PLANET
Networking & Communication

User's Manual

Internet Telephony PBX System

► IPX-2200

► IPX-2500

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CE Mark Warning

This is a class B device. In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

Energy Saving Note of the Device

This power required device does not support Standby mode operation. For energy saving, please remove the DC-plug or push the hardware Power Switch to OFF position to disconnect the device from the power circuit.

Without removing the DC-plug or switching off the device, the device will still consume power from the power circuit. In view of Saving the Energy and reducing the unnecessary power consumption, it is strongly suggested to switch off or remove the DC-plug from the device if this device is not intended to be active.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

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Revision

User’s Manual of PLANET Internet Telephony PBX System

Model: IPX-2200/IPX-2500

Rev: 1.0 (Jul., 2016)

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Chapter 1. Introduction

Intuitive, Ease-of-Use IP PBX Machine Management

PLANET IPX-2200/IPX-2500 IP PBX telephony system is SIP-based for optimizing communications among the small and medium businesses. The IPX-2200 and IPX-2500 are able to accept 200/500 user registrations, and easy to manage a full voice over IP system with the convenience and cost advantages.

Off-net Calling Capability, Call Restriction, Call Access Control

The IPX-2200/IPX-2500 integrates **up to 8 calls** via the IPX-21FO (4 FXO) and IPX-21GS (4 GSM) modules to form a feature-rich PBX system that supports seamless communications between the existing PSTN calls, analog, IP phones and SIP-based endpoints.

Replacing Old PBX Easily without New Wiring

Cost-effective, easy-to-install and simple-to-use, the IPX-2200/IPX-2500 converts standard telephones to IP-based networks. It enables the service providers and enterprises to offer users traditional and enhanced telephony communication services via the existing broadband connection to the Internet or corporation network.

With the IPX-2200/IPX-2500, home users and companies are able to save the installation cost and extend their past investments in telephones, conferences and speakerphones. The IPX-2200/IPX-2500 can be the bridge between traditional analog systems and IP network with an extremely affordable investment.

Distributed VoIP Network Infrastructure

For the new-generation communication age, the IPX-2200/IPX-2500 supports IPv6 and VPN (client/server) connection to provide users with more flexible and advantageous communications products. With PLANET DDNS function, the IPX-2200/IPX-2500 also helps users to apply and remember the login information easier. Moreover, its multiple language feature helps user to quickly and friendly manage the system. The IPX-2200/IPX-2500 supports Lync server to which smart phone (using third-party app) and analog phone are connected via its communication with other devices of Lync server.

Standard Compliance

Compliant with the Session Initiation Protocol 2.0 (RFC 3261), the IPX-2200/IPX-2500 are able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

Green IP Office

Virtual fax functionality on IPX-2200/IPX-2500 system allow faxes to be sent and received without requiring a fax machine. This useful feature will allow businesses to demonstrate their green credentials while at the same time reduce fax related costs across the enterprise. Inbound faxes can be automatically received and converted to TIF files and saved in the IPX-2200/IPX-2500 system. It is also possible to configure the IPX-2200/IPX-2500 system to send the TIF files to a user's email box. Sending outbound faxes is as easy as uploading a file from the extension user web portal, thus creating a paperless or green office.

Full Security with VPN Support

The IPX-2200/IPX-2500 VPN securely and cost-effectively connects geographically disparate offices of an organization, creating one cohesive virtual network. The IPX-2200/IPX-2500 VPN technology is also used by ordinary Internet users to connect to proxy servers for the purpose of protecting one's identity. They include VPN server and client function that can support users full security login.

1.1 Features

➢ **System Highlights**

- 60 concurrent calls and up to 200 registers (For IPX-2200)
- 100 concurrent calls and up to 500 registers (For IPX-2500)
- HD voice codec G.722 for perfect voice quality
- Virtual Fax for green office
- Voicemail to Email for not missing any important message
- Paging and intercom function strengthens work efficiency.
- Built-in SIP Proxy Server following RFC 3261
- Multiple Languages of GUI for international business
- Web-based Control Panel for easy configuration and management of the system.
- Hardware Echo Cancellation module for great and smooth communication.
- Strong security features protect your system from hacking.
- Supports maximum 8 ports for FXO/GSM (on 2 slots)
- Records voice and voicemail to external USB disk
- Supports Lync server

➢ **Codec and Protocol**

- SIP 2.0 (RFC 3261), IAX2 compliant
- Audio Codec: G.722/G.711-Ulaw/G.711-Alaw/G.726/G.729/GSM/SPEEX
- Video Codec: H.261/H.263/H.263+/H.264
- DTMF: RFC 2833, SIP info, in-band

➢ **Network and Security Features**

- DDNS Client (PLANET DDNS, Dyndns.org, No-ip.com, zoneedit.com, freedns.afraid.org, www.oray.com, 3322.org)
- DHCP Server/SNMP v1/v2
- IEEE 802.1Q of VLAN
- IPv4/IPv6, SIP over IPv6
- Manual Configuration of Static Route Table
- Troubleshooting (Ping, Traceroute)
- VPN Server (L2TP/PPTP/OpenVPN/IPSec, up to 20 connections for VPN clients)
- VPN Client (L2TP/PPTP/OpenVPN/N2N/IPSec)
- Refuse SIP Register DoS
- Refuse Abort Invite Dos

- Refuse SSH Login DoS
- Firewall/SRTP
- Enhances HTTPS connection

PBX Features

- Auto-Provision (PLANET/Cisco IP Phone)
- Black List
- BLF (Busy Lamp Field), Speed Dial
- CDR (Call Detailed Record) (20000 records)
- Conference Room (36 rooms)
- Call Queue Record, Ring Group Record
- DoD (Direct Outward Dialing) and DID (Direct Inward Dialing) numbers
- DISA (Direct Inward System Access)
- DND (Do Not Disturb)
- Feature Codes, Flash Operation Panel
- Flexible Dial Plan, Follow Me
- IVR (Interactive Voice Responses)
- LDAP Server for phonebook
- Multi-language System Prompt
- Multiple Languages of GUI
- One Number Stations
- Phone Book/PIN Set
- Phonebook/LDAP (5000 contacts)
- Record Files Download
- Ring Group, SIP Trunk
- Skype for SIP/Smart DID/System Log/System Backup
- T.38 fax (pass-through)/time-based rule
- Virtual Fax/Voicemail & Voicemail to Email
- WebRTC

Call Features

- Attend Transfer, Call Waiting
- Call Back, Call Forward, Call Group
- Call Hold, Call Paging and Intercom
- Call Park, Call Pickup, Callback
- Call Center Queues (36)
- Call Record, Call Route, Blind Transfer

- Caller ID, Dial by Name
- Customized IVR, On-hold Music, Transfer
- Three-way Conferencing, Video Call

1.2 Package Contents

Thank you for purchasing PLANET Internet Telephony PBX system, IPX-2200 and IPX-2500. Open the box of the Internet Telephony PBX system and carefully unpack it. The box should contain the following items:

- Internet Telephony PBX system unit x 1
- Quick Installation Guide x 1
- User's Manual CD x 1
- Power Cord x 1
- RJ45 x 1
- Bracket x 2

If any of the above items are damaged or missing, please contact your dealer immediately.

1.2.1 Physical Specifications of IPX-2200

Dimensions

Dimensions (W x D x H)	343 x 154 x 35 mm
Weight	1.4 kg (gross weight), 1.8 kg (with package)

Front Panel

Rear Panel

LED definitions

Front Panel LED	Status	Description
PWR	Steady Green Off	PBX Power ON PBX Power OFF
SYS	Blinking Green On Off	System is working System doesn't boot System failure

Front Panel LED	Status	Description
WAN	Blinking Green On Off	Data transfer PBX network connection is established Waiting for network connection
LAN	Blinking Green On Off	Data transfer PBX network connection is established Waiting for network connection
FXO	Steady Red Flashing Off	Ready/Standby Ringing Module not available

Physical interfaces description

1	**Power Switch**	Switch the power on or off
2	**Power Cord**	AC 100~240V, 50/60Hz, 1.5A max
3	**WAN/LAN**	The WAN/LAN port supports auto negotiating Fast Ethernet 10/100BASE-TX networks. The WAN port allows your IP PBX to be connected to an Internet Access device, e.g., router, cable modem or ADSL modem through a Cat5 twisted-pair Ethernet cable.
4	**HDMI Port**	For video output (factory use)
5	**USB**	For external store device to store voice and voicemail
6	**Audio In/Out**	For external paging
7	**Module Slot 1/Slot 2**	2 external slots with compliant FXO/FXS/GSM module -**FXO module** is connected to PBX or CO line with RJ11 analog line. FXO port is connected to the extension port of a PBX or directly connected to a PSTN line of carrier -**GSM module** is connected to Global System for Mobile Communications (GSM) with SIM card

Note

Supporting 2 slots, user can buy expansion module like IPX-21FO (4FXO) or IPX-21GS (4GSM) for extending port service.

1.2.2 Physical Specifications of IPX-2500

Dimensions

Dimensions (W x D x H)	343 x 154 x 35 mm
Net Weight	1.4 kg (gross weight), 1.8 kg (with package)

Front Panel

Rear Panel

LED definitions

Front Panel LED	Status	Description
PWR	Steady Green Off	PBX Power ON PBX Power OFF
SYS	Blinking Green On Off	System is working System doesn't boot System failure
WAN	Blinking Green On Off	Data transfer PBX network connection is established Waiting for network connection
LAN	Blinking Green On Off	Data transfer PBX network connection is established Waiting for network connection
FXO	Steady Red Flashing Off	Ready/Standby Ringing Module not available
GSM	Steady Red Flashing Off	Ready/Standby (SIM card inserted) Ringing No SIM card inserted
FXS	Steady Green Flashing Off	Ready/Standby Ringing Module not available

Physical interfaces description

1	**Power Switch**	Switch the power on or off
2	**Power Cord**	AC 100~240V, 50/60Hz, 1.5A max
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Note	Supporting 2 slots, user can buy expansion module like IPX-21FO (4FXO) or IPX-21GS (4GSM) for extending port service.

1.3 Specifications

Product	IPX-2200 Internet Telephony PBX system (200 SIP Users registrations)	IPX-2500 Internet Telephony PBX system (500 SIP Users registrations)
Hardware Specifications		
WAN	1 x 100BASE-TX RJ45 for WAN, connecting to broadband modem or a WAN router	
LAN	1 x 100BASE-TX RJ45 for LAN, connecting to a LAN switch	
HDMI Port	For video output (factory use)	
USB	For external store device to store voice and voicemail	
Audio In/Out	For external paging	
2 Slots	Supports maximum 8 ports (FXO/GSM)	
USB	Store data for external disk	
LED Indications	PWR: 1, LNK/Off SYS: 1, LNK/Off WAN: 1, LNK/Off LAN: 1, LNK/Off SLOT: 2, FXO/GSM (Red), FXS (Green)	
Dimensions (W x D x H)	343 x 154 x 35 mm	
Power Requirements	100 - 240 VAC	100V - 240 VAC
EMC/EMI	CE, FCC Class B, RoHS	
Protocols and Standard		
Standard	SIP 2.0 (RFC3261), IAX2	
Protocols	RFC 793 TCP RFC 826 ARP RFC 1034, 1035 DNS RFC 2068 HTTP RFC 2131 DHCP RFC 2516 PPPoE RFC 3261, RFC 3311, RFC 3515 RFC 3265, RFC 3892, RFC 3361 RFC 3842, RFC 3389, RFC 3489 RFC 3428, RFC 2327, RFC 2833 RFC 2976, RFC 3263	
Voice Codec	G.722, G.711-Ulaw, G.711-Alaw, G.726, G.729, GSM, SPEEX	
Video Codec	H.261, H.263, H.263+, H.264	
Fax over IP	T.38 Fax (pass-through) Note: T.38 support is dependent on fax machine, SIP provider and network/transport resilience.	
Voice Processing	DTMF detection and generation In-band and RFC 2833, SIP info	

Protocols	SIP 2.0 (RFC-3261), TCP/IP, UDP/RTP/RTCP, HTTP/HTTPS, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE	
System Capacity		
System Capacity	• 60 concurrent call legs • Up to 200 IP phone registers/extensions • Recording and Voicemail (GSM/default): 1500 hours Wav: 150 hours	• 100 concurrent call legs • Up to 500 IP phone registers/extensions • Recording and Voicemail (GSM/default): 75000 hours Wav: 7500 hours
Network and Configuration		
Access Mode	Static IP, PPPoE, DHCP	
Environment		
Operating Environment	0~40 degrees C 5~95% humidity	

Chapter 2. Installation Procedure

2.1 Web Login

Step 1. Connect a computer to a LAN port on the IPX-2200 or IPX-2500. Your PC must be set up to the same domain of 192.168.0.X as that of the IPX-2200 or IPX-2500.

Step 2. Start a web browser. To use the user interface, you need a PC with Internet Explorer (version 8 and higher), Firefox, or Safari (for Mac).

Step 3. Enter the default IP address of the IPX-2200 or IPX-2500: https://192.168.0.1 in the URL address box.

Step 4. Enter the default user name **admin** and the default password **admin**, and then click Login to enter Web-based user interface.

(Default IP)

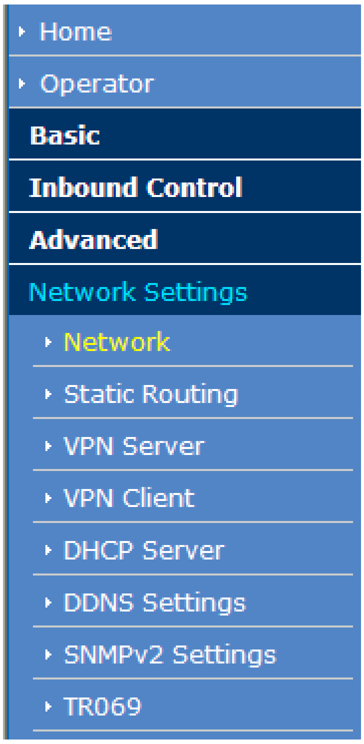
Default LAN IP: https://**192.168.0.1**
Default WAN IP: https://**172.16.0.1**
Default User Name: **admin**
Default Password: **admin**

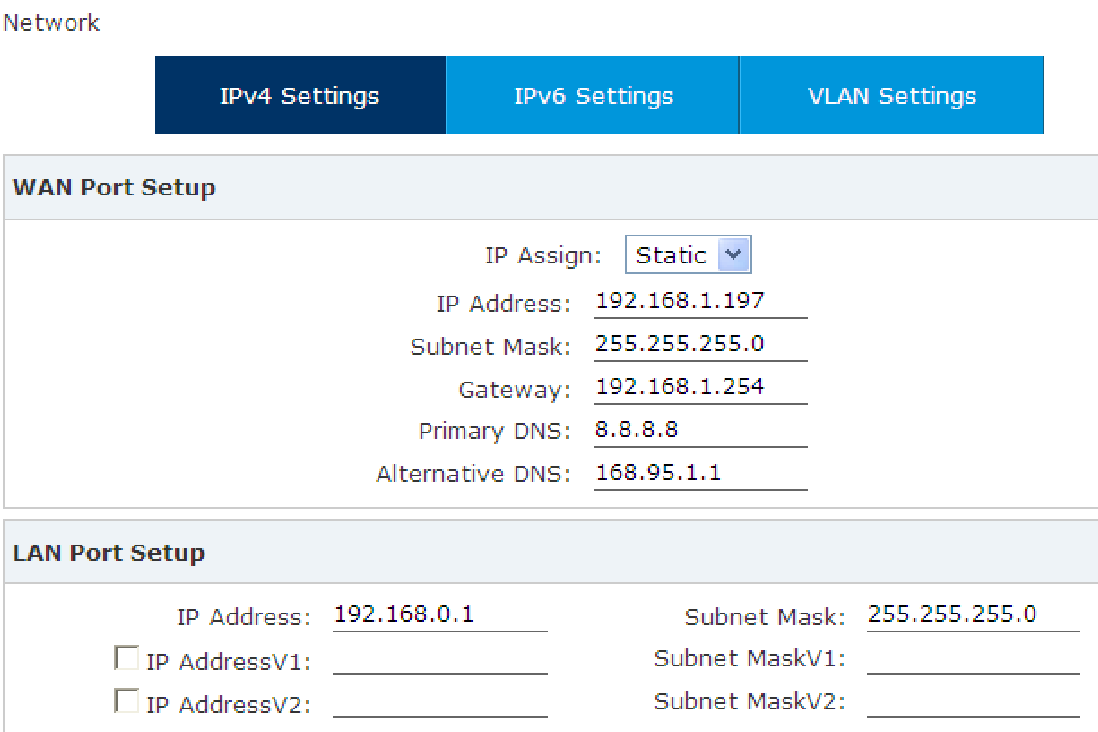
Figure 2-1. Login page of the IPX-2200/IPX-2500

Note	For security reason, please change and memorize the new password after this first setup.

2.2 Configuring the Network Setting

Step 1. Go to Network Settings → **Network**





Step 2. Edit your WAN port IP information.

There are three types of Ethernet port connection. They are **Static IP**, **DHCP** and **PPPoE** (Point-to-Point Protocol over Ethernet)**.** You can find detailed setting process in the user manual.

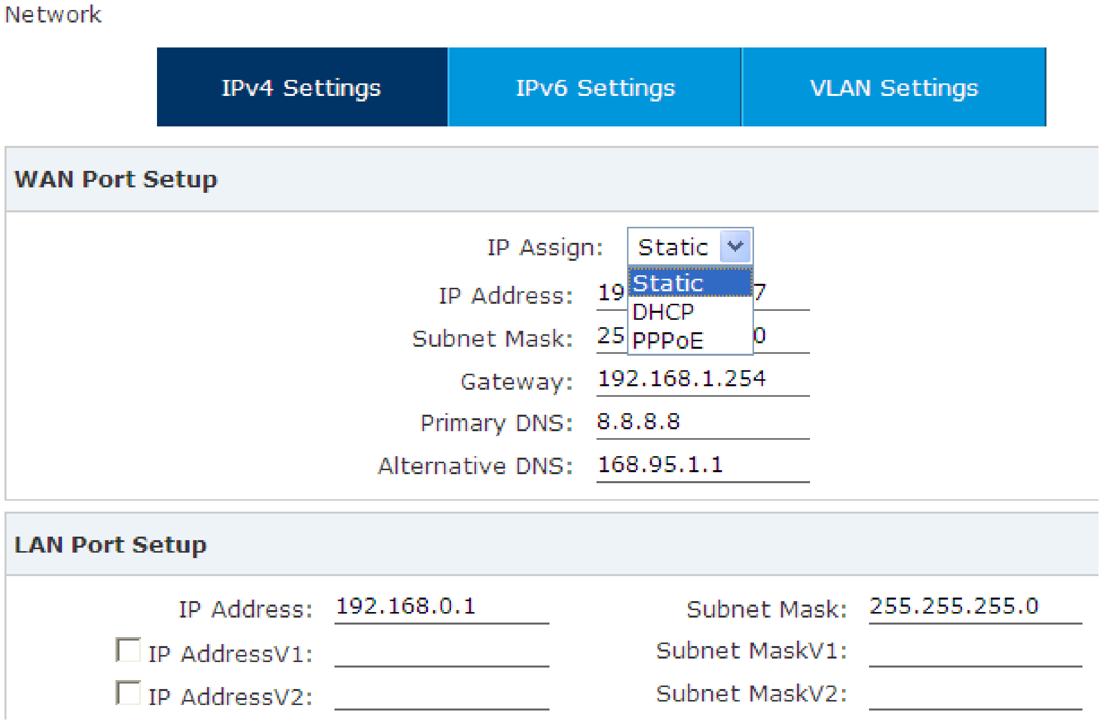


Figure 2-4. Selection of IP Connection Type

Chapter 3. Basic Configuration

3.1 Preparation Before Operation

What kind of IP phone can be used with the IP PBX IPX-2200 and IPX-2500?

