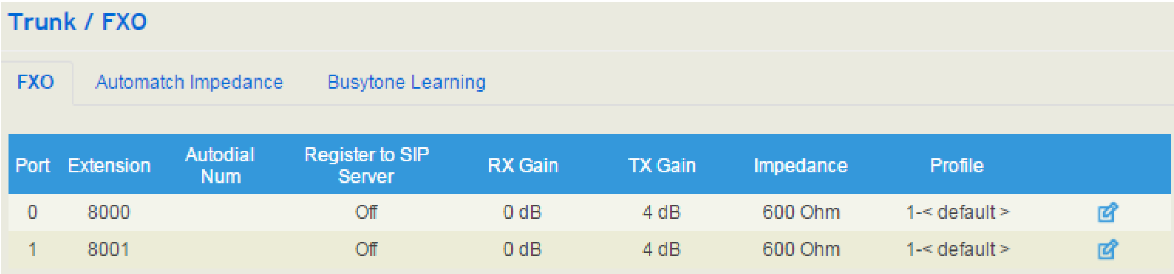
New

Name	
Address	
Port	
Outbound Proxy	
Port	
Transport	UDP
Register	On
Username	
Auth Username	
Password	
Specify Transport Protocol on Register URL	Off
Expire Seconds	1800
Retry Seconds	60
From Header User Part	Caller's Number
From Header Display Name	Caller's Number
From Header Host	Local Address
Heartbeat	Off
AutoCLIP Profile	Off
DNIS	Off
SIP Profile	1-< lan_default >
Outbound Codec Profile	1-< default >
Extra Param	
Status	Enable

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 → 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 "**user=phone**", 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.

5.7.2 FXO

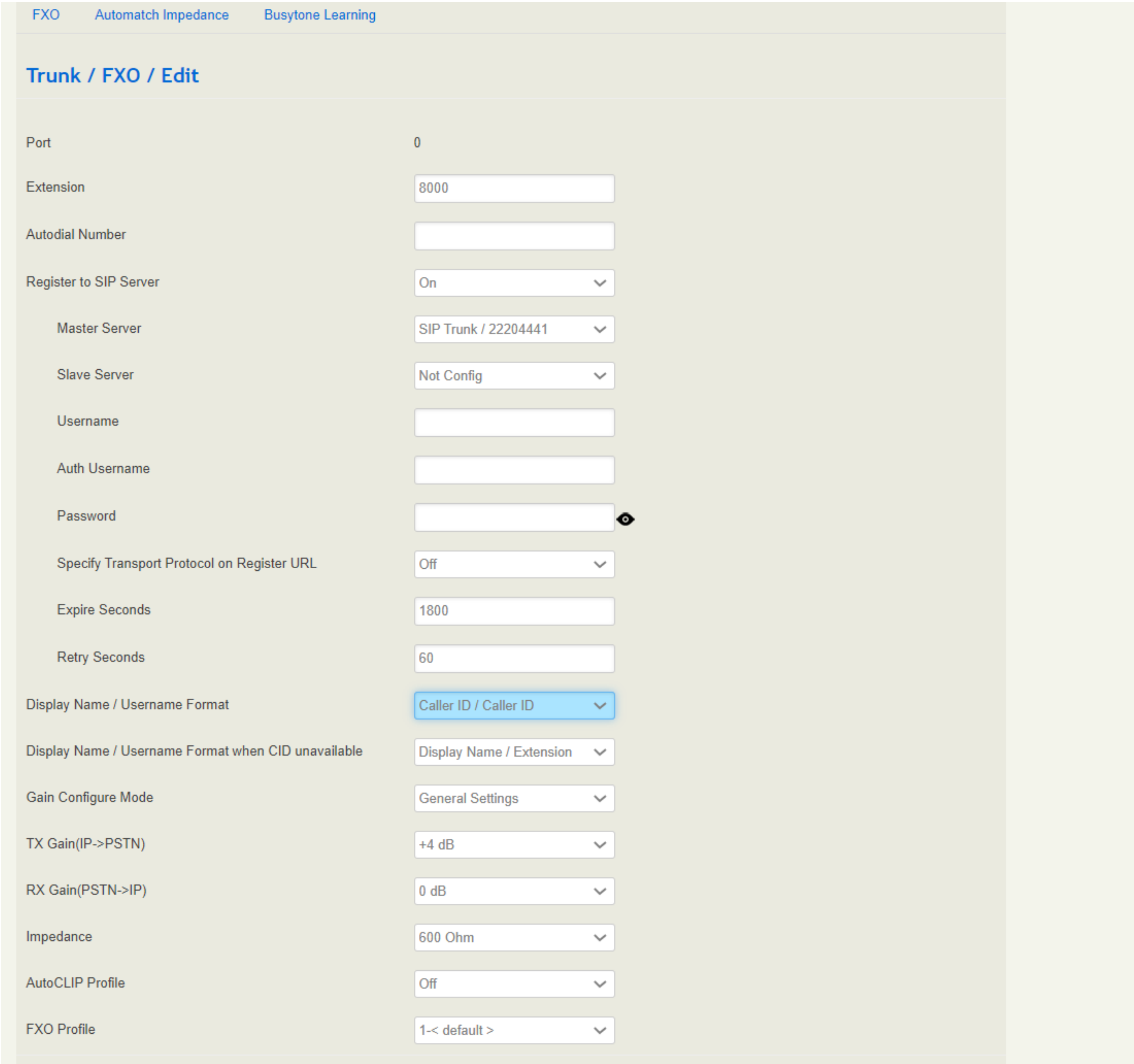
FXO Trunk interconnects the PSTN with XonTel Plus. Calls from the PSTN can come into the PBX and calls can go out from the PBX to search telephone numbers under the PSTN.