- Our IPX-2200 and IPX-2500 is based on SIP 2.0 (RFC 3261); any IP phone model based on the same protocol can work with the IPX-2200 and IPX-2500.

3.2 Before Making a Call

3.2.1 System Information

Default LAN IP: https://**192.168.0.1**

Default WAN IP: https://**172.16.0.1**

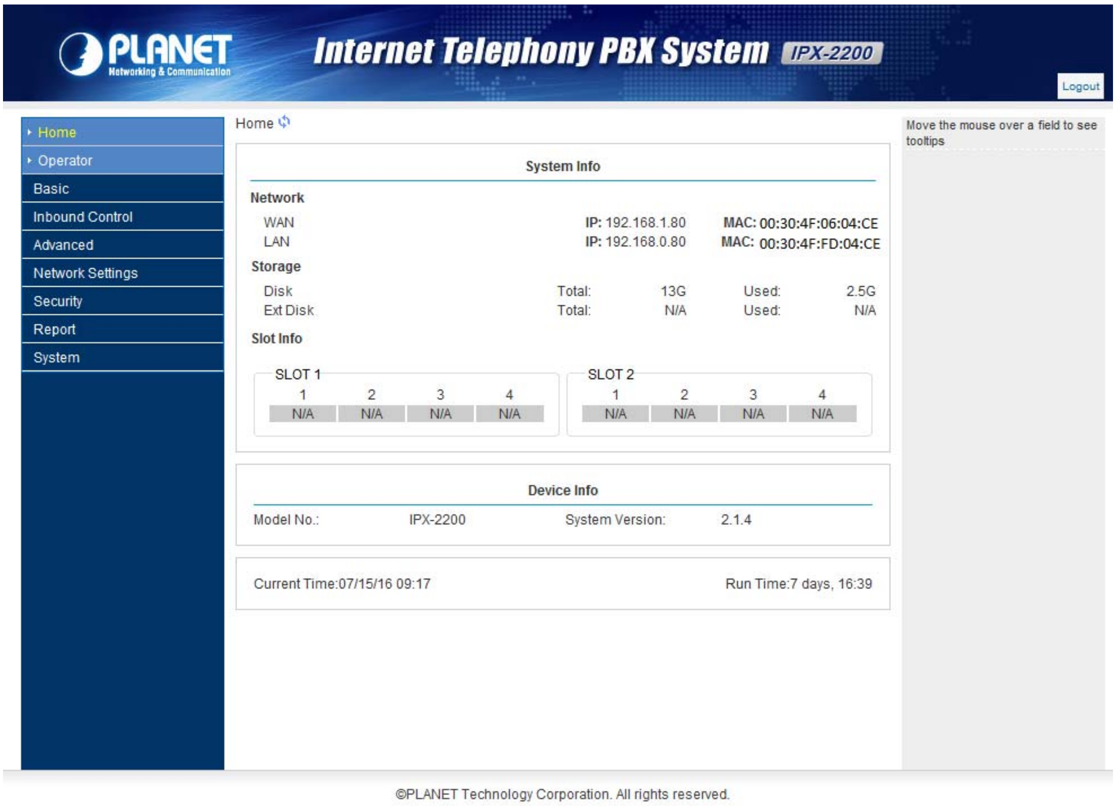
Default User Name: **admin**

Default Password: **admin**

Note	1. To login to the IPX-2200 or IPX-2500, your PC must use the same domain as the LAN IP address of the IPX-2200 or IPX-2500. 2. For security reason, please modify the user name and password after you login. You can modify it on this page: **"System"---"Management"**

	3. Every time after saving the change, please press **"Activate Changes"** to make modification effective.

If user name and password are right, this following page will be displayed:



1	**Network**	WAN/LAN IP and MAC will be displayed
2	**Storage**	Total storage and used storage will be displayed
3	**Slots Info**	Channel information will be based on the product model
4	**Device Info**	Product Model and System Version will be displayed

Note	1. If FXO is connected, the slot color and the front panel LED will be red and steady red, respectively. 2. If FXS is connected, the slot color and the front panel LED will be green and steady green, respectively.

Commonly Used Button

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

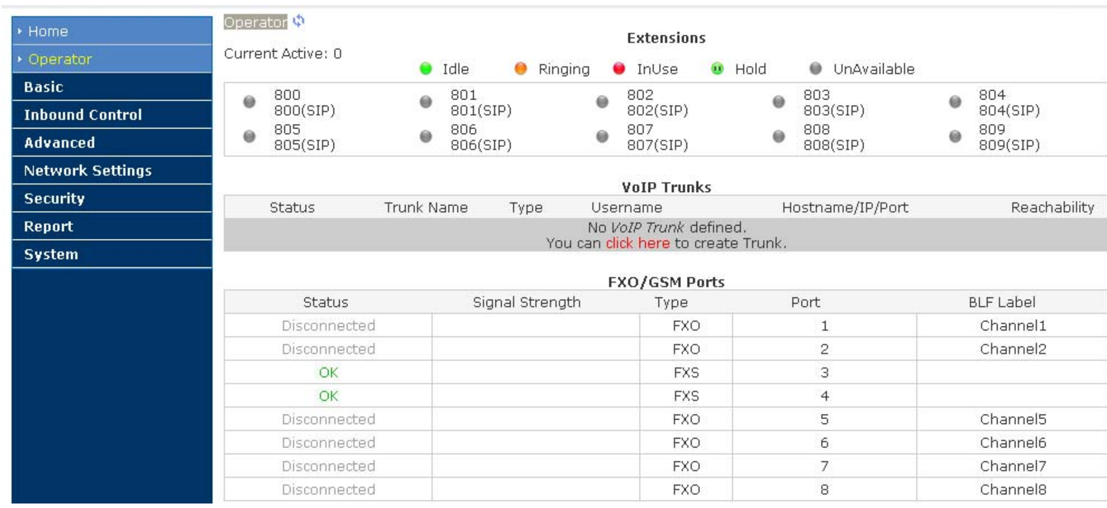
1	**Logout**	Logout the Web panel
2	**Activate Change**	Activate the changes for your current configuration

System Menu

System Menu includes the following sub menu:

1	**Home**	Display device information
2	**Operator**	Extension/Trunk/Channel Status
3	**Basic**	Basic configuration on extension, trunks, etc
4	**Inbound Control**	Configuration of Inbound Route, IVR and Black List, etc
5	**Advanced**	Configuration of extension's default information, Conference Call, Call Transfer, Function Key, etc.
6	**Network Settings**	Configuration of Routing, Network, VPN, DHCP and other related network parameters
7	**Security**	Configuration of Firewall, SSH, FTP.
8	**Report**	Record List, Call Logs and System Logs.
9	**System**	Time Settings, Management, Back Up and Upgrade, etc.

3.2.2 Operator



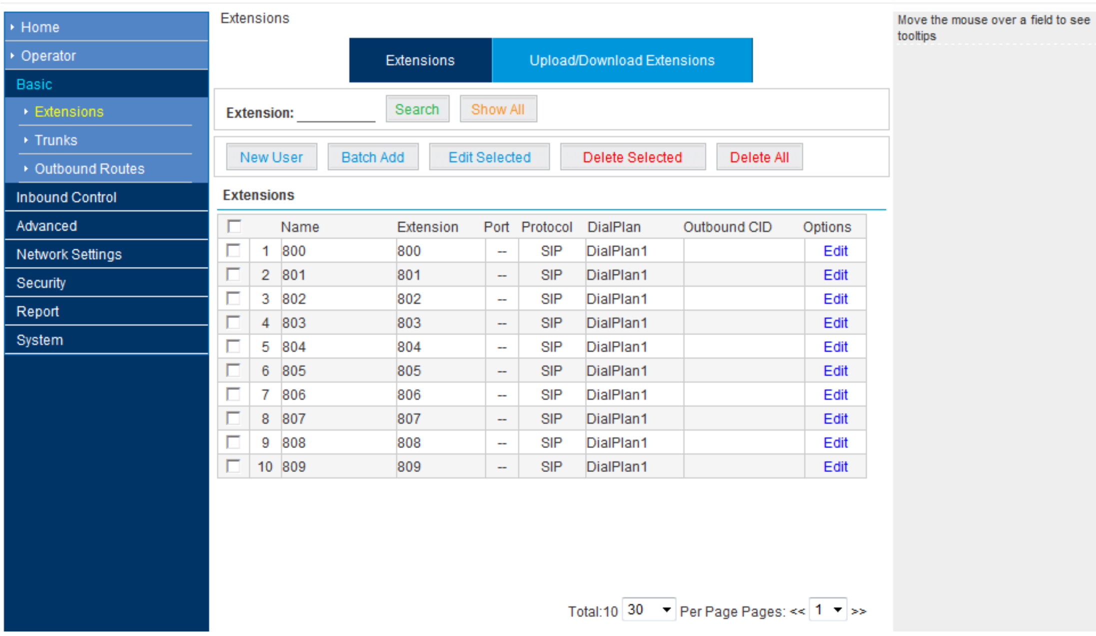
Display all the Extension, VoIP Trunk and Slot information.

About extension:

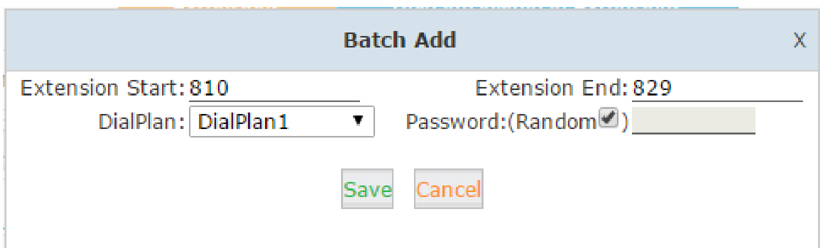
1		Idle
2		Ringing
3		In use
4		Hold
5		Unavailable

3.2.3 Basic Configuration

Add new extensions



You can add more extensions one by one by clicking the “New User” button or bulk add extensions by clicking the “Batch Add” button.



Field description

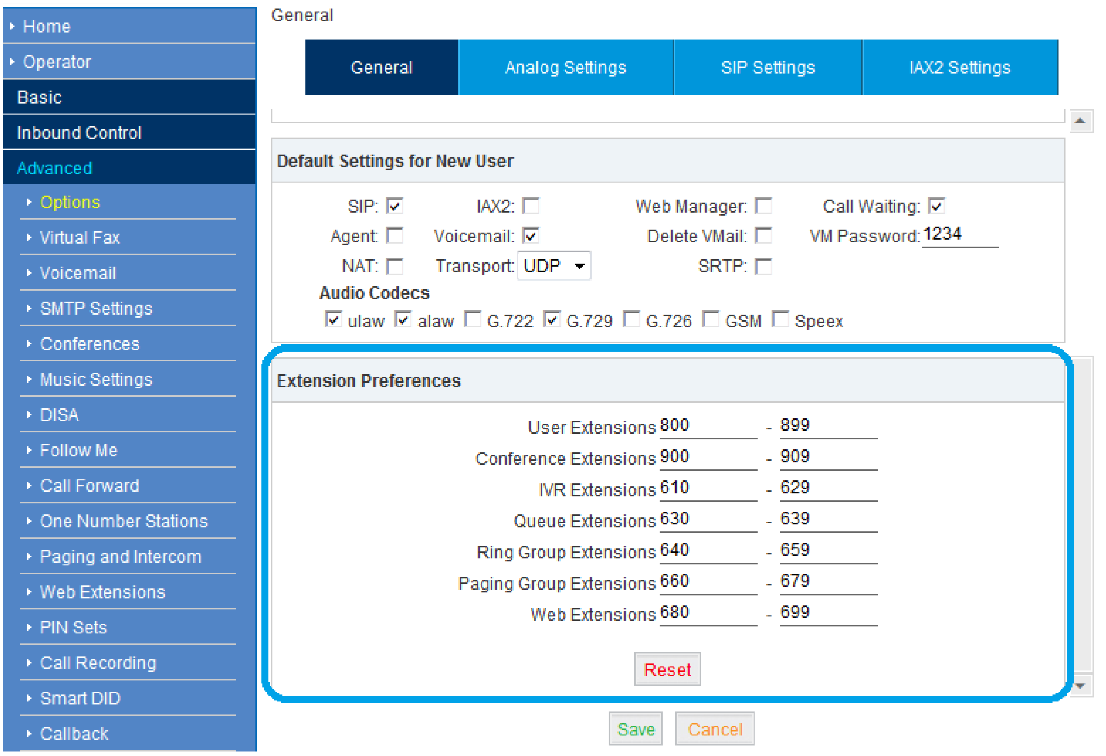
Item	Explanation
Extension Start/Extension End	These two fields define the new extension range to be generated.
Dial Plan	Select a same dial plan for these new extensions.
Password	Can be different random passwords consisting of numbers, letters and special characters (suggested) by checking the "Random" checkbox. Or you can specify the same password for all new extensions.

Other Extension Ranges

In Planet IP PBX system, we limited the user extension range within 800 and 899. If you want more extensions or you want the extensions in other ranges you need to change the extension range before you can add new extensions.

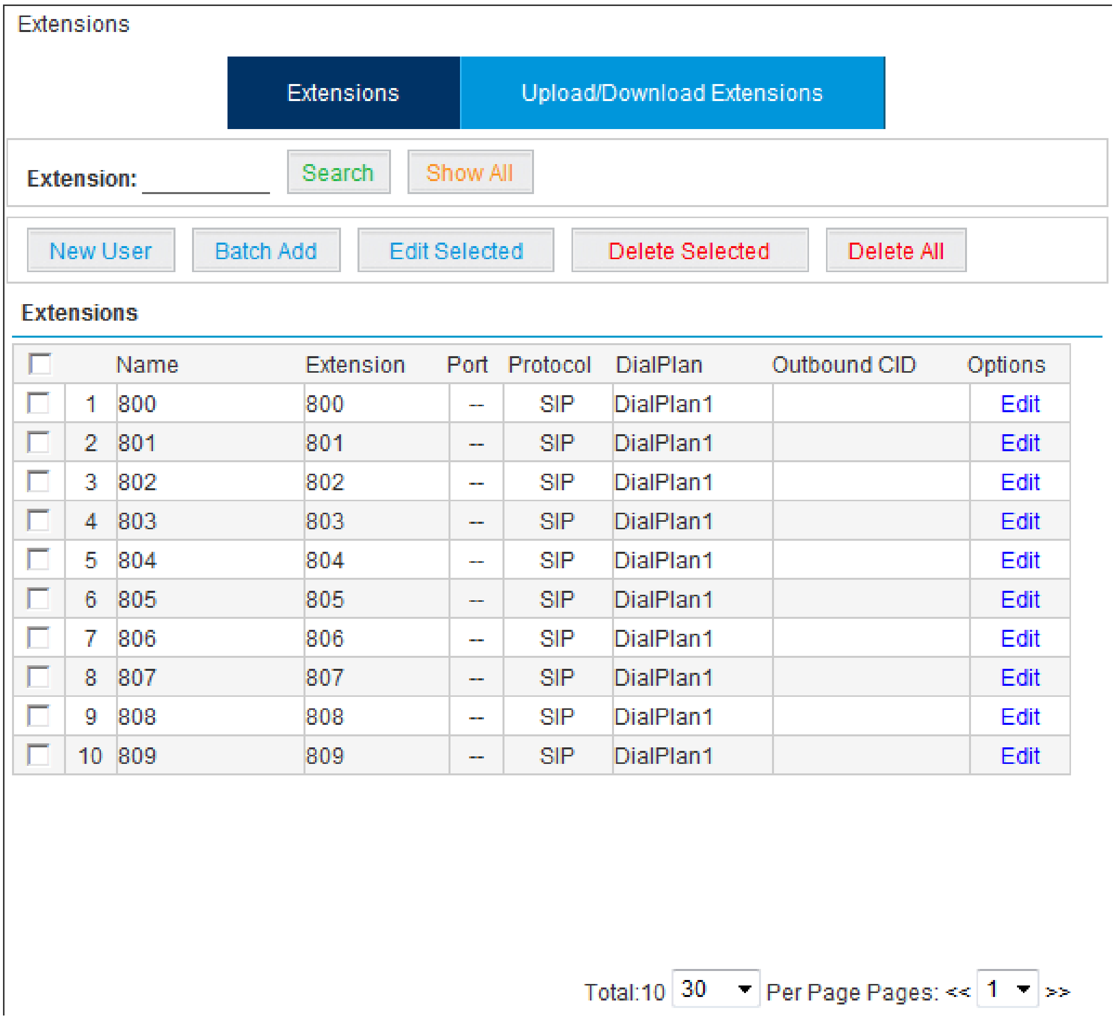
Please navigate to web menu *Advanced->Options->General.*

In the "Extension Preferences" section you can change the user extension range.



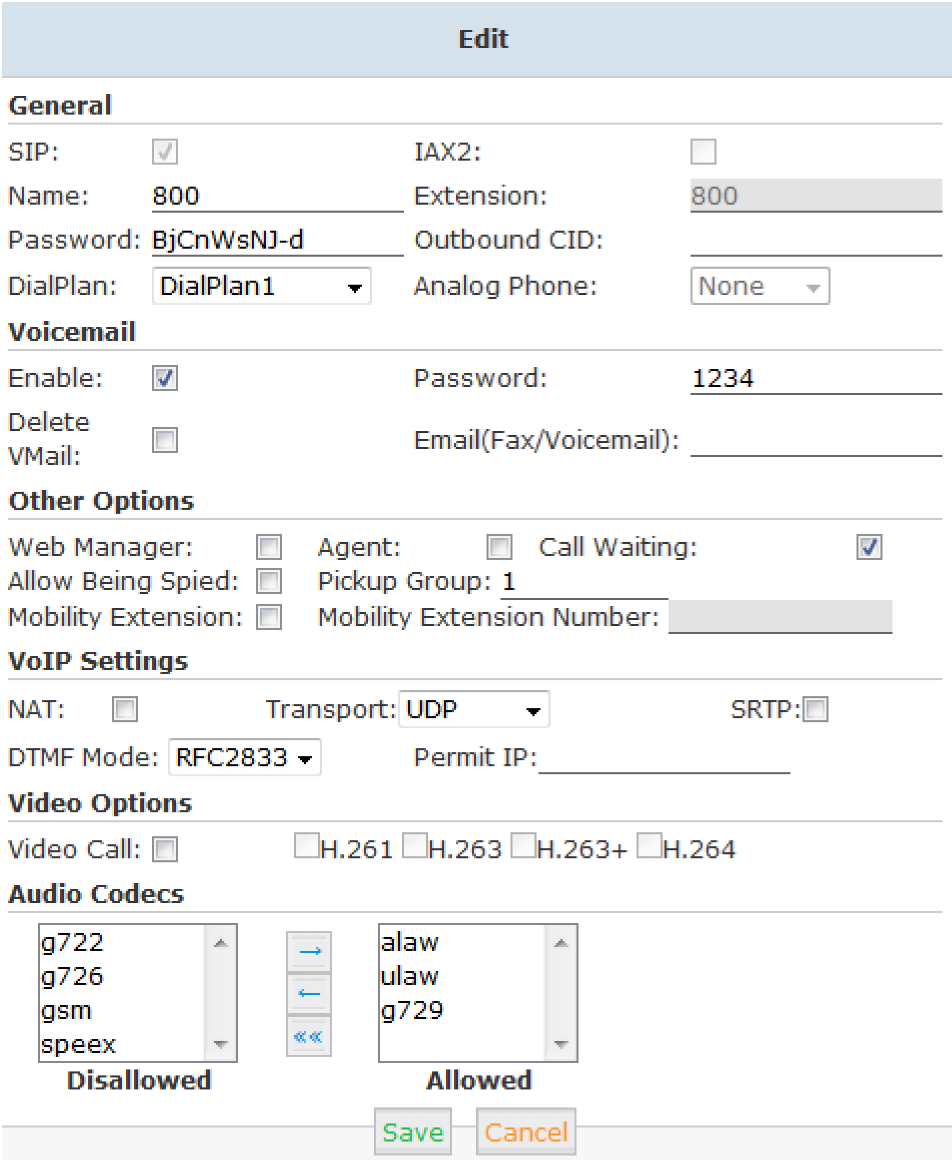
Configure Extensions

Planet IP PBX supports SIP/IAX2 and analog extension; configure extension on this page: 【Basic】----【Extensions】



By default, 10 existing extension numbers have already been given. They are from 800 to 809.

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



Extension Settings

Item	Explanation
SIP/IAX2	Choose extension protocol.
Name	Extension Name (English Character Only), e.g. Tom.
Extension	Extension Number connected to the phone, e.g. 888.
Password	Random password. (6-16 digits, e.g.123456)
Outbound CID	Override the caller ID when dialing out with a trunk.
Dial Plan	Please choose the Dial Plan which is defined in the menu “Outbound Routes”.
Analog Phone	Please select the related FXS port for your analog phone.

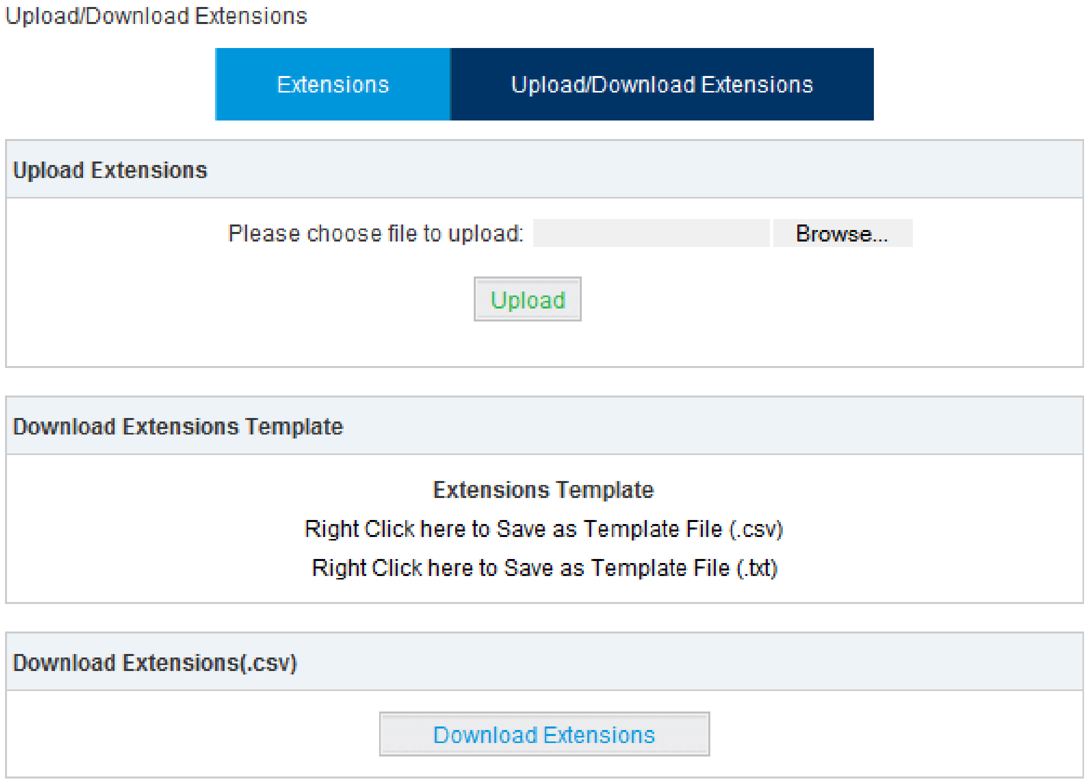
Item	Explanation
Voicemail	Select this option to open the voicemail account
VM Password	Set password for Voicemail, e.g. "1234"
Delete VMail	Check this option to delete voicemail from system after it's sent to mail box.
Email (Fax/Voicemail)	Extension user's mail box, which is used for receiving fax or voicemail (you need to open the function to fax to email/voicemail), e.g. Tom@gmail.com
Web Manager	It's allowed to login Extension Management Panel to manage extension like voicemail, call recording, call transfer, etc. when you select this option.
Agent	Check this option to set this extension user as agent.
Call Waiting	Enable call waiting
Allowing Being Spied	Check this option to allow being spied.
NAT	Check this option if extension user or the phone is located after the NAT (Network Address Translation) available gateway.
Pickup Group	Select the Pickup Group which the extension user belongs to.
Mobility Extension	After checking this option, you must set mobility extension number. User can make calls to the IP PBX server with this mobility number, and have all rights of this extension, e.g. Outbound Call, Internal Call, Listen to the voicemail.
Transport	Select the Transport Protocol: UDP, TCP, TLS
SRTP	Enable SRTP
DTMF Mode	Default DTMF is rfc2833. It can be changed if necessary.
Video Call	Check to enable video call for this extension. And select the audio codecs you need to use.
Permit IP	Set computer permitted IP to visit this IP PBX, e.g.,192.168.1.77 or 192.168.10.0/255.255.255.0. Computer with other IPs is not allowed to visit this IP PBX.
Audio Codec	Select what audio codec you need to use.

Note

1. There are a few default extensions whose number starts with "8XX". You can add or delete extension as required
2. Maximum extensions: **500 on IPX-2500**; **200 on IPX-2200**.
3. For security reason, the default password consists of random characters or numbers, e.g., BjCnWsNJ-d, and every time when you reset to default system, it will randomly have a new password again

Upload/Download Extensions

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

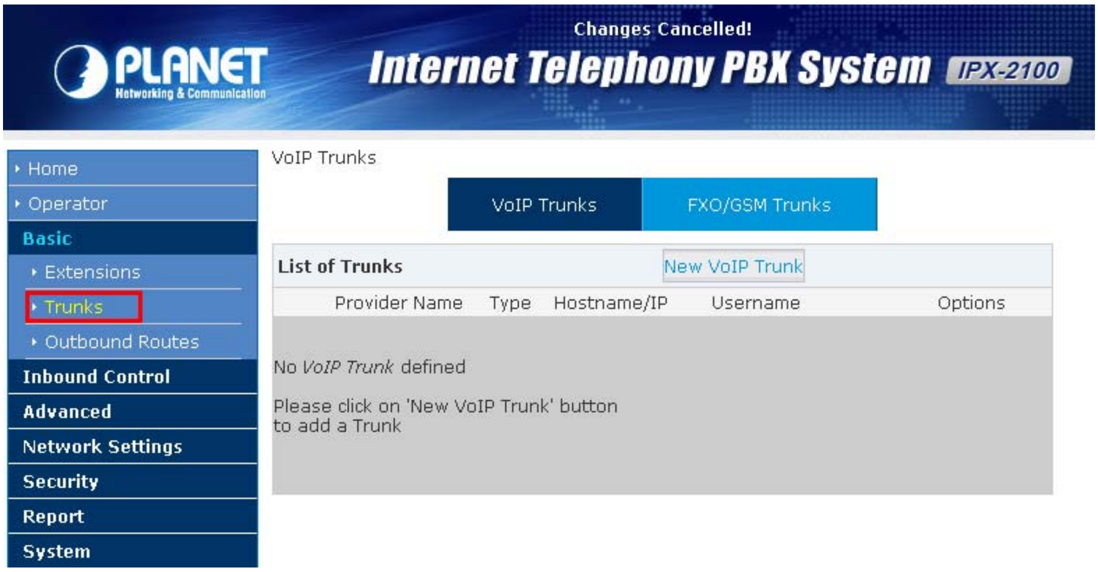


- Upload Extensions: Here you can upload .csv or .txt file to generate extensions.
- Download Extensions Template: Here you can download a template file in .csv or .txt format. Inside there are examples given, you can follow the examples to add your desired new extensions in the same format, and the new file can be used to upload to IP PBX system to generate new extensions.
- Download Extensions (.csv): Here you can download the existing extensions in the system for backup. The downloaded CSV file can be used for extension list recovery.

3.3 Outbound Call

3.3.1 Trunks

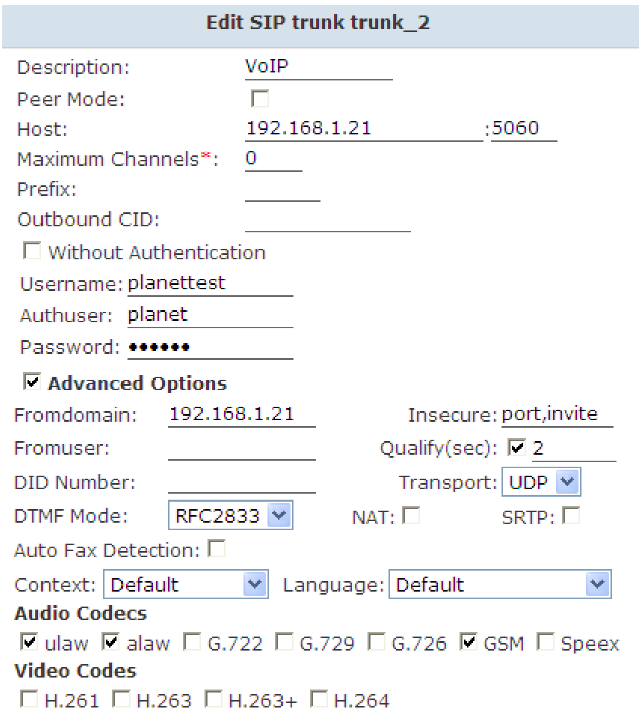
If you want to set up outbound call to connect to PSTN (Public Switch Telephone Network), GSM (Global System for Mobile Communications) or VoIP provider, please configure on this page: 【Basic】->【Trunks】



Planet IP PBX supports 3 kinds of trunks: VoIP Trunks, FXO Trunks and GSM Trunk.

VoIP Trunks

1.Click【VoIP Trunk】->【New VoIP Trunk】:



Planet IP PBX can register as a SIP user agent to a SIP proxy (provider). If you have subscribed VoIP service from ITSP, then with the account details given by them you can setup a VoIP trunk on Planet IP PBX system for the user extensions to share this trunk to make outbound phone calls.

Item	Explanation
Description	Define the VoIP (figure or character).
Protocol	Select protocol for outbound route, SIP or IAX2.
Host	Set host address (provided by VoIP Provider).
Maximum Channels	Set maximum channels for simultaneous call. (Only for outbound call; “0” = no limitation).
Prefix	The prefix will be added in front of your dialed number automatically when the trunk is in use.
Caller ID	This Caller ID will be displayed when user makes an outbound call. Note: This function must be supported by local provider.
Without Authentication	If you don’t need the Authentication when connecting the IP PBX, please check this option.
User Name	User Name provided by VoIP Provider.
Authuser	The optional authorization user for the SIP server
Password	Password provided by VoIP Provider.
Advanced Options	Advanced options for this trunk, e.g., codec, dial plan, etc.
Domain	The domain is where you register your username.
Insecure	Default value is “port, invite”; “port”-- Allow matching of peer by IP address without matching port number; “invite”-- Do not require authentication of incoming INVITEs.
From User	Fromuser = yourusername; Many SIP providers requires this.
Qualify (sec)	Asterisk sends a SIP OPTIONS command regularly to check that the device is still online. Default value is 2 (sec).
DID number	Self defined, it can be used to set up number DID.
Transport	Default transport type for SIP messages
DTMF Mode	Used to tell the system how to detect the DTMF (Dual Tone Multi Frequency) key press. Choices are inband, rfc2833, or info. By default, we use RFC2833.
NAT	With this option enabled Asterisk may override the address/port information specified in the SIP/SDP messages, and use the information (sender address) supplied by the network stack instead.

Item	Explanation
Context	Custom dial plan for this trunk, by default it's using the "default" dial plan. Configure only if this trunk is for branch office integration, so the calls coming from the other side can dial out from this IPPBX trunk directly. DO NOT change it unless you know how exactly this option works.
Language	You can choose a language here; the system will indicate the incoming calls from this trunk with the voice prompts you selected.
Audio Codecs	Select the audio codec/codecs the provider can support.
Video Codecs	If the ITSP supports video call, you can enable compatible video codecs here for video phone calls.

Note	Except the configuration options related to the service provider and your account details, please do not change the trunk advanced parameters if you are not familiar with. After the SIP trunk is successfully added you can see it's listed here on this page

You can configure the Analog/GSM line through PLANET IP PBX. The same analog line can't be used in multiple trunks. If you don't have available analog/GSM trunk, you can't set up trunk.

2) FXO/GSM Trunk

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

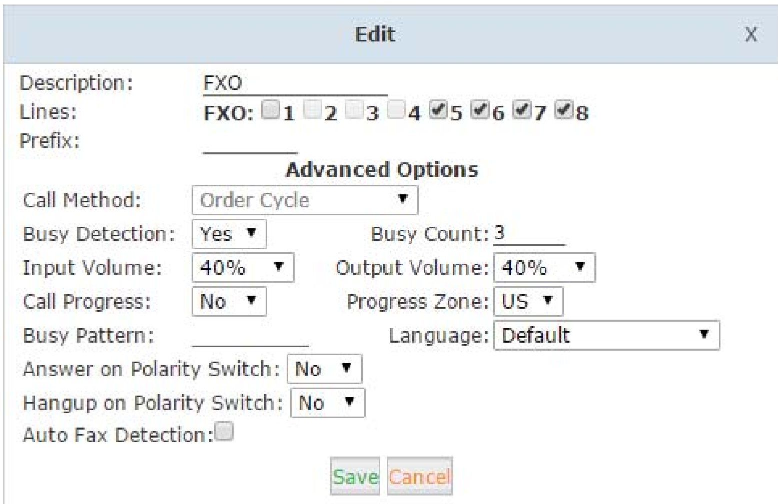
On the IPPBX front panel, red LED indicates the RJ11 interface is FXO. You should attach the telephone wire from your telecom to the FXO ports. Once connected you should be able to see the connection status on the *Operator* page in the "FXO/FXS/GSM Ports" section.

FXO/FXS/GSM Ports

Status	Signal Strength	Type	Port	BLF Label
Connected		FXO	1	Channel1
Connected		FXO	2	Channel2
Connected		FXO	3	Channel3
Connected		FXO	4	Channel4
Disconnected		FXO	5	Channel5
Connected		FXO	6	Channel6
Connected		FXO	7	Channel7
Connected		FXO	8	Channel8

To be able to use these lines connected on FXO ports to make phone calls, you have to use them to create trunk/trunks first. Navigate to web menu *Basic->Trunks->FXO/GSM Trunks.*

Click the “New FXO/GSM Trunk” button and you’ll see available port numbers that can be used.



Item	Explanation
Description	Define the description for this trunk (figure or character).
Lines	Available FXO and GSM ports.
Prefix	The numbers you dialed will first be manipulated by the dial rules, while the manipulated numbers reached the trunk before finally sending out to this prefix, which will be added to the numbers and then send out through this trunk. Usually you don’t need this prefix. Please leave this field blank.
Call Method	If in this trunk you have more than 1 FXO/GSM port selected, then this parameter defines how to use these ports for outbound phone calls.
Busy Detection	Enable busy tone detection; it is also possible to specify how many busy tones to wait for before hanging up.
Busy Count	Specify how many busy tones to wait for before hanging up, configurable only if Busy Detection is enabled.
Input Volume	The volume of the calls from FXO channel/channels which have been received.
Output Volume	The volume of the calls from FXO channel/channels which have been made.
Call Progress	If turned on, call progress attempts to determine answer, busy, and ringing on phone lines. This feature is HIGHLY EXPERIMENTAL and can easily detect false answers so don't count on it being very accurate.

Item	Explanation
Progress Zone	Progress zone also affects the pattern used for busy detection, only effective when Call Progress is turned on.
Busy Pattern	If busy detect is enabled, it is also possible to specify the cadence of your busy signal.
Language	You can choose a language here; the system will indicate the incoming calls from this trunk with the voice prompts you selected.
Answer on Polarity Switch	For FXO (FXS signal) ports watch for a polarity reversal to mark when an outgoing call is answered by the remote party.
Hang up on Polarity Switch	In some countries, a polarity reversal is used to signal disconnect of a phone line. If the hang up polarity switch option is selected, the call will be considered "hung up" on a polarity reversal.

3) GSM Trunk

If you have ordered GSM modules for your IP PBX, the user extensions will be able to make and receive phone calls from the mobile network. You have to insert the SIM cards into the SIM slots of the GSM modules (Called IPX-21GS) and then install the modules to the IP PBX module slots. Antennas should be properly installed and placed in the open space for better signal reception. After this, power on the IP PBX and you'll be able to configure GSM trunks the same as you configure FXO trunks.