Trunk / FXO

FXO | Automatch Impedance | Busytone Learning

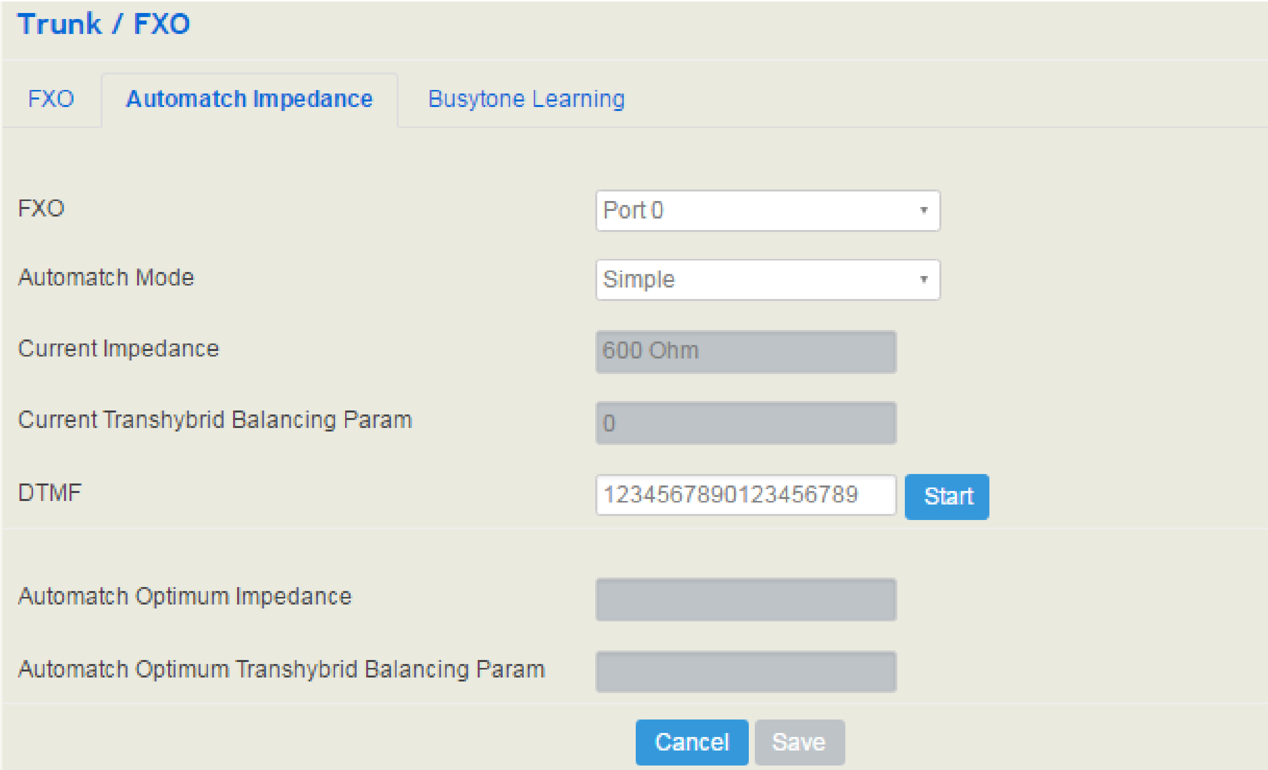
Port	Extension	Autodial Num	Register to SIP Server	RX Gain	TX Gain	Impedance	Profile
0	8000		Off	0 dB	4 dB	600 Ohm	1-< default >
1	8001		Off	0 dB	4 dB	600 Ohm	1-< default >



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 → 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→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.