GSM Specifications

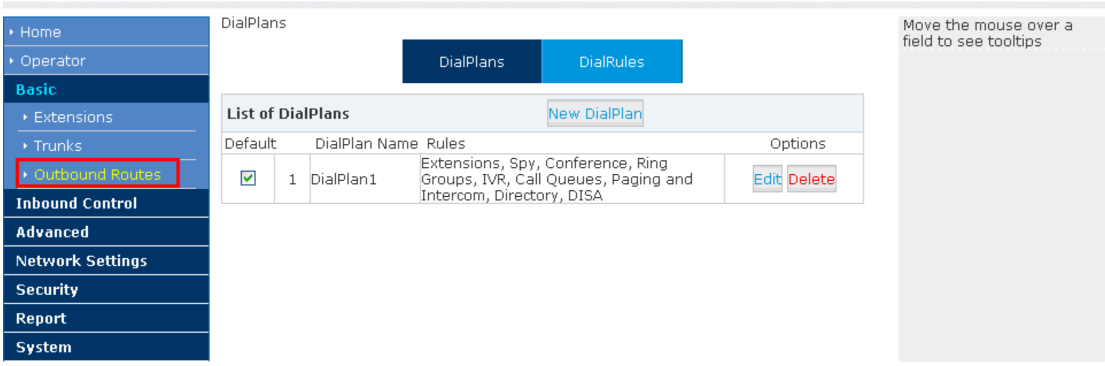
Module	Working Frequencies
IPX-21GS	GSM/GPRS 850/900/1800/1900MHz

3.3.2 Outbound Routes

Outbound Routes enable you to tell Planet IP PBX which Trunks (phone lines) to use when people dial external telephone numbers. A simple installation will direct Planet IP PBX to send all calls to a single trunk. However, a complex setup could have an outbound route for emergency calls, another outbound route for local calls, another for long distance calls, and perhaps even another for international calls.

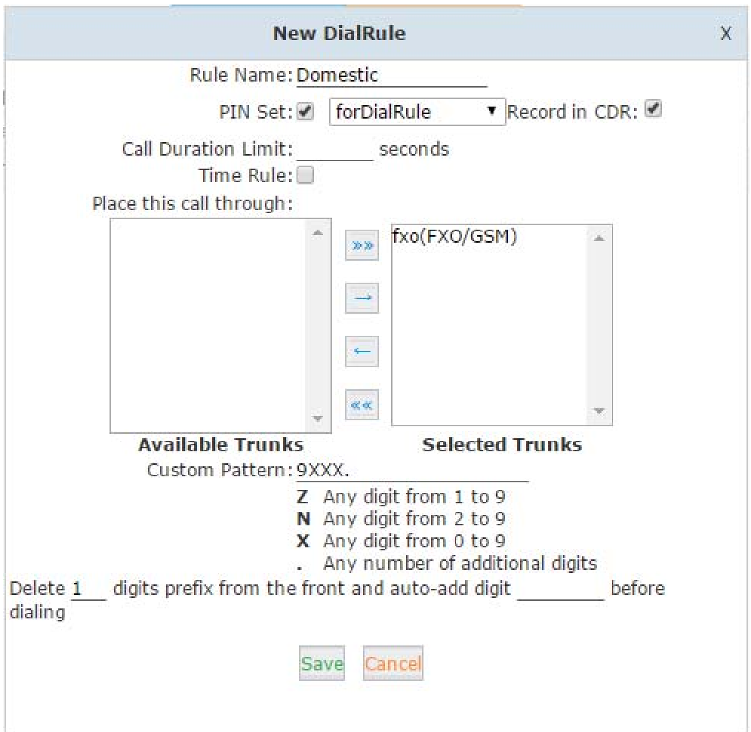
With the above mentioned possibilities, you may already have several trunks configured in the Planet IP PBX system. To be able to use different trunks for outbound phone calls, you'll have to configure several dial rules and maybe also several dial plans.

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



Dial rules

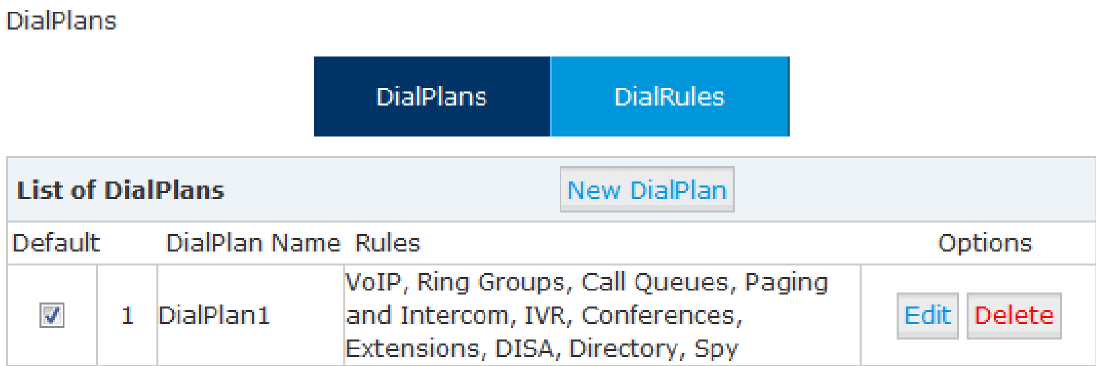
On this page, user can configure the basic match pattern of the outbound routes and create different dial plans. Please configure by clicking【Add a Dial Rule】



Item	Explanation
Rule Name	A name for this dial rule
PIN set	A collection of PIN codes for granting outbound phone calls.
Record in CDR	Record the PIN codes used for outbound phone calls along with the user extension number and the dialed numbers.
Call Duration Limit	Specify how long the calls can be made using this dial rule.
Time Rule	Set a time condition when this dial rule can be used.

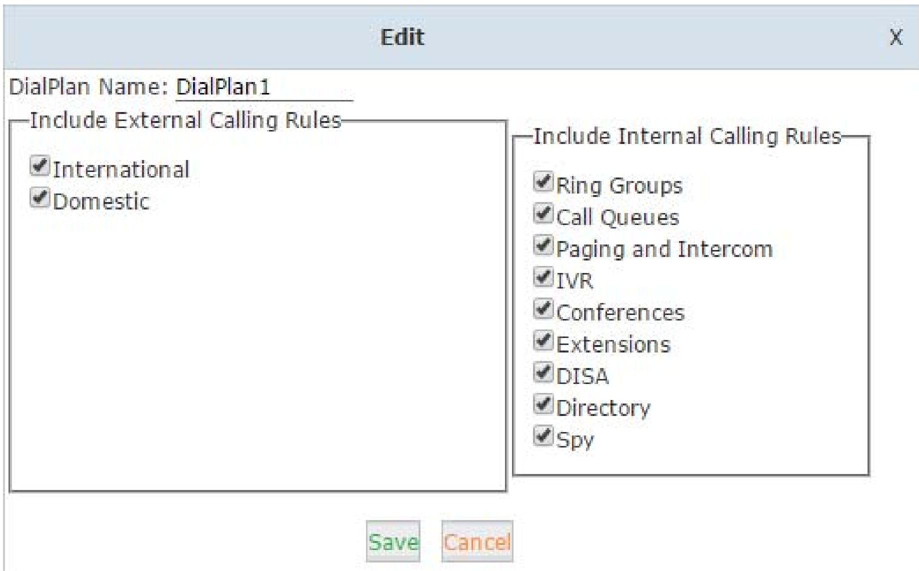
Item	Explanation
Available Trunks	All existing trunks in the IPPBX system.
Selected Trunks	Trunk/Trunks can be used by this dial rule.
Custom Pattern	Dial patterns act like a filter for matching numbers dialed with trunks. The various patterns you can enter are similar to Asterisk's definition of them: X — Refers to any digit between 0 and 9 N — Refers to any digit between 2 and 9 Z — Any digit that is not zero. (e.g. 1 to 9) . — Wildcard. Match any number of anything. Must match *something*.
Delete ____ digits prefix from the front and auto-add ________ digit before dialing	The first blank is to strip some digit/digits before dialing out. Here you need to fill in a count of number. The second blank is to prepend some digit/digits before dialing out. Here you need to fill in the exact number to be added in front of the dialed number. For example a user dialed 912345678 using the dial rule introduced above, the prefix 9 at the first digit will be removed, and 00 will be added, so eventually the user will call the number 0012345678.

Dial plans

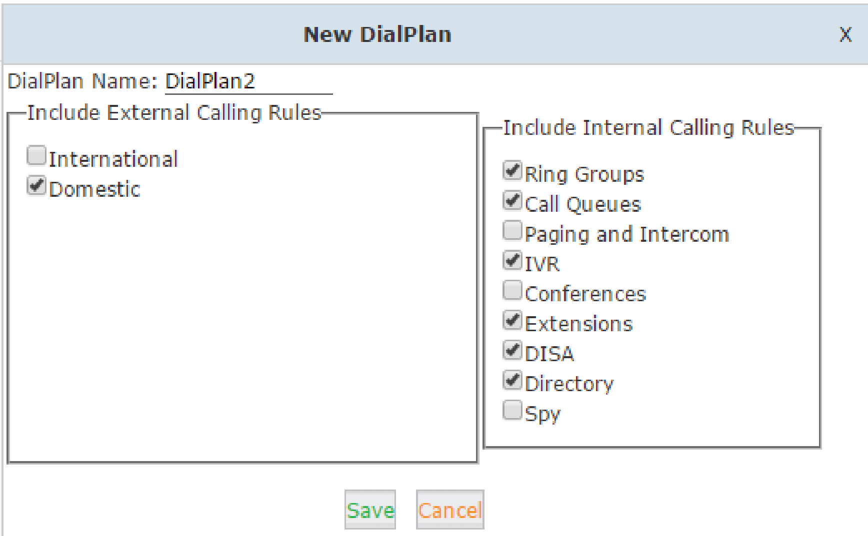


There’s a default dial plan already existed in the IP PBX system. Normally you just have to click the “Edit” button on the default dial plan “DialPlan1” and tick on all dial rules to enable to the extension users to call any destinations using the trunk lines of the IP PBX system.

User can create dial rule for dial plan on this page:



The calling rules in the left column are for external calls and calling rules in the right column are for internal calling. If you want to restrict some uses from calling out through some trunk lines or you don't want them to be able to call some internal destinations, you can create new dial plan by clicking the "New DialPlan" button.



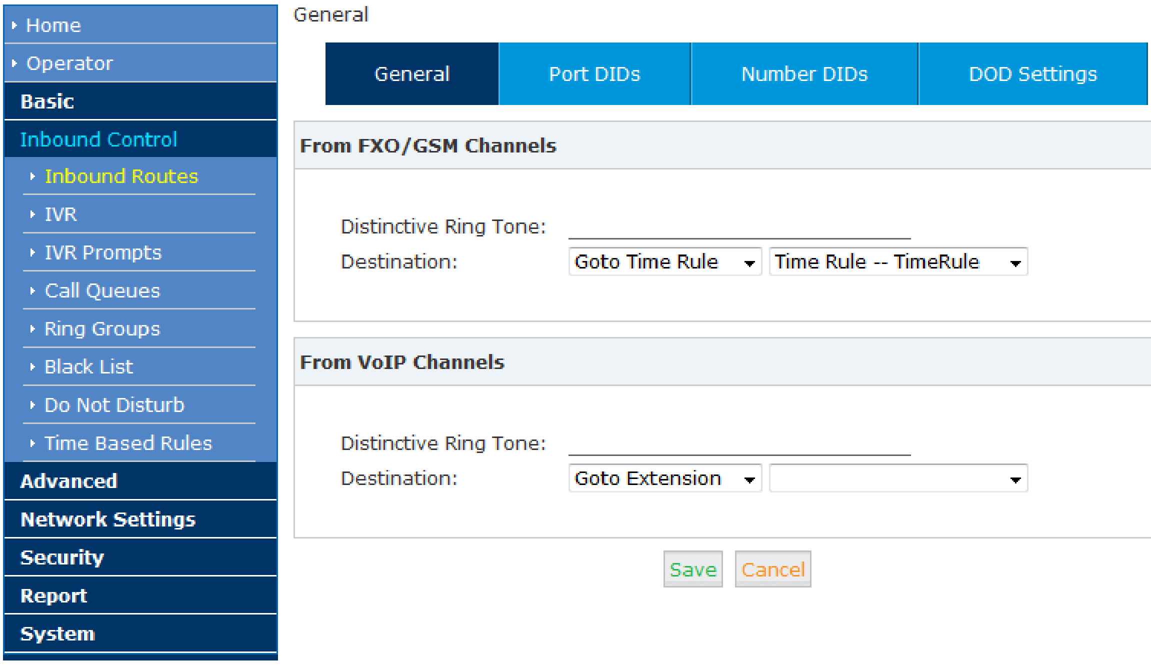
In the new dial plan you disable the rules you don't want others to use and save. After this on the extension configure page give them different dial plans; then they have different dial permissions.

3.4 Inbound Call

3.4.1 Inbound Routes

When a call is made from outside, you want to forward this call to an extension or IVR. This Chapter will introduce you how to deal with the inbound calls. The Inbound Control section is where you define how IP PBX system handles incoming calls. Typically, you determine the phone number that outside callers have called (DID Number) and then indicate which extension, Ring Group, Voicemail, or other destination to which the call should be directed.

Please configure it on this page:【Inbound Routes】



General

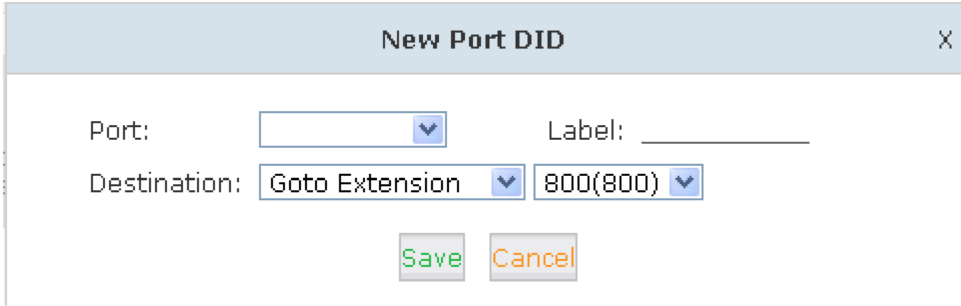
Distinctive Ring Tone: Mapping the custom ring tone file, e.g., set distinctive ring tone as “External”, the phone will play this ring tone when receiving the call. Note: The phone must support such feature as well.

When incoming calls come from outbound line (FXO/GSM, VoIP), the calls can be accessed to Extension User, Call Queue, Conference, IVR, etc. You can choose freely based on your condition.

Port DIDs

If user wants to make the incoming call from the outbound line (FXO/GSM trunk) access to the specified extension user, call queue, conference or IVR, please configure it here:

Click【Port DIDs】->【New Port DIDs】:

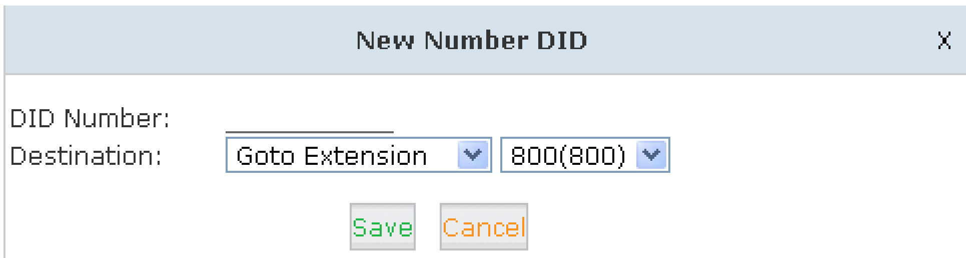


Item	Explanation
Port	Select the port for outbound line.
Label	Set a label for this port. When incoming calls are from this port, the label will be displayed.
Destination	Incoming calls will access directly to this destination (extension user, call queue, conference, or IVR).

Number DIDs

If user wants to make an outbound line (VoIP Trunk) access to the specified extension/queue/conference/IVR, please use this feature:

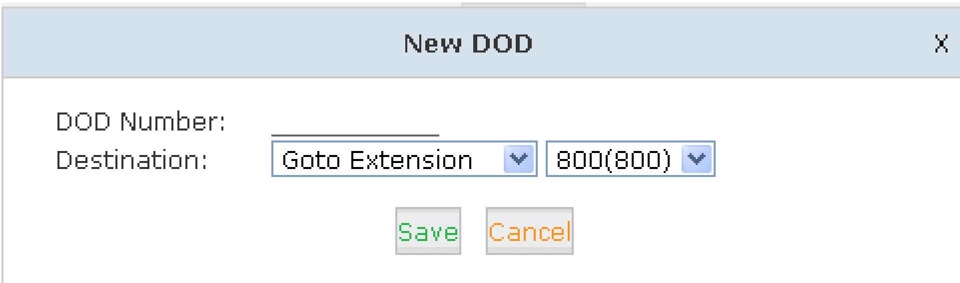
Click【Number DID】->【New Number DID】:



Item	Explanation
DID Number	DID number calling into VoIP (This number is configured in the advance option of VoIP trunk).
Destination	Choose a specified extension, call queue, conference or IVR to be directed to call.

DOD Settings

If user wants to make the outbound call directly to the specified extension user, call queue, conference, IVR, please configure it here. Click【DOD Settings】->【New DOD】

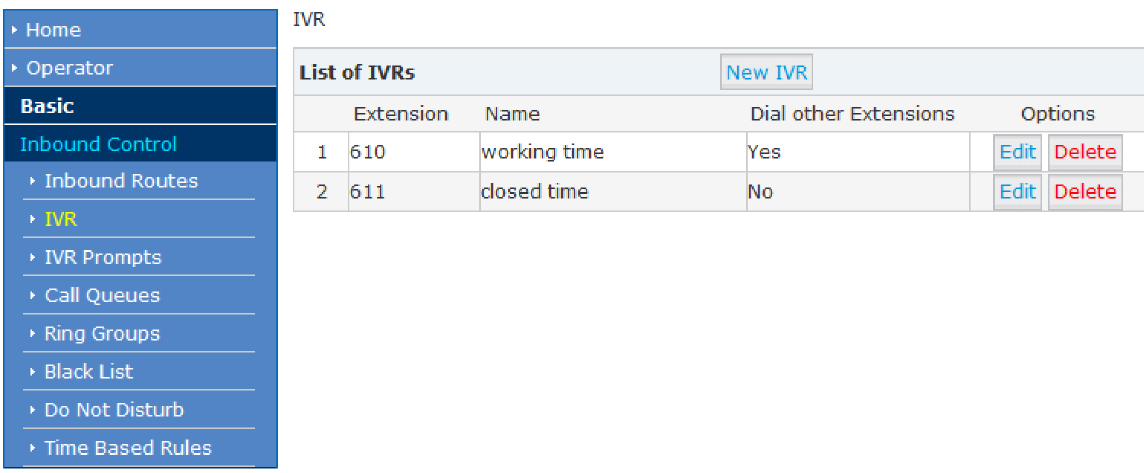


Item	Explanation
DOD Number	Set the DOD number, and use it to match the Caller ID. If matched, the call will access to the defined destination.
Destination	Outbound calls will access directly to this destination (extension user, call queue, conference, or IVR).

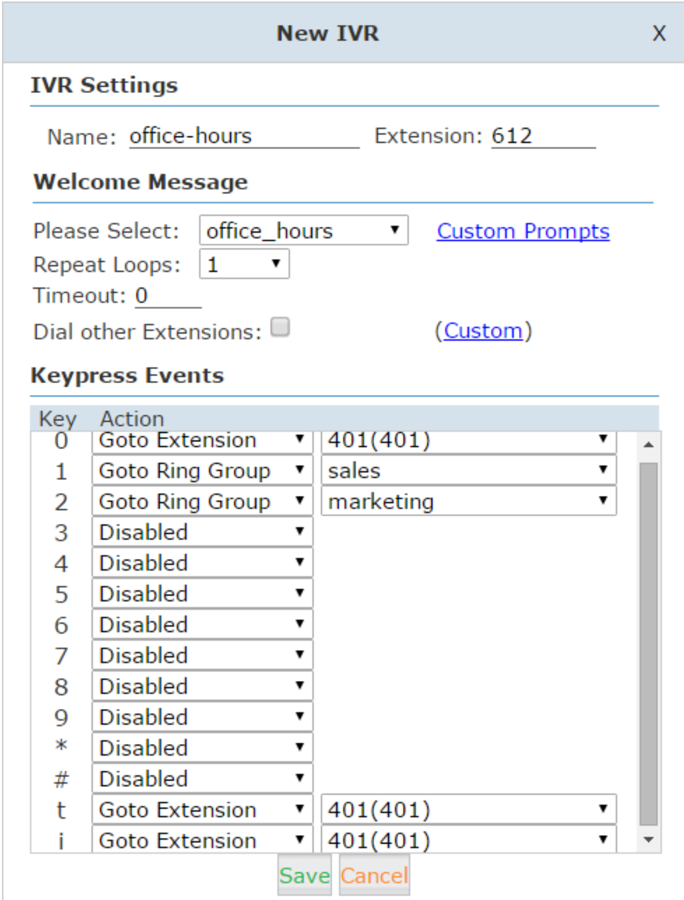
3.4.2 IVR

IVR will improve office efficiency based on your requirement.

Please configure on this page【Inbound Control】->【IVR】:



Click【New IVR】to create a new IVR:

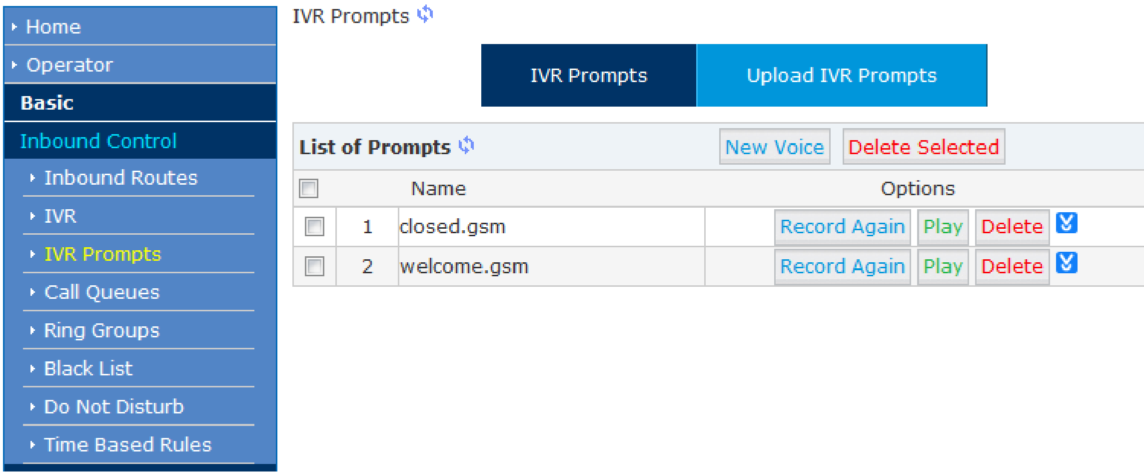


Item	Explanation
Name	Set a name for the IVR
Extension	Extension number for the IVR, by calling this number can access the IVR menu.
Please Select	Select a voice prompt for this IVR menu.
Custom Prompts	Click this button to navigate to *Inbound Control->IVR Prompts* page for new voice prompts.
Repeat Loops	Define how many times to play the IVR menu to the caller.
Timeout	Timeout for key pressing of each IVR loop.
Dial Other Extensions	If enabled, the caller can dial extension number directly on IVR.
Custom	By clicking “Custom” you can set dial plan for this IVR menu. The callers on IVR would be able to dial some other destinations the dial plan allows.(Not recommended)
Key Press Events	Define which destination to go by pressing a key on the phone keypad. If the undefined keys is pressed, then it will be handled by the “i” parameter; “i” means invalid. And “t” stands for

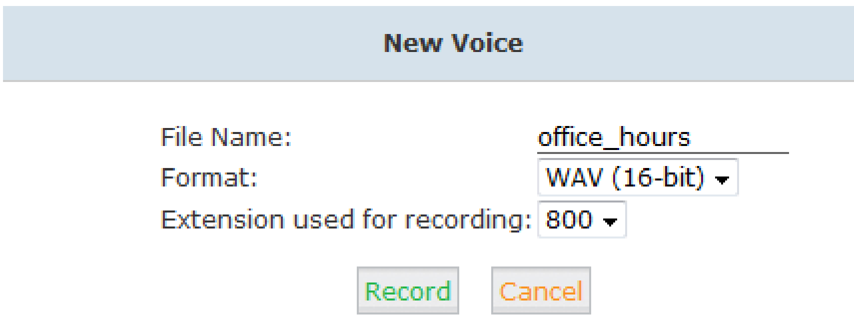
Item	Explanation
	timeout, after all IVR loops played completely without pressing any key the incoming call will be handled by "t" parameter.

3.4.3 IVR Prompts

To configure IVR menu on IP PBX system you'll first need to record the IVR prompts. The IVR prompts will indicate the callers how to place their calls

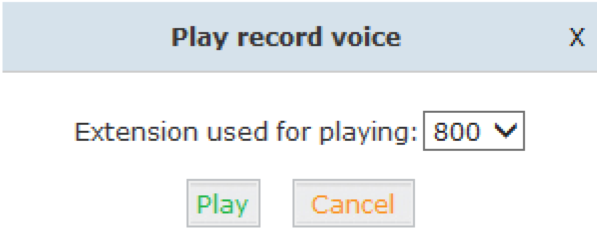


Click【IVR Prompts】----【New Voice】to create new IVR prompt:



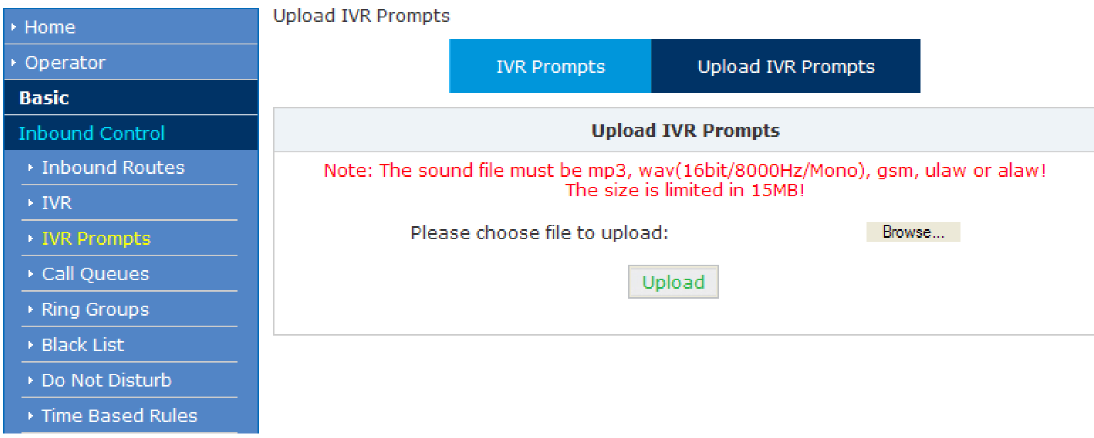
Item	Explanation
File Name	Define a name for this voice file.
Format	Select the voice format, GSM/WAV (16bit) supported only.
Extension used for recording:	Select the extension which is used for recording the IVR prompt. Click【Record】, this extension will ring, and then you can pick up the phone and record.

If you want to hear the prompt, please click【Play】:



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

Upload IVR prompt

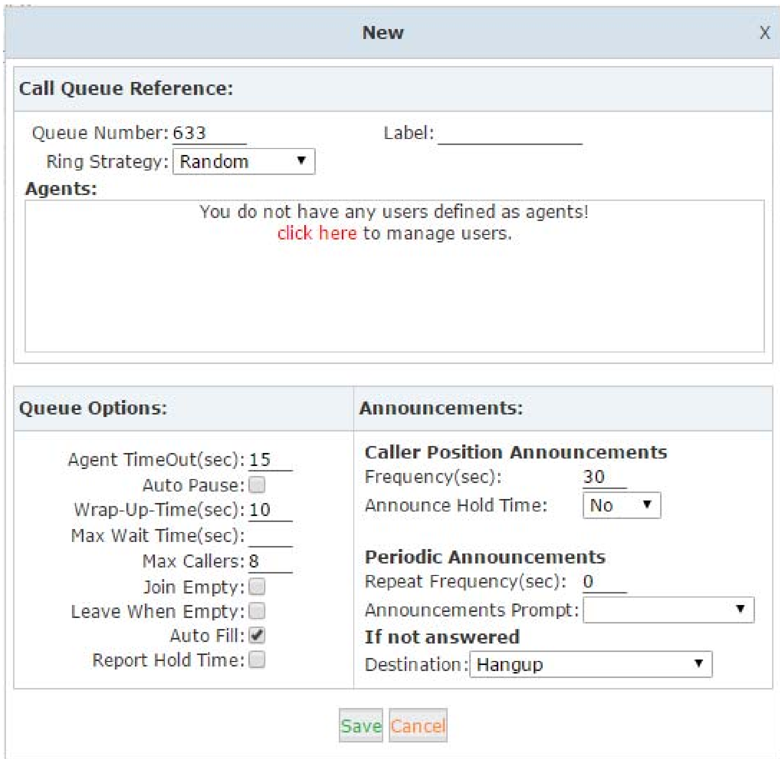


Note	Uploading customized audio file must be in the mp3, wav, gsm, ulaw, alaw format, and size must be less than 15MB.

3.4.4 Call Queue

A call queue places incoming calls in line to be answered while extension users are busy with other calls. The queued calls are distributed to the next available extension user in the order received. After they have been created, they can be assigned to specific extensions and configured to feature greetings, messages, and hold music.

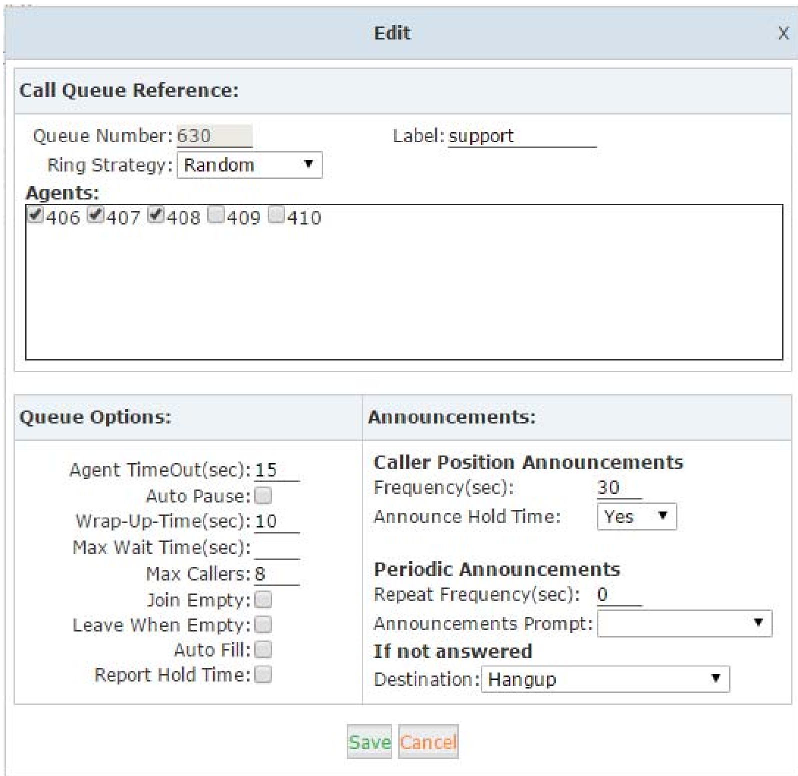
There are 3 existing call queues. All you have to do is click the “Edit” button to configure them. If you want more call queues, you can click “New Call Queue” to add more queues.



Here we can see in the “Agents” field there’re no available agents to be assigned to the call queues. Click “click here” you’ll be redirected to the extension page to determine which extensions will be employed as call queue agents.

Tick the checkbox of the extension numbers which will be employed as call queue agents, then click the “Edit Selected” button and tick the “Agent” option in “Other Options” section.

Save and go back to *Inbound Control->Call Queues* page again and configure the existing call queues and add new call queues with available agents.



Item	Explanation
Queue Number	Define an extension number to identify the queue.
Label	Define the label for the queue. A user can be agent of multiple queues, by giving a label for the call queue; if an incoming call is distributed to an agent the label will be displayed on the phone screen along with the caller ID. So a call queue agent can tell from which call queue the call is coming from.
RingAll	Ring all available agents until one answers (default).
RoundRobin	Starting with the first agent, ring the extension of each agent in turn until the call is answered.
LeastRecent	Ring the extension of the Agent who has least recently received a call
FewestCalls	Ring the extension of the Agent who has taken the fewest number of calls.
Random	Ring the extension of a random Agent.
RRmemory	RoundRobin with Memory, like RoundRobin above, except instead of the next call starting with the first agent, the system

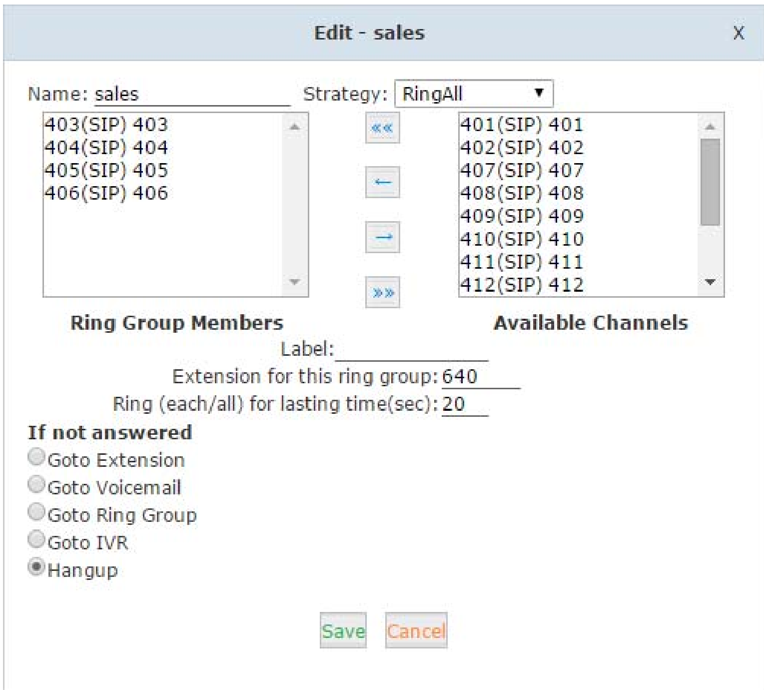
Item	Explanation
	remembers which extension was called last and begins the round robin with the next agent.
Agent	Check each agent that is to be a member of this specific Call Center Queue.
Agent TimeOut (sec)	Specify the number of seconds to ring an agent's extension before sending the call to the next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

Item	Explanation
	play).
Announcement Prompt	Select a prompt as the Announcements Prompt from the IVR Prompts.

3.4.5 Ring Groups

Ring Group is a collection of extensions. When a call to a ring group is made, all extensions in this ring group will ring in different ways based on their different configurations. If ring time exceeds a defined time, the call will be directed to IVR or others based on your configuration.

There isn't any data in the factory default 【Ring Groups】, please configure it here.
Click【Inbound Control】-> 【Ring Groups】->【New Ring Group】:



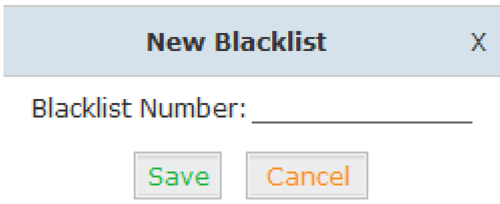
Item	Explanation
Name	Define a name for the Ring Group.
Strategy	Define how to ring the group members; select “RingAll” will ring all the member extensions at the same time, select “Ring In Order” will ring the member extensions one by one.
Ring Group Members	The extensions selected to be the members of the ring group.
Available Channels	All available extensions/channels can be added to the ring group.
Label	The extensions can be members of multiple ring groups, by

Item	Explanation
	giving each ring group a different label, if an incoming call rings a ring group the label will be displayed on the phone screen along with the caller ID. So a ring group member can tell from which ring group the call is coming in.
Extension for this ring group	By calling this extension can reach the ring group members
Ring(each/all) for lasting time(sec)	Ring duration of the group members.
If not answered	Setup a destination to redirect the incoming calls to, if no one answers.

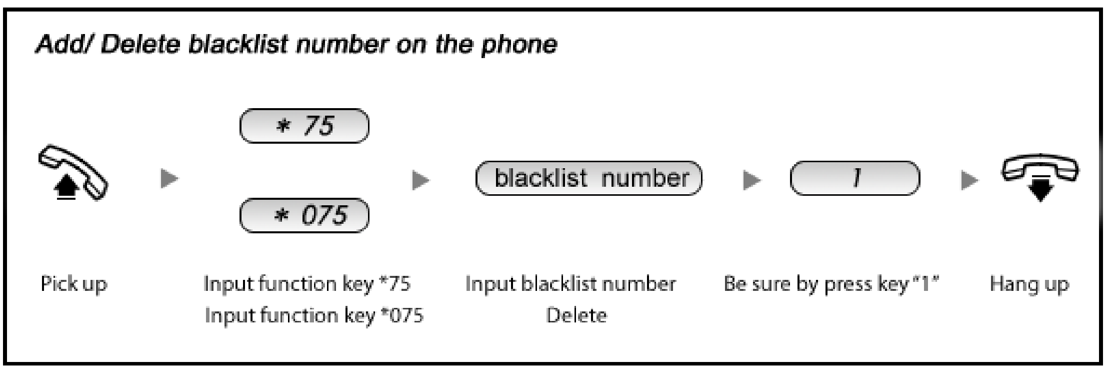
3.4.6 Black List

Before call spy can work, you have to make sure the extensions to be spied on have the “Allow Being Spied” option enabled on extension settings page.