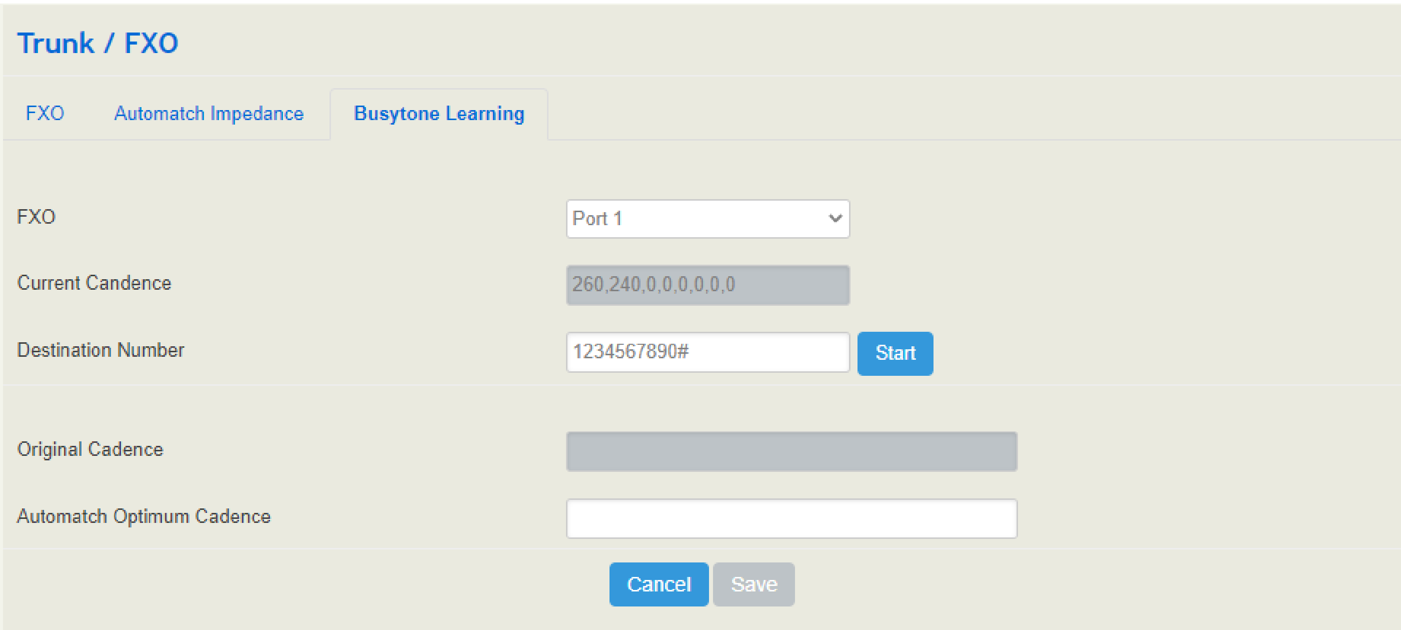
FXO Automatch Impedance:

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



FXO Busytone Learning:

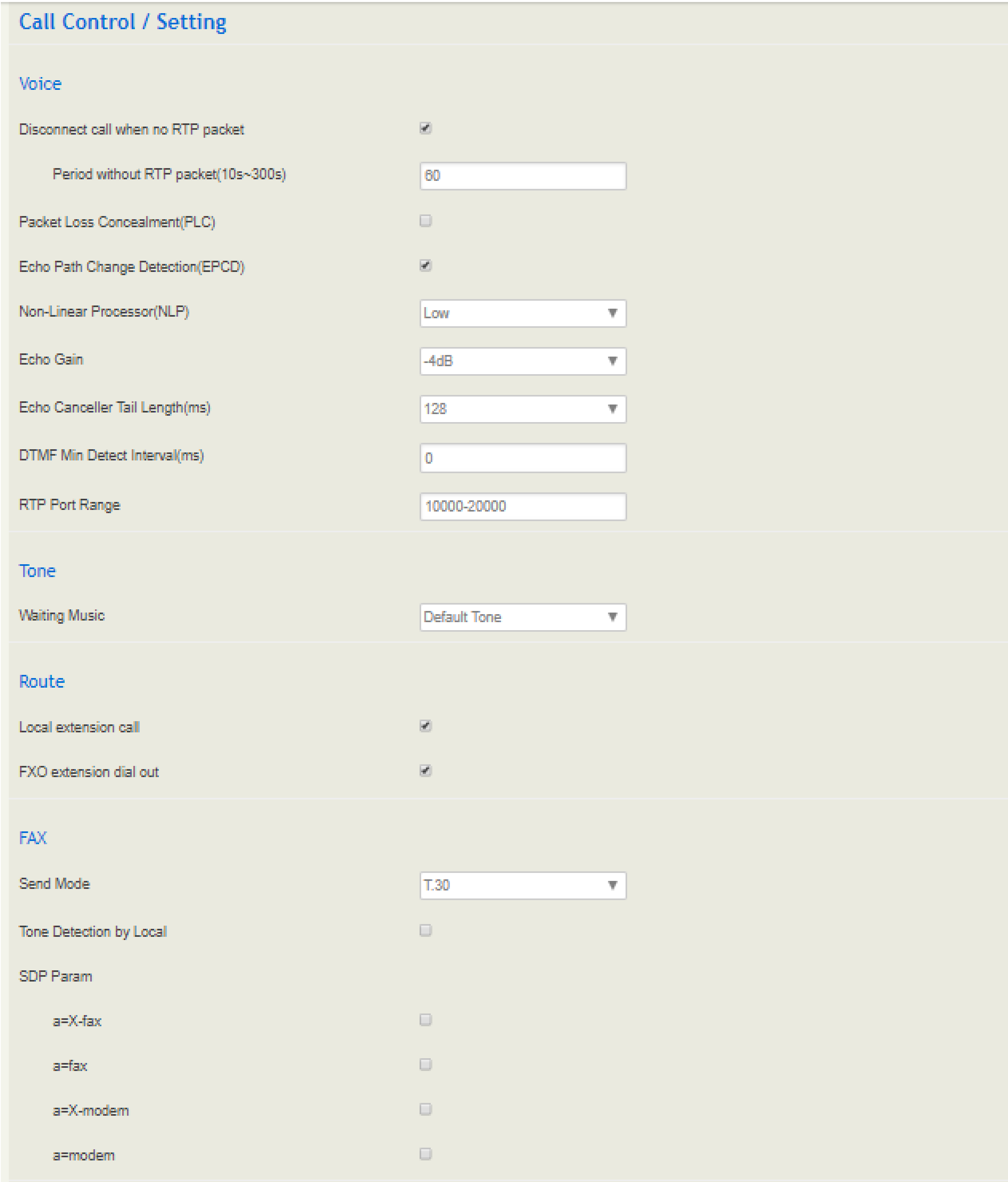
Click the **Detection** button, and the XonTel Plus PBX will automatically detect the most-matched cadence.



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.

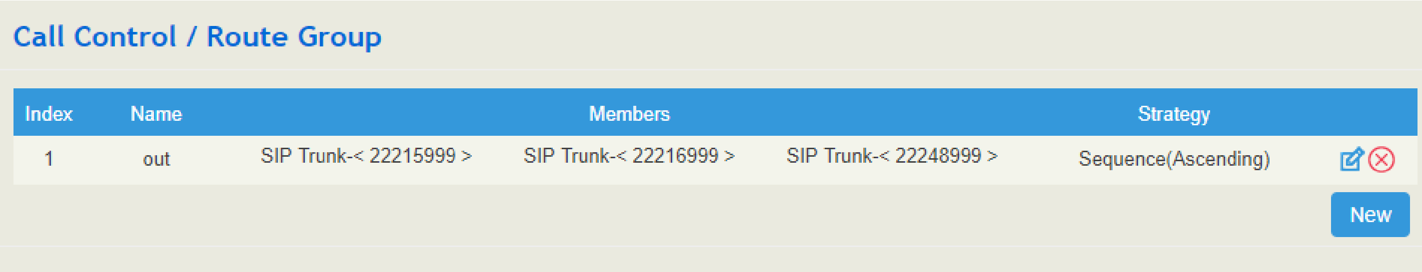
5.8.1 **Setting**

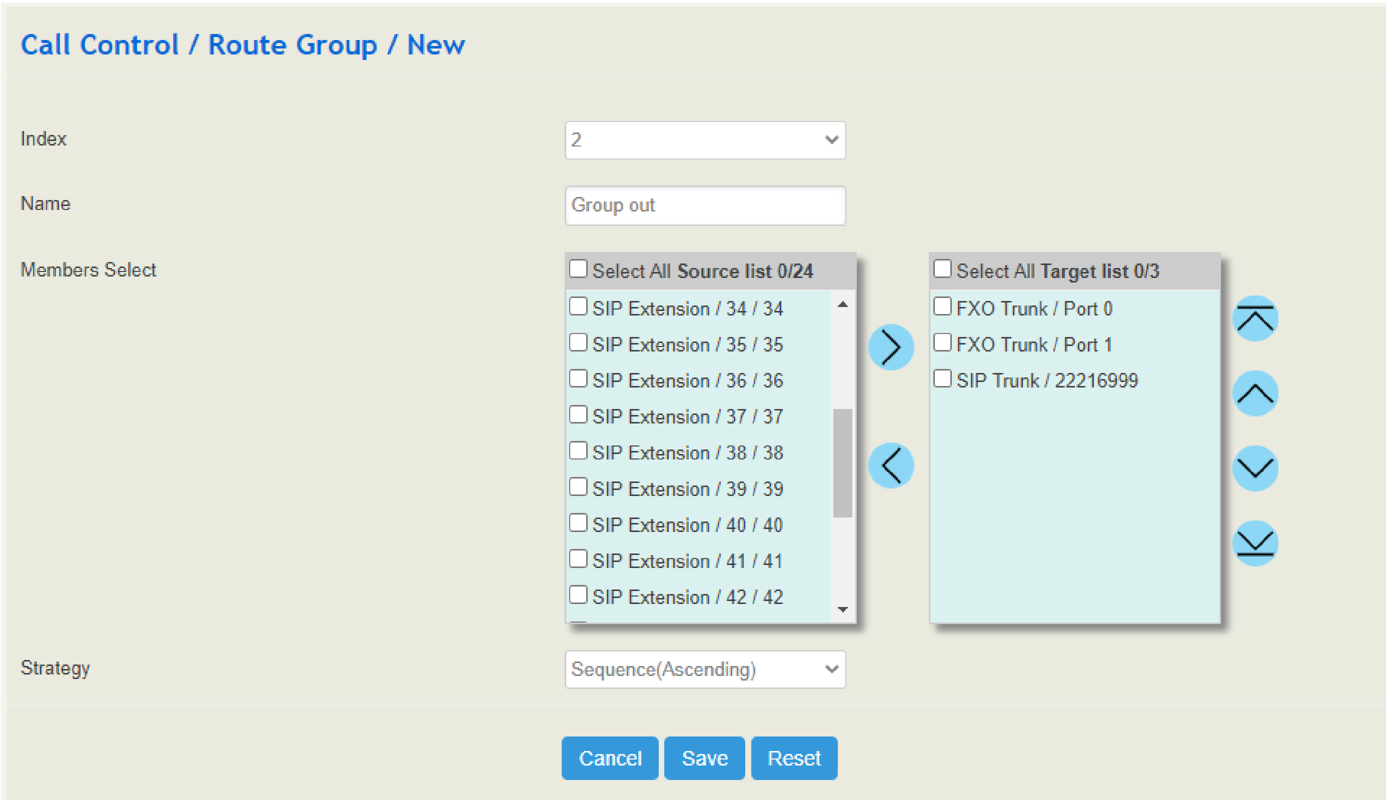


Disconnect call when no RTP packet	If it is enabled, and no RTP packets are received in PBX, calls will be disconnected.
Period without RTP packet(10s~300s)	If no RTP packets are received within the present time, calls will be disconnected.
Packet Loss Concealment(PLC)	Whether to enable the ‘Packet Loss 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 →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.