If some numbers need to be blocked, you can use this functionality. Please configure it here: Click【Inbound Control】->【Blacklist】->【New Blacklist】



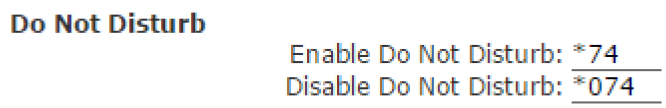
Input caller's number in the blank, then this caller's number will be blocked when the call comes again. Meanwhile, extension user can add or delete the blacklisted number by function key on the phone. Please operate according to the following diagram:



Reference Parameters and Explanation of the Blacklist:

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

3.4.7 Do Not Disturb

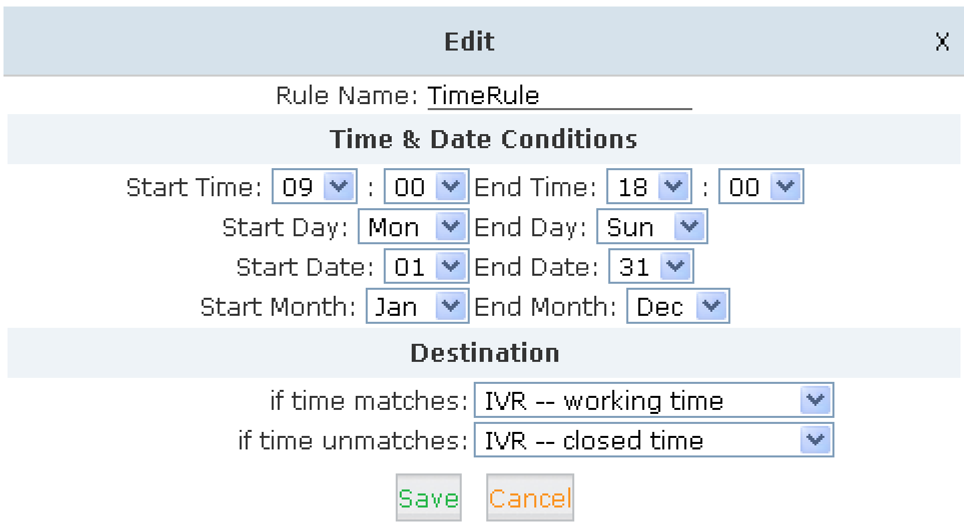


With Do Not Disturb (DND) feature enabled, an extension can make phone calls but others cannot call this extension. An extension user of the IP PBX system dials *74 from their phone, system will play a beep sound to indicate DND has been activated. To disable DND, just dial *074, another beep sound will be played and DND has been deactivated.

3.4.8 Time-based Rules

For the companies and shops, they all have their own business hours and non-business hours. Routing the incoming calls by proper time conditions is much more reasonable.

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



New Time Rule:

Item	Explanation
Rule Name	Define the name for this Time Rule.
Time & Date Conditions	Set time segment for Day/Date/Month.
Destination	How to deal with the inbound call in different time segments. For

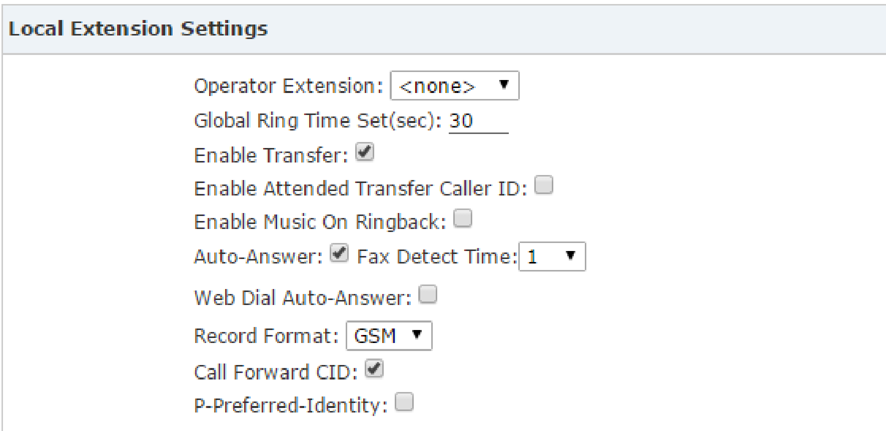
Item	Explanation
	example, inbound call can be directed to operator in working time.

3.5 Advanced

3.5.1 Options

3.5.1.1 General

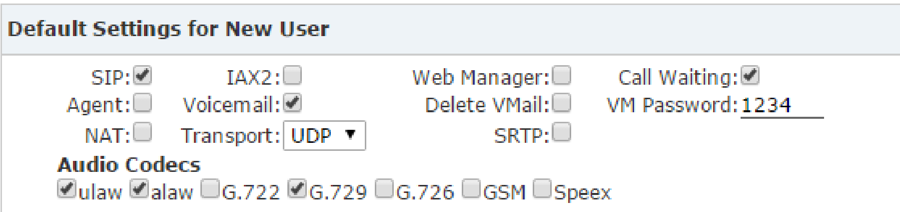
Here on this page you can configure some global options for all the user extensions. In the "Local Extension Settings" section you have the options shown below that can be configured.



Item	Explanation
Operator Extension	Choose an extension to be operator extension. While an incoming call had been directed to voicemail, by pressing '0' the caller can get to operator extension.
Global Ring Time Set(sec)	If not specifically configured, the incoming call will ring the extension for the time given here.
Enable Transfer	If enabled, the extension users will be able to do call transfer.
Enable Attended Transfer Caller ID	Normally if you use feature code *2 to transfer a call to another extension, the extension user only sees your extension number as caller ID but not the actual caller ID, by enabling this option the real caller will be passed to the user extension.

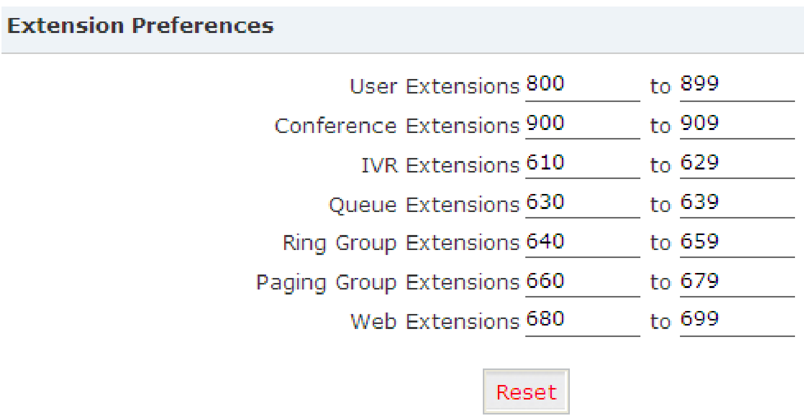
Item	Explanation
Enable Music On Ringback	If enabled this option, callers will hear music instead of ringback tone while calling other extensions.
Auto-Answer	Auto answer enables the IPPBX to automatically answer the inbound calls from analog ports.
Fax Detect Time	If auto answer enabled, you are able to configure the fax auto detection time here.
Web Dial Auto-Answer	Enable/disable auto answer of the extension numbers while dialing from Web GUI.
Record Format	Choose GSM or WAV as the call recording format.
Call Forward CID	Allow passing the real caller ID to the forwarded number.
P-Preferred-Identity	The P-Preferred-Identity header is used among trusted SIP entities (typically intermediaries) to carry the identity of the user sending a SIP message as it was verified by authentication.

Default Settings for New User



In this section the options are for new extensions. If you have one of the options enabled, then the newly created extensions will all have this option enabled.

Extension Preferences

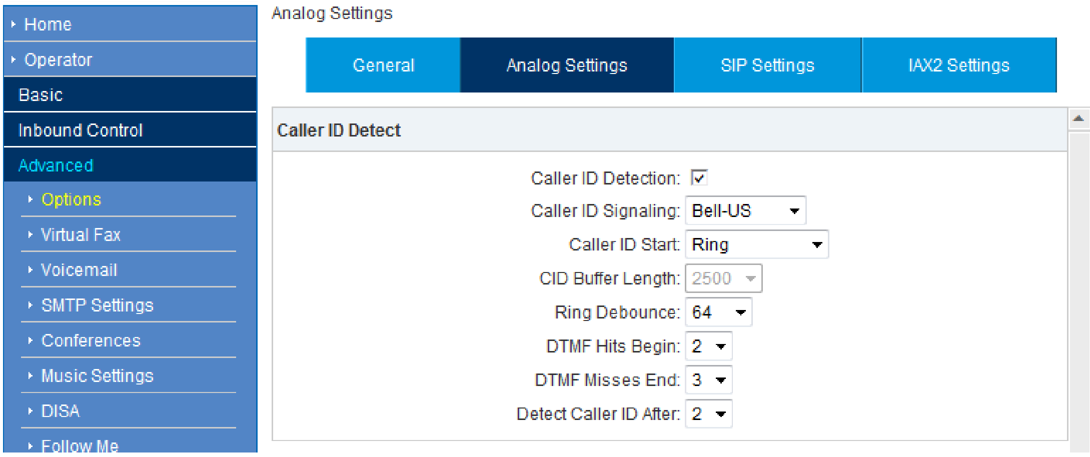


The user extension number and system extension number ranges are defined here to avoid confusion of the numbers in the IP PBX system. You can modify these number ranges according to your real applications.

3.5.1.2 Analog Settings

Analog Settings are used for configuring the IP PBX system working seamlessly with your telephone lines from the telecom.

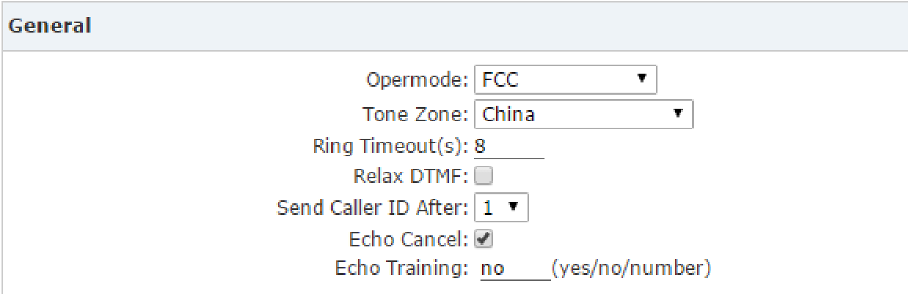
Caller ID Detect



These options are used to teach the IP PBX system how to detect caller identity (caller ID) from the PSTN lines on FXO ports.

Item	Explanation
Caller ID Detection	Enable/Disable Caller ID Detection
Caller ID Signaling	The signaling type applied on the PSTN lines to pass caller ID. Bell-US—Also known as BellcoreFSK. Used in the Canada, China, Hong Kong and US. DTMF—Dual Tone Multi-Frequency. Used in Denmark, Finland and Sweden. V23—Mostly used in UK. V23-Japan—Mostly used in Japan.
Caller ID Start	When the caller ID starts. Ring—Caller ID starts when a ring received. Polarity—Caller ID starts when polarity reversal starts. Polarity (India)—Can be used in India. Before Ring—Caller ID starts before a ring received
CID Buffer Length	The buffer length can be used to store caller ID info.

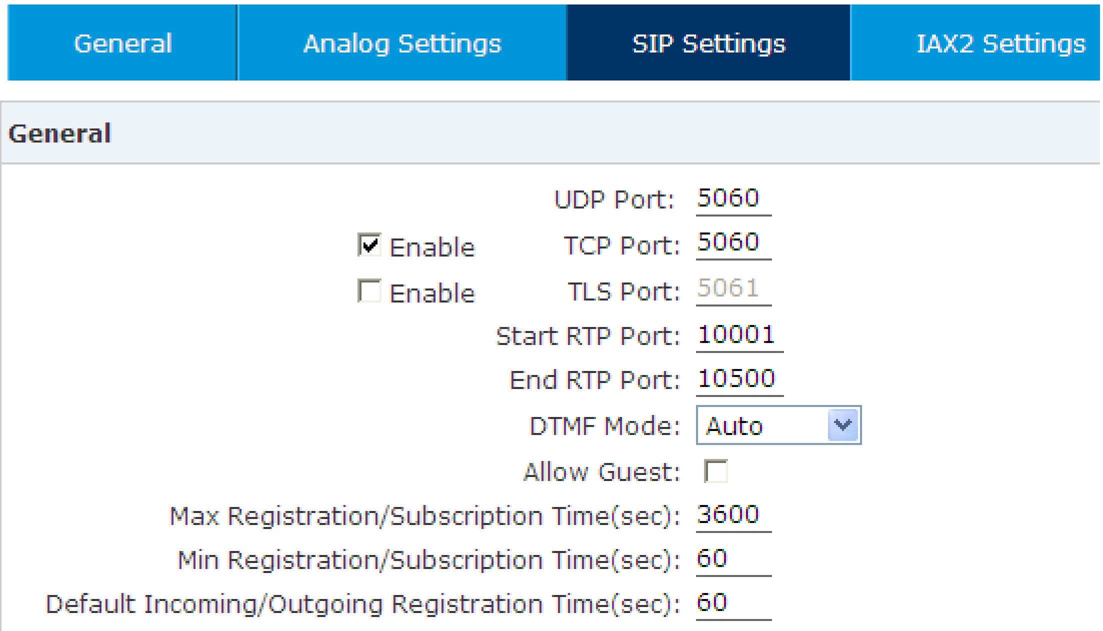
General



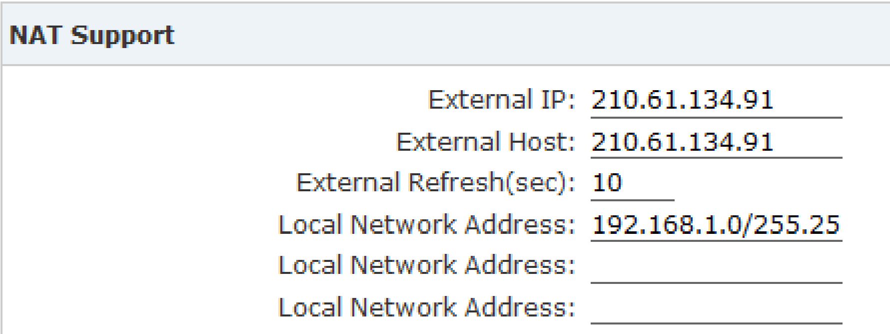
Item	Explanation
Opermode	Set the Opermode for FXO Ports
ToneZone	Select the tone zone of your country.
Ring Timeout(s)	FXO (FXS signaled) devices must have a timeout to determine if there was a hangup before the line was answered. This value can be tweaked to shorten how long it takes before DAHDI considers a non-ringing line to have hung up.
Relax DTMF	Relax DTMF
Send Caller ID After	Some countries (UK) have ring tones with different ring tones (ring-ring), which means the caller ID needs to be set later on, and not just after the first ring, as per the default (1).
Echo Cancel	Enable/Disable software Echo Cancel algorithm.
Echo Training	Enabling echo training will cause the PBX system to mute the channel, send an impulse, and use the impulse response to pre-train the echo canceller so it can start out with a much closer idea of the actual echo. Value may be "yes", "no", or a number of milliseconds to delay before training (default = 400). This option does not apply to hardware echo cancellers.

3.5.1.3 SIP Settings

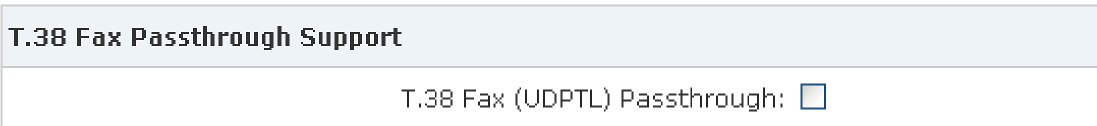
【Global SIP Settings】is appropriate for professionals. If anything needs to be modified, please contact our tech-support people.



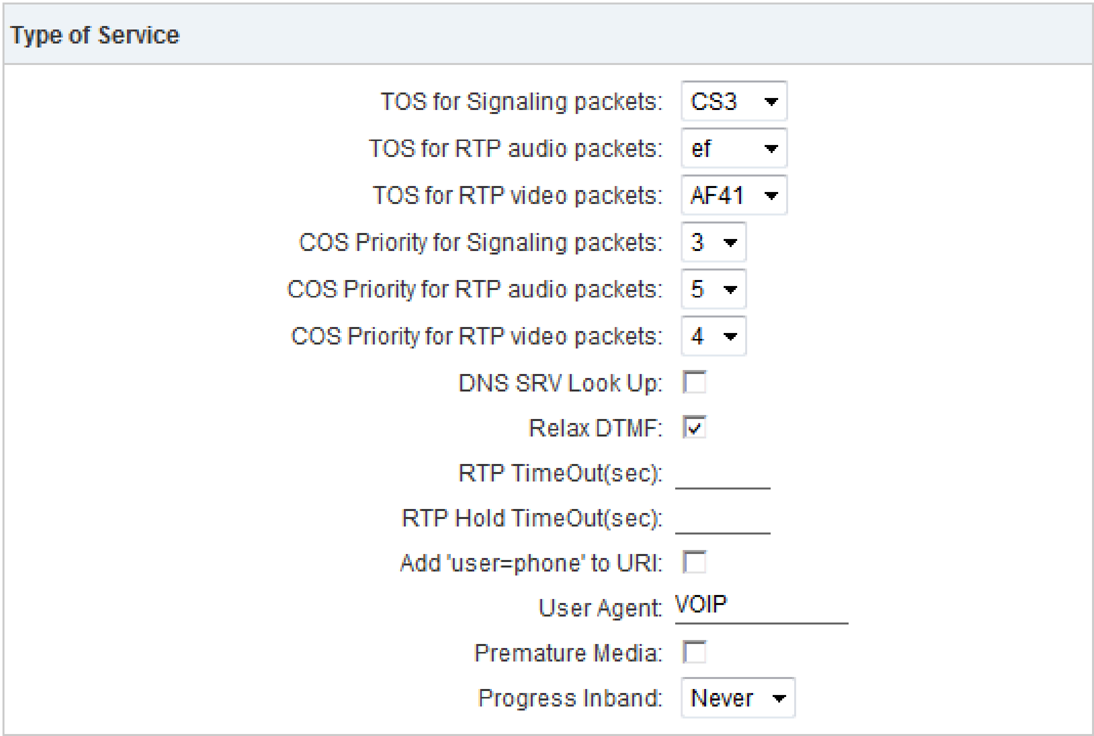
Item	Explanation
UDP Port to bind to	SIP standard port is 5060
TCP Port	Default TCP port is 5060
TLS Port	Default TLS port is 5061
Start RTP Port	RTP port range
End RTP Port	RTP port range
DTMF Mode	Set default DTMF mode for sending DTMF, support auto, RFC2833, inband, info. Default: RFC 2833
Allow Guest	This setting determines if anonymous callers are permitted to place calls to the IP PBX system. For security precautions please do not enable this option.
Max Registration/Subscription Time	Maximum duration (in seconds) of incoming registrations/subscriptions is 3600 seconds by default
Min Registration/Subscription Time	Minimum duration (in seconds) of registrations/subscriptions is 60 seconds by default
Default Incoming/Outgoing Registration Time	Default duration (in seconds) of incoming/outgoing registration



Item	Explanation
External IP	Address that we're going to put in outbound SIP messages if we're behind a NAT
External Host	Alternatively, you can specify an external host, and Asterisk will perform DNS queries periodically. Not recommended for production environments! Use external IP instead
External Refresh	How often to refresh external host if used. You may specify a local network in the field below
Local Network Address	192.168.1.0/255.255.255.0' : All RFC 1918 addresses are local networks, '10.0.0.0/255.0.0.0' : Also RFC1918, '172.16.0.0/12' : Another RFC1918 with CIDR notation, '169.254.0.0/255.255.0.0' : Zero conf local network

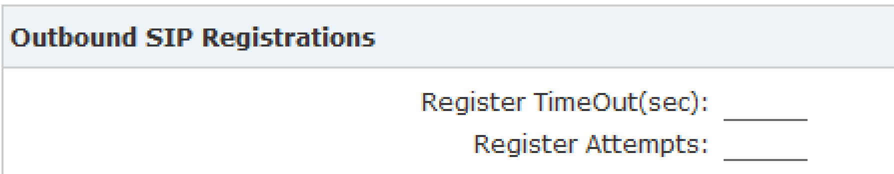


Item	Explanation
T.38 fax (UDPTL) Passthrough	Enables T.38 fax (UDPTL) passthrough on SIP to SIP calls



Item	Explanation
TOS for Signaling packets	Sets Type of Service for SIP packets
TOS for RTP audio packets	Sets Type of Service for RTP audio packets
TOS for RTP video packets	Sets Type of Service for RTP video packets
COS Priority for Signaling packets	Sets 802.1p priority for SIP packets.
COS Priority for RTP audio packets	Sets 802.1p priority for RTP audio packets.
COS Priority for RTP video packets	Sets 802.1p priority for RTP video packets.
DNS SRV Look Up	Enable DNS SRV lookups on outbound calls.
Relax DTMF	Relax DTMF handling.
RTP TimeOut(sec)	Terminate call if there is 60 seconds of no RTP or RTCP activity on the audio channel when we're not on hold. This feature enables the ability to hangup a call in the case of a phone disappearing from the network, for instance if the phone loses power.
RTP Hold TimeOut(sec)	Terminate call if 300 seconds of no RTP or RTCP activity on the audio channel when on hold.
Add 'user=phone' to URI	Enable this option if the SIP provider requires

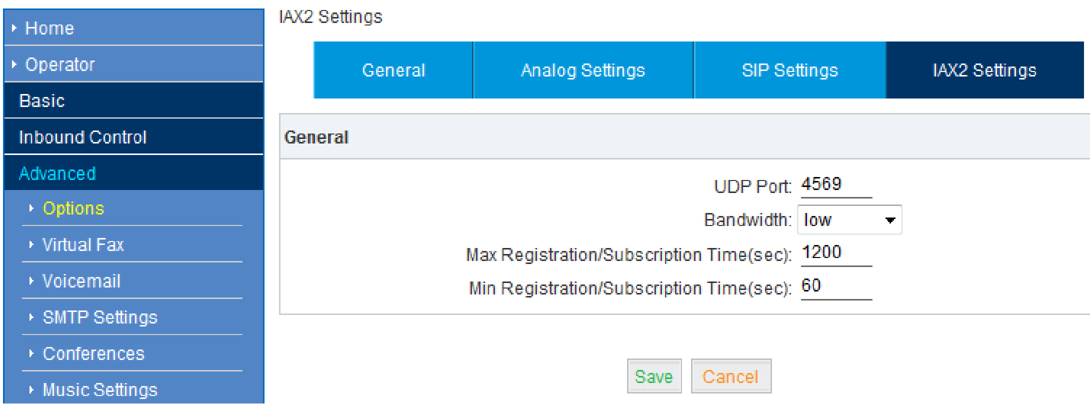
Item	Explanation
	";user=phone" on URI.
UserAgent	Allows you to change the user agent string. The default user agent string also contains the Asterisk version. If you don't want to expose this, change the user agent string here.



Item	Explanation
Register Time Out	Retry registration calls at every 'x' seconds (default 20)
Register Attempts	Number of registration attempts before we give up; 0 = continue forever

Note	In the extension "**Audio Codecs Configure**" the priority is higher than "Allowed Codec" items, "Allowed Codec" items are the default codec setting, if user marks the extension "**Audio Codecs Configure**", then system will use it first, if not system will let the "Allowed Codecs" define what codec can be used in extension.

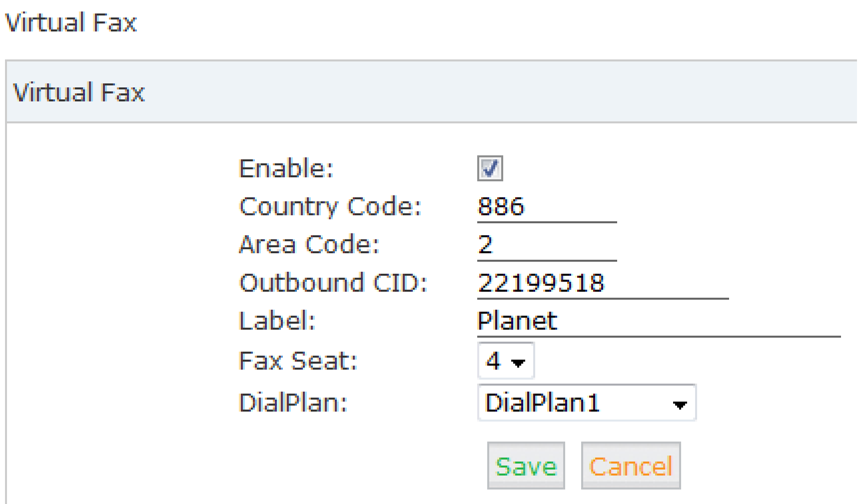
3.5.1.4 IAX2 Settings



Item	Explanation
UDP Port	IAX2 signaling and media port, default is 4569.

Item	Explanation
Bandwidth	Specify bandwidth of low, medium, or high to control which codecs are used in general.
Max Registration/Subscription Time (sec)	Maximum amounts of time that IAX peers can request as a registration expiration interval (in seconds).
Min Registration/Subscription Time (sec)	Minimum amounts of time that IAX peers can request as a registration expiration interval (in seconds).

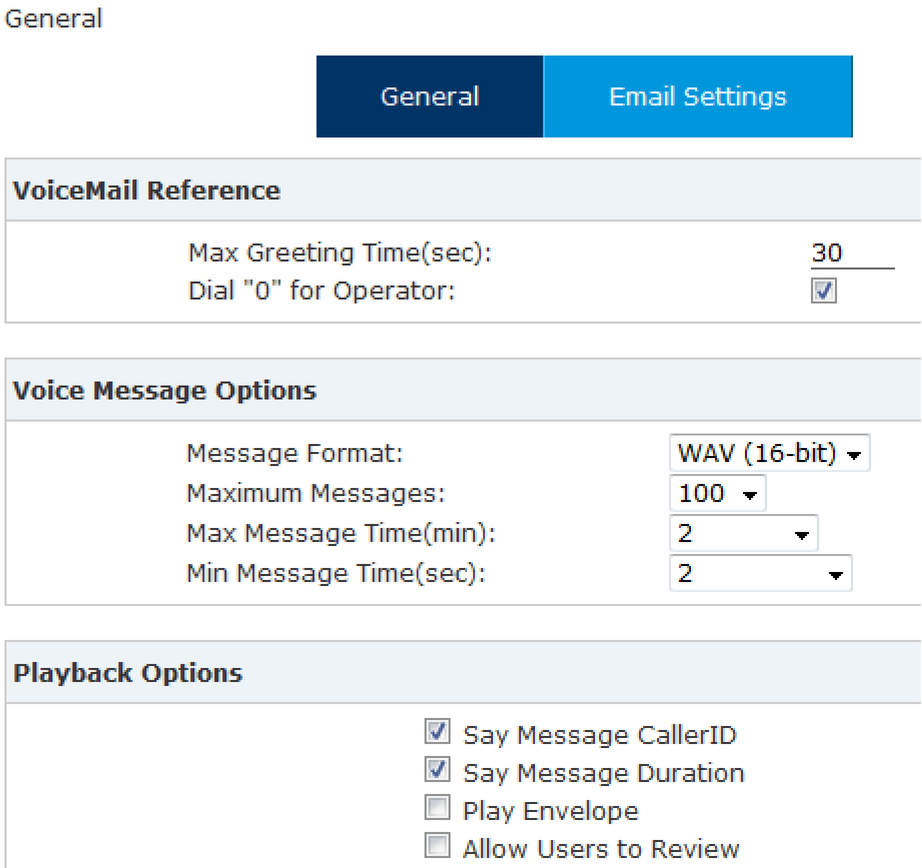
3.5.2 Virtual Fax



Item	Explanation
Enable	Enable the following settings for outbound fax.
Country Code	Enter your country code here. (Optional).
Area Code	Enter your Area Code here. (Optional)
Outbound CID	Only works if the outbound fax goes out through VoIP trunks. The other side receives your fax with this number.
Label	Some custom information to be printed to the header of the fax pages.
Fax sent	Defines how many users can send fax at the same time.
DialPlan	A proper dial plan to send faxes.

3.5.3 Voicemail

Details configuration on Voicemail: Voicemail Reference/Voice Message Options/Playback Options. If you need to send message by mail to your defined mailbox, you must configure SMTP and Email model. Click【Voicemail】to display the dialog as shown below:

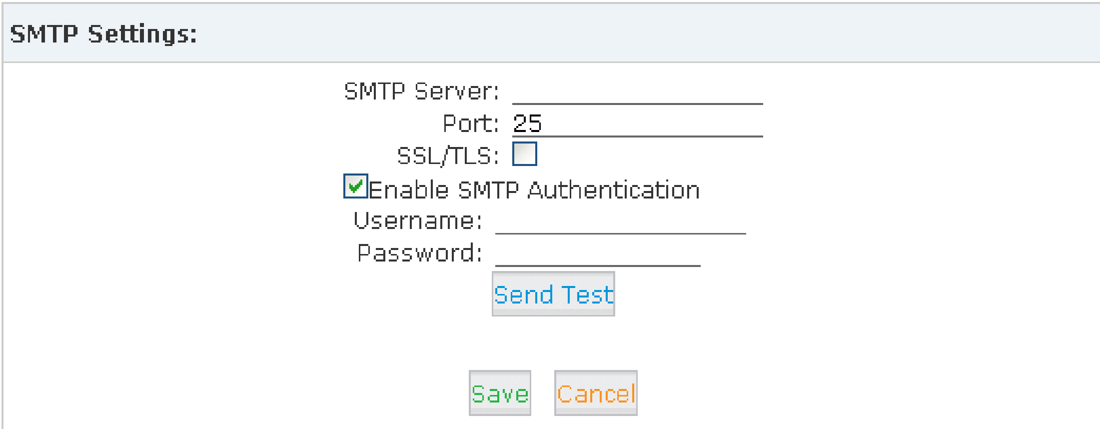


Item	Explanation
Max Greeting Time(sec)	Maximum Greeting Time
Dial "0" for Operator	Dial "0" to cancel the voicemail and forward to Operator.
Message Format	Save the voice message as this format, WAV (16-bit) or Raw GSM.
Maximum Messages	Maximum messages to be allowed to leave.
Max Message Time(min)	Maximum Time for each message to be allowed to leave.
Min Message Time(sec)	Minimum Time for each message. The message will be deleted automatically if the time is less than the minimum message time.
Say Message Caller ID	Checking this option, Caller ID will be played when user login email to receive the voice message.
Say Message Duration	Checking this option, the message duration will be played before playing the voice message.
Play Envelop	Envelop includes date, time and caller ID.

Item	Explanation
Allow Users to Review	Check this option to allow users to review the voice message.

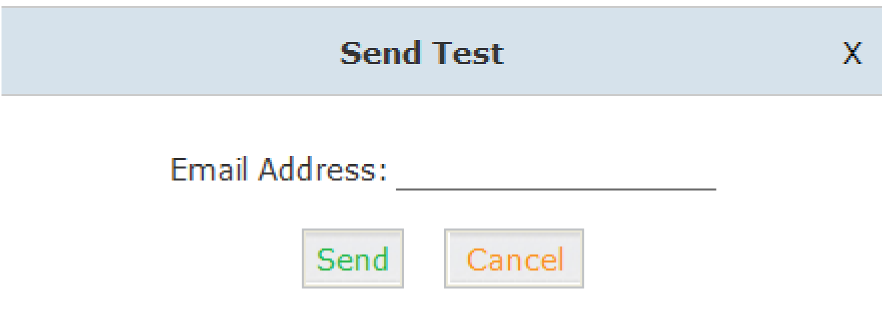
3.5.4 SMTP Setting

SMTP Settings



Item	Explanation
SMTP Server	In order to send e-mail notifications of your voicemail, set the IP address or domain name of a SMTP server that your IP PBX may connect to. e.g. mail.yourcompany.com
Port	The port number the SMTP server runs is generally port 25. If SSL is encrypted, please use port 465 instead.
SSL/TSL	Enable SSL/TLS to send secure messages to server.
Enable SMTP Authentication	If your SMTP server needs Authentication, please enable SMTP Authentication, and configure the following information.
User Name	Input user name of your email box.
Password	Input password of your email box.

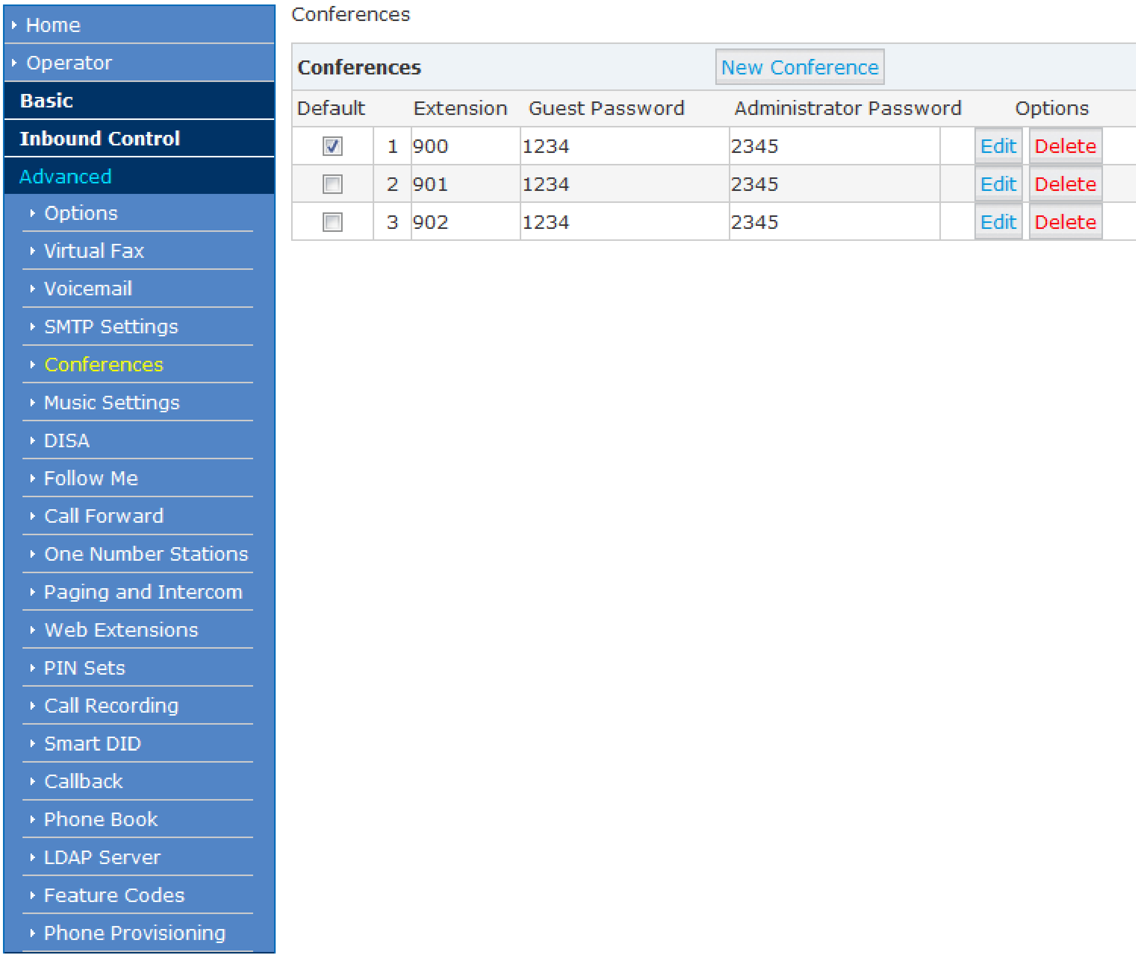
Click【Send Test】after configuration, the following diagram will be displayed to ask you to input the Email for receiving.

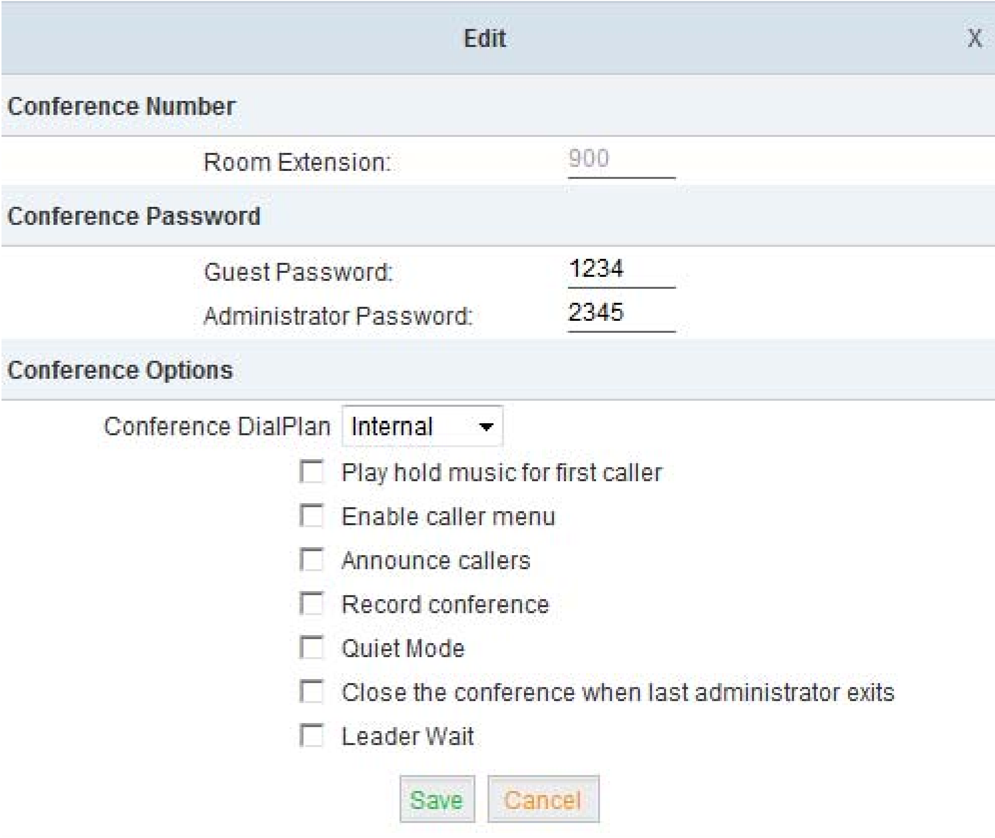


Input the Email and click【Send】to send the test email. Login your Email to check; configuration is successful if you receive the test email; otherwise, it fails. Please check your email settings.