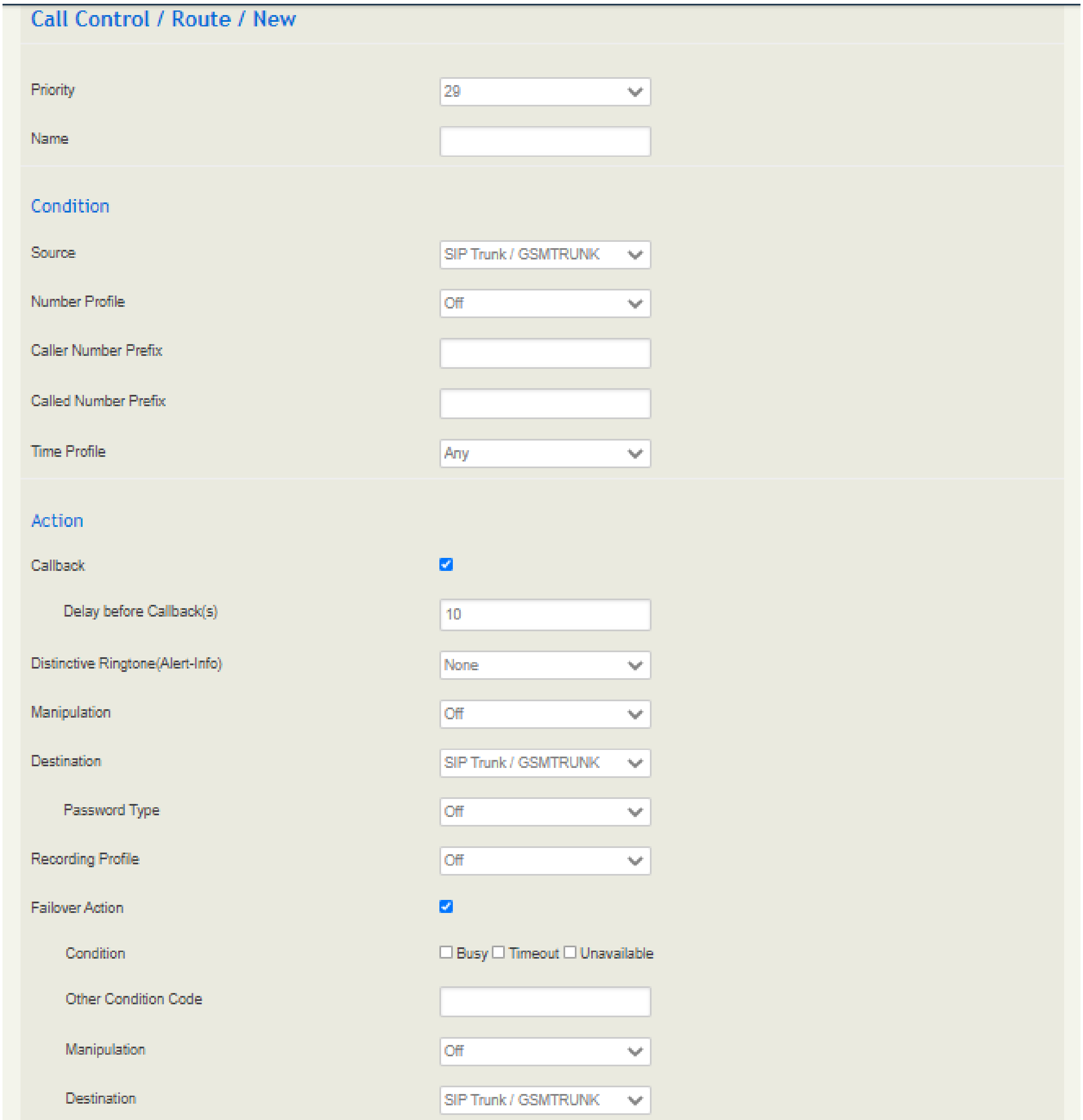




Name	The name of the route group.
Members Select	Select SIP extension(s), SIP trunk or FXO trunks.
Strategy	The strategies for choosing which route under the route group as the destination route, including Sequence (Ascending), Sequence (Cyclic Ascending), Simultaneous and Random

5.8.3 Route

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



Priority	The priority for choosing the route; the higher value, the lower priority.
Name	The name of the route.
Condition	The condition under which the route will be used
Source	The source of the call; it can be SIP extension, FXO trunk, SIP trunk a customized source or any.
Number Profile	The profile of the caller number and the called number; please make reference to the Profile → 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’.
Caller Number Prefix	The prefix of caller number; it supports regular expression.
Called Number Prefix	The prefix of called number; it supports regular expression.
Time Profile	The profile of time during which the route can be used; make reference to the **Profile → Time section**
Action	Include manipulating number and sending call to destination.
Callback	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. **Delay before Callback(s):** Set the number of seconds before PBX calling back a caller.
Distinctive Ringtone(Alert-Info)	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.**
Manipulation	If it is on, the caller number or called number of the route will be manipulated; make reference to the **Profile→ Manipulation** section.
Destination	The destination of the route.
Password Type	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.
Recording Profile	Choose Off or a recording profile.
Failover Action	The processing when a call through this route fails.

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: On — Save

Index	Feature	Key	Description	Status
1	Inquiry LAN IP	*158	Inquiry LAN IP	Enabled
2	Inquiry WAN IP	*159	Inquiry WAN IP	Enabled
3	Inquiry Phone Number	*114	Inquiry Phone Number	Enabled
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
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
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
10	Call Waiting Activate	*70	Enable Call Waiting service	Enabled
11	Call Waiting Deactivate	*71	Disable Call Waiting service	Enabled
12	Blind Transfer	*3	Example:*38000#,you can blind transfer to the extension number 8...	Enabled
13	Attended Transfer	*2	Example:*28000#,you can attended transfer to the extension numb...	Enabled
14	Call Forwarding Uncondition Activate	*72	Enable Call Forwarding Uncondition service.Example:*728000,set ...	Enabled
15	Call Forwarding Uncondition Deactivate	*073	Disable Call Forwarding Uncondition service	Enabled
16	Call Forwarding Busy Activate	*90	Enable Call Forwarding Busy service.Example:*908000,set the call...	Enabled
17	Call Forwarding Busy Deactivate	*91	Disable Call Forwarding Busy service	Enabled
18	Call Forwarding No Reply Activate	*52	Enable Call Forwarding No Reply service.Example:*528000,set th...	Enabled
19	Call Forwarding No Reply Deactivate	*53	Disable Call Forwarding No Reply service	Enabled
20	DND Activate	*78	Enable Do Not Disturb service	Enabled
21	DND Deactivate	*79	Disable Do Not Disturb service	Enabled
22	Call Pickup	*4	Pick up the ringing extension, Example:*48000, pick up the extensi...	Enabled
23	WAN Access Control	*160*	*160*1# - Allow HTTP WAN access, *160*0# - Deny HTTP WAN a...	Enabled
24	Voicemail Service	*98	*981# - Leave messages, *982# - Play messages	Enabled
25	Callback Service	*163	Callback the last received call	Enabled
26	Recording Service	*1	Start or stop recording when manual recording	Enabled
27	Call Park	*6	Example: *6, you can park another part during the call. *6100, you ...	Enabled
28	Call Monitor	*22	*222 - Listen Mode, *223 - Whisper Mode, *224 - Barge-in Mode. E...	Enabled
29	Auto Answer	*80	Make an intercom with a specific extension user, Example: dial *80...	Enabled

- To a disable a specific feature code, click the button 
- To a enable a specific feature code, click the button 

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

***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:

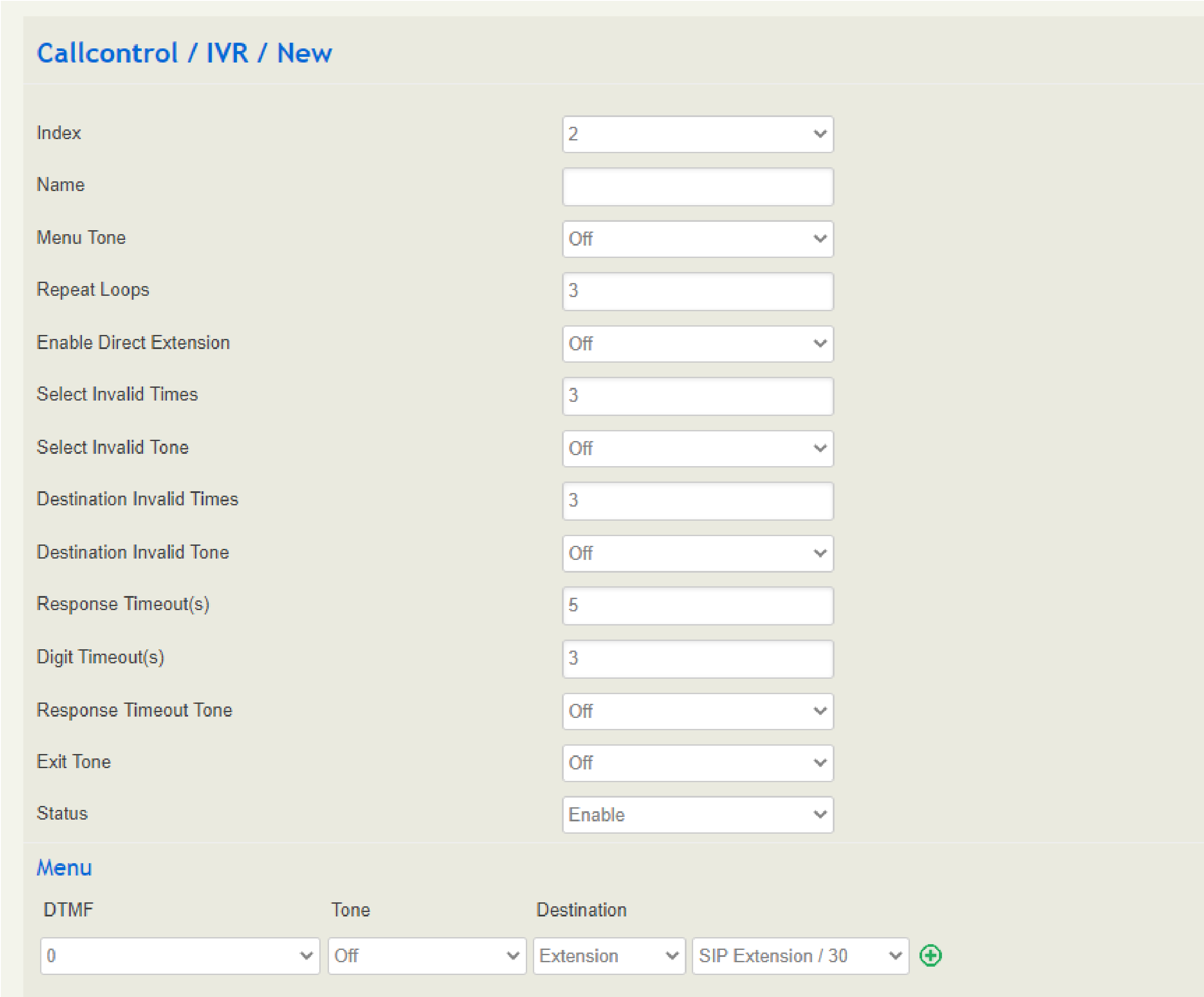
1. **A voice prompt indicating successful configuration will be given after each configuration procedure. Please do not hang up until hearing this voice prompt.**
2. **You can edit and customize your feature codes as shown below.**

Call Control / Feature Code / Edit

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

5.8.5 IVR

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



Name	The name of the IVR
Menu Tone	Choose Off or a voice prompt that you uploaded in **System → 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 “ **Enable Direct Extension** “ option is enabled.

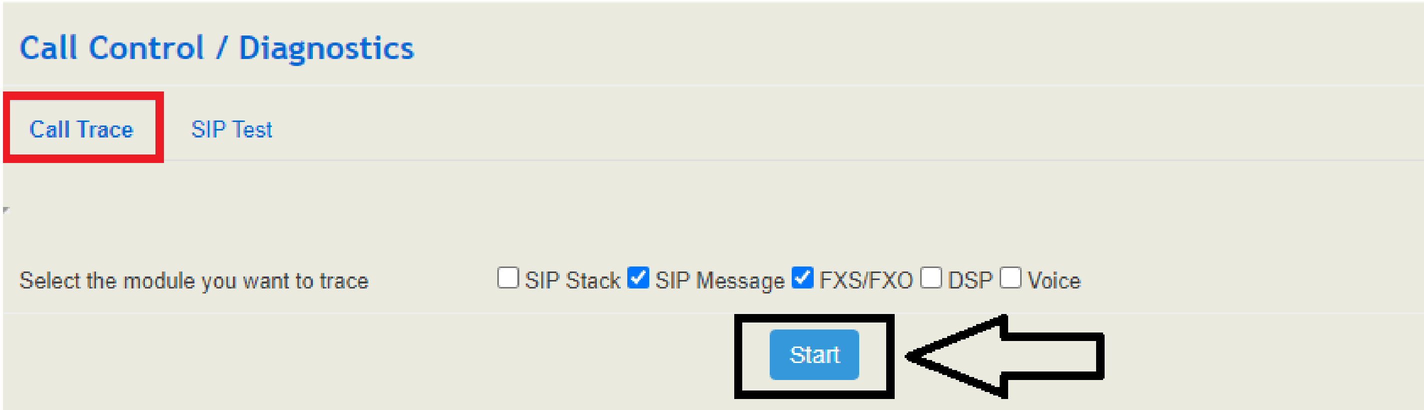
Select Invalid Tone	Prompt to be played when an invalid/unmatched response is received from the caller. Please note that this option will be ignored when " **Enable Direct Extension** " option is enabled.
Destination Invalid Times	The number of times to retry when receiving an invalid/unmatched extension number from the calling side if " **Enable Direct Extension** " option is enabled.
Destination Invalid Tone	Prompt to be played when an invalid/unmatched extension number is received from the calling side if " **Enable Direct Extension** " 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 PBX 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 **Timeout**.
Exit Tone	Prompt to be played for IVR exit.
Status	If it is disabled, the IVR cannot be seen in the destination of route.
Menu	DTMF: It can be 0-9 quick-dial numbers, *, #, others or timeout. 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

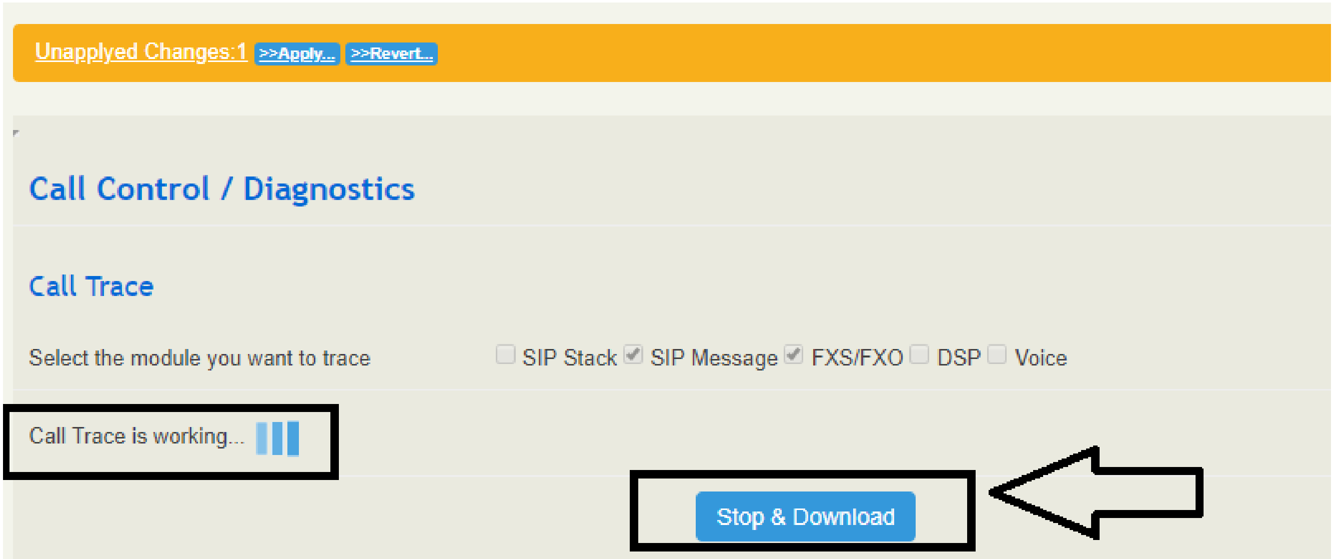
In case that call cannot be connected or voice has quality problem, you can enter into the **Call Control →Diagnostics → Call Trace** to collect fault-related information and then send it to technical support to locate fault.

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.



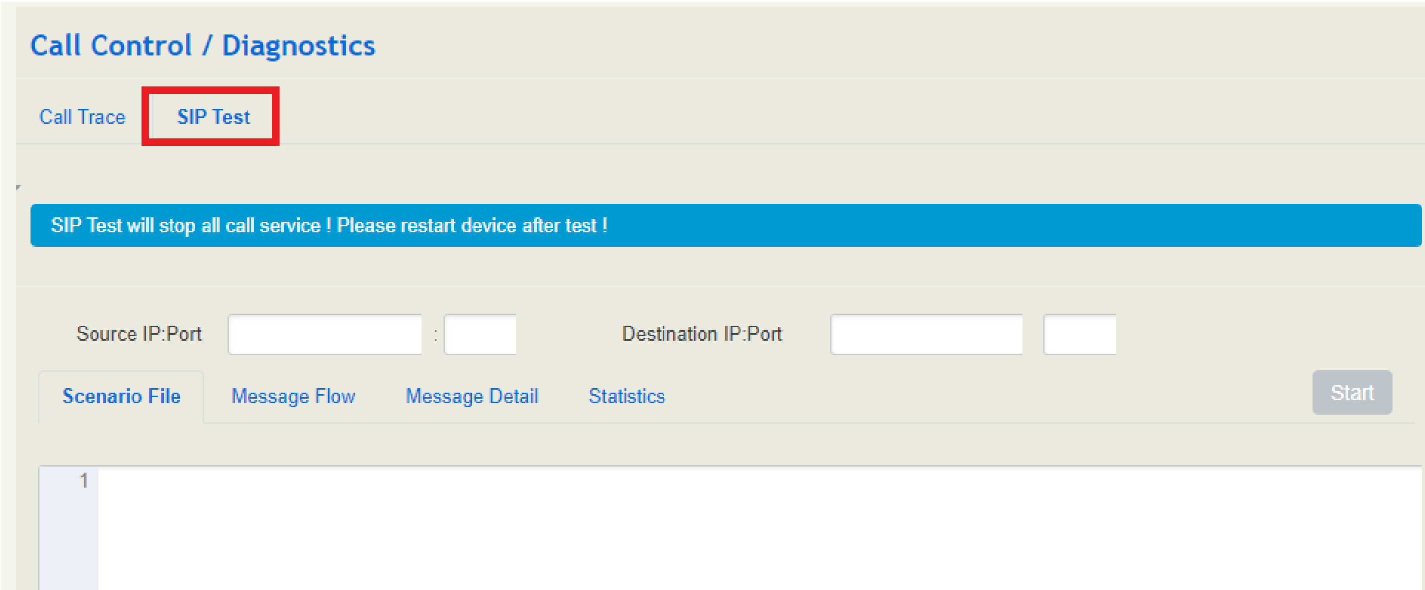
2. Give a call, and come back to the **Call Control →Diagnostics** interface after the call ends then click **Stop & Download** to stop the call trace and download the tracing file.



3. In order to locate faults more quickly, you sometimes need to enter into the **System →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 →Diagnostics → 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.



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
TCP	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