3.5.5 Conference

If you want to create a conference room for some extension users or with external lines, you can input conference room number 900, input conference room password 1234 (Admin's password is 2345), then enter the conference room. IPX-2200 and IPX-2500 support 3 conference rooms. Please configure it on this page【Conference】:



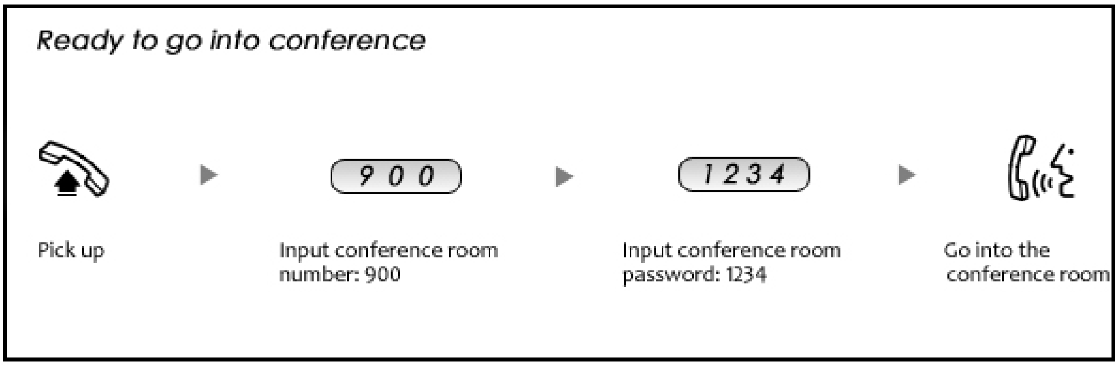


Item	Explanation
Room Extension	By calling this extension number to enter the conference room
Guest Password	If the callers use this password to enter the conference then they are ordinary participants
Administrator Password	If the callers use this password to enter the conference then they are administrators, they have advanced conference menu for example inviting people to participate the conference.
Conference DialPlan	Conference admin can use this dial plan to invite other participants.
Play hold music for first caller	Play the hold music for the first participant in the conference until another participant enters in this conference.
Enable caller menu	Check this option to allow the conference admin to access the conference menu by pressing “*” on the phone.
Announce Callers	Announce all the participants in the room that new participant is coming in.
Record Conference	Record this conference. (Recording format is wav.) The recorded conference can be searched from *Report->Record List->Conference* page.

Item	Explanation
Quiet Mode	If check this option, system will not give any announcement when the participants enter or leave the conference
Close the conference when last administrator exits	If checked this option, the conference will be terminated when the last administrator exits
Leader Wait	Wait until the conference leader(administrator) enters the conference before starting the conference

Please check the following diagram to learn:

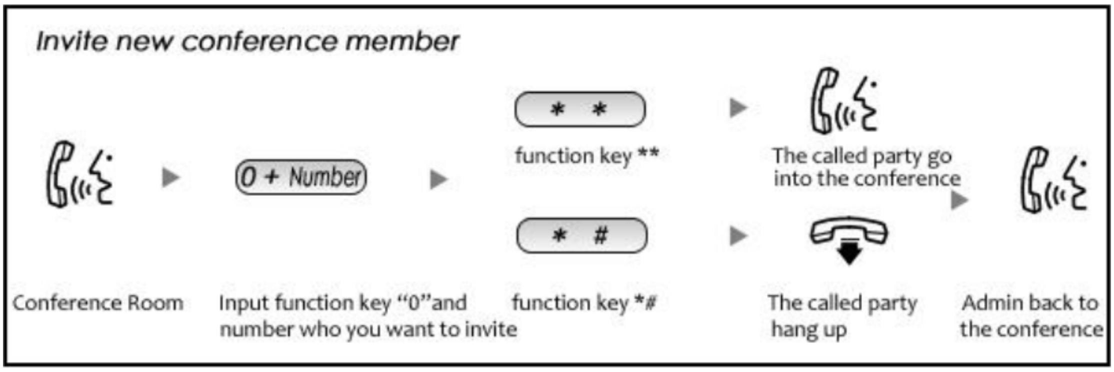
Go to conference:



In the conference, admin can add new participant (extension user or external number) to the conference.

In the conference, the administrator can invite new guest (extension user or external number) to the conference. (Default password for admin is 2345)

Learn how to invite new guest to the conference as the diagram is shown below:

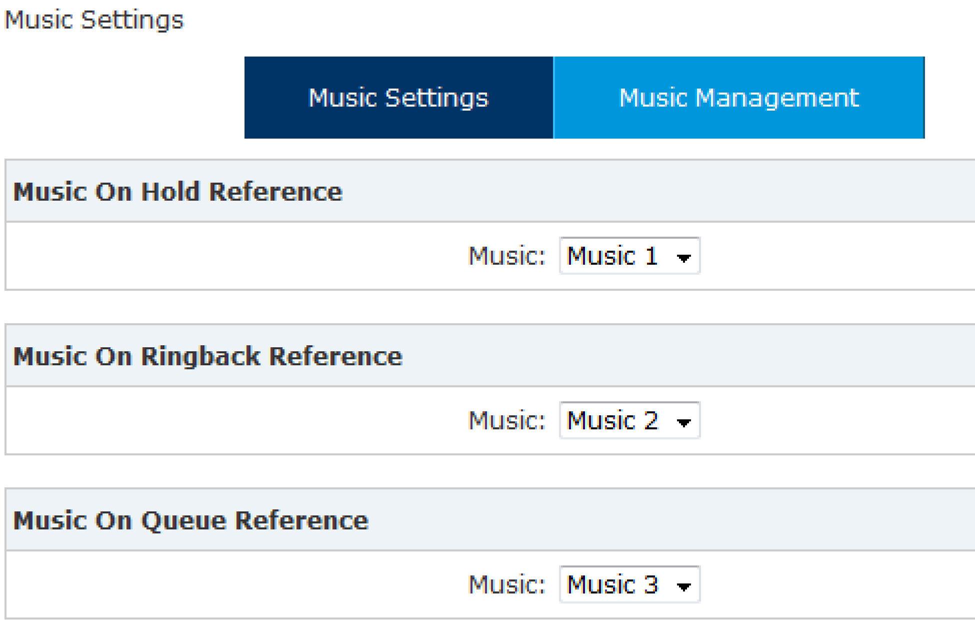


3.5.6 Music Settings

Management for music on hold, music on ring back, music on call queue…

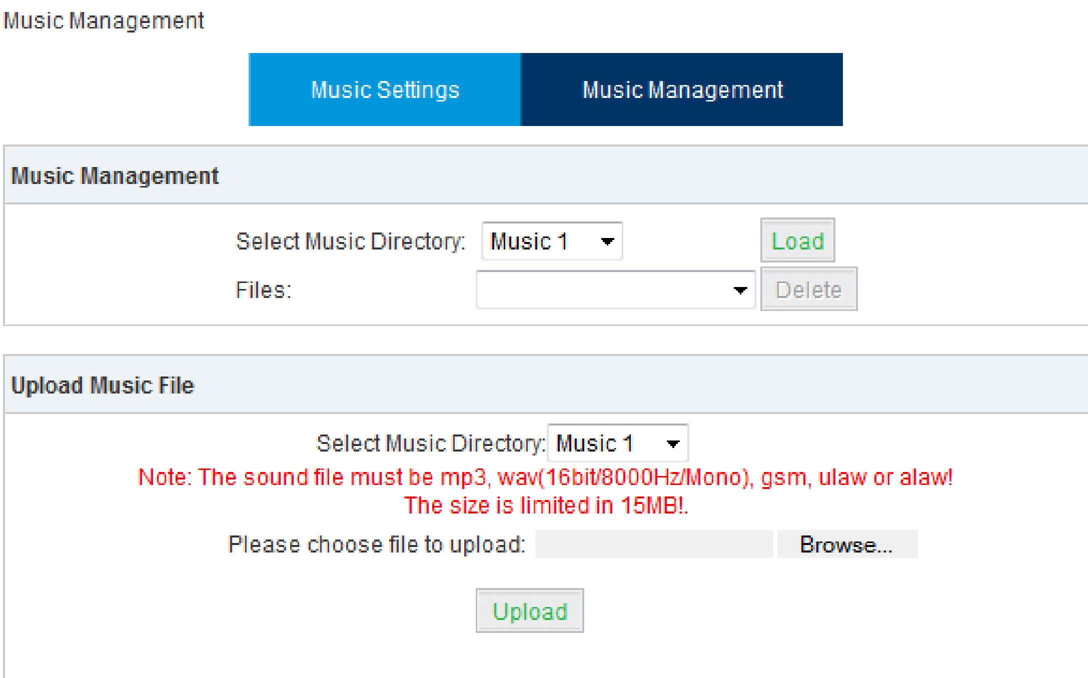
Click【Music Settings】 to display the dialog shown below:

Music Settings:



Please define different music files for different music folders.

Music Management:



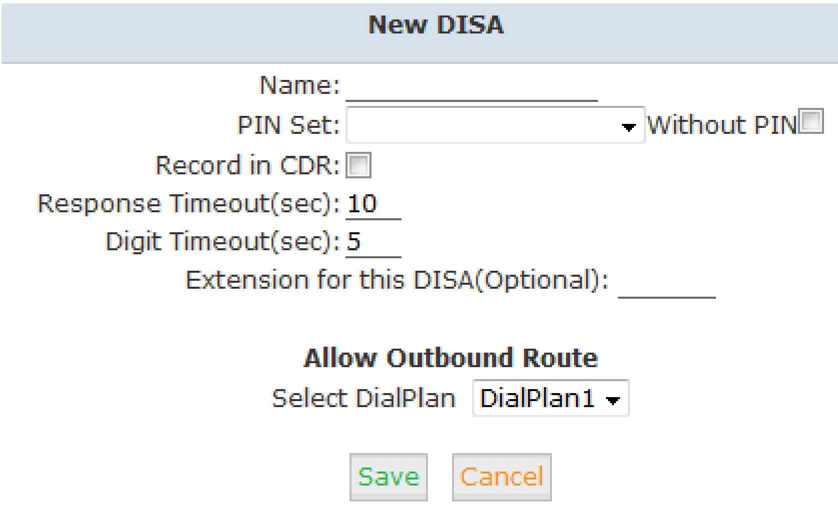
Item	Explanation
Select Music Directory	Load music in the music file.
File	Display music name under the music file. You can delete it.
Select Music Directory	Select the file where you want to save your uploaded music.
Please choose file to upload	Select the music you want to upload. Note: music file must be MP3, WAV (16bit/8000Hz/Mono), GSM, ulaw or alaw, and less than 15MB.

Note	The sound file must be MP3, wav (16bit, 8000Hz, mono), gsm, ulaw and alaw audio file format. The size is limited to **15MB.**

3.5.7 DISA

A trunk call is made to the PBX, and call is made to another trunk through outbound route of the PBX. This trunk can make international calls. You are out of the office and want to contact your customer in a foreign country. Now you can dial DISA number after PIN authentication. You are now connected to your customer, and you can speak to your customer now.

Click【DISA】---【New DISA】to display the dialog as shown below:



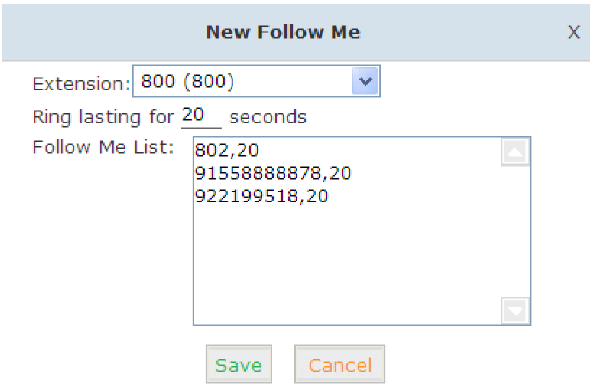
Item	Explanation
Name	Define a name for DISA.
PIN Set	A set of PIN codes to authorize the callers using the system features and facilities.
Without PIN	If enabled, the callers will not be required to enter any PIN code to be able to use the system features can facilities (Not recommended).
Record in CDR	The PIN code that has been used will be stored into call logs

Item	Explanation
	which can be traced on *Report->Call Logs* page.
Response Timeout (sec)	The maximum time for waiting before hanging up if the dialed number is incomplete or invalid. Default is 10 seconds
Digit Timeout (sec)	The maximum interval time between digits when typing extension number is 5 seconds by default.
Extension for this DISA (optional)	If you want to access DISA by dialing an extension, you can define an extension number for this DISA.
Select Dial Plan	Select a dial plan for this DISA so the callers will be able to make outbound phone calls using the trunks on the IP PBX system.

3.5.8 Follow Me

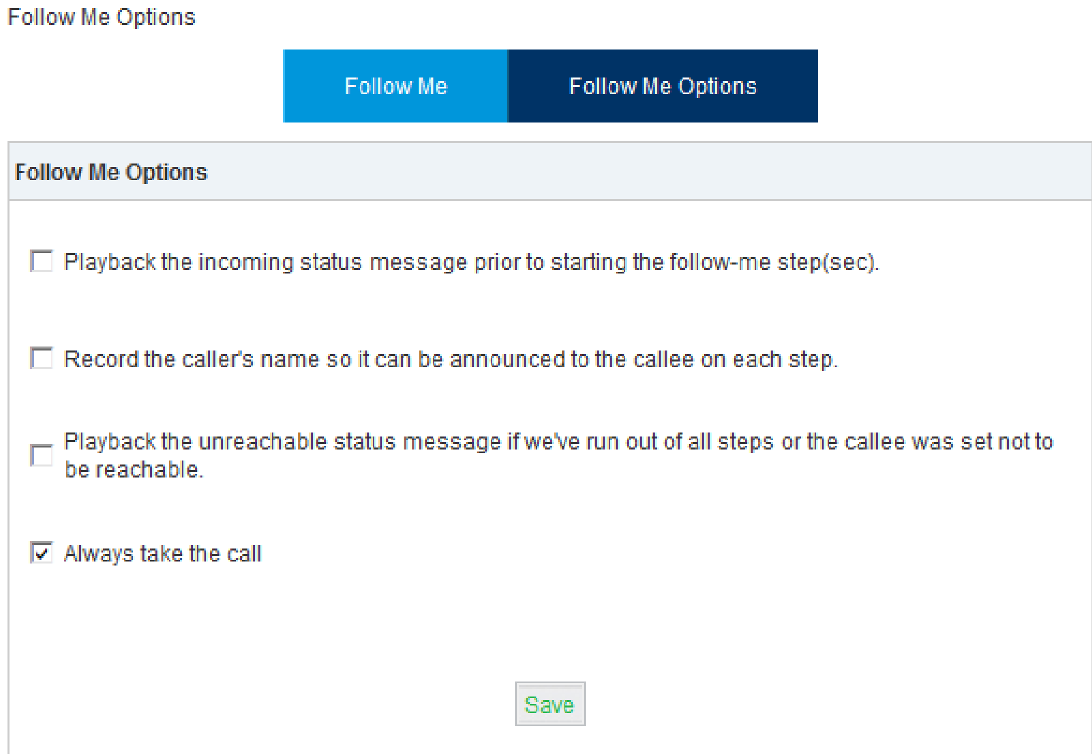
The Follow Me feature allows you to create a more specialized method of routing calls that are sent to a specific extension. Using this module, you can cause a call to an extension to ring several other extensions, or even external phone numbers. So the inbound calls can ring all the numbers which can possibly find you.

Navigate to web menu *Advanced->Follow Me*. Click on "New Follow Me" to configure follow me for an extension.



Item	Explanation
Extension	Select the extension number which will be configured with follow me.
Ring lasting for _20_ seconds	Define how long to ring the extension before the call is forwarded out. By default 20 seconds.
Follow Me List	The list of numbers to forward the calls to. Each line is written with the format "number,time", "number" is one of the number to forward the calls to, "time" defines how long to ring this number, they are separated with a comma without space. The order of

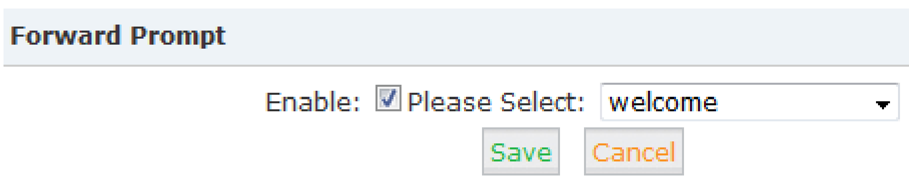
	ringing these numbers are the order you writing in this column.



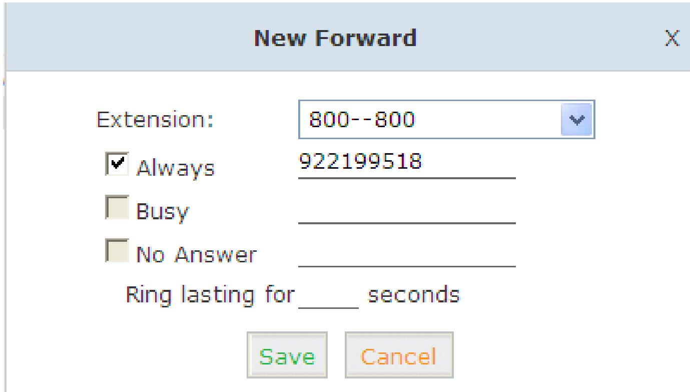
3.5.9 Call Forward

3.5.9.1 Configure From the Web

This feature allows calls to an extension to be automatically forwarded to a specific internal extension or external phone number. Before configuring call forward you can enable the IP PBX system to play a voice prompts before the call was forwarded out. This voice prompts can be recorded or uploaded from the *Inbound Control->IVR Prompts* page. Once the voice prompts file is ready you can navigate to web menu *Advanced->Call Forward.* And enable the system to play back the voice prompts before the incoming call was forward out.



After the voice prompt is set, you can click the “New Forward” button to set call forward for an extension.



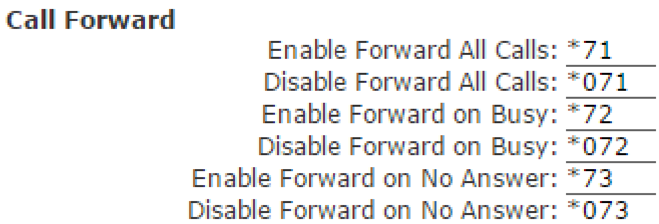
Item	Explanation
Always	Unconditionally forward the incoming calls.
Busy	Forward the incoming calls only if the extension is busy.
No Answer	Forward the incoming call only if the extension didn’t answer.
Ring lasting for ____ seconds	Only is call forward on “No Answer” this option is available to be configured. It defines how long to ring the extension before forwarding.

Note	
	1. If you forward a call to an external phone number please make sure to add a prefix in front of the number if your system requires prefix to dial out. 2. The forward condition “Always” is mutually exclusive to “Busy” and “No Answer”.

3.5.9.2 Configure From the Phone

Navigate to web menu *Advanced->Feature Codes.*

You’ll see feature codes for call forward as follows:



With these feature codes, you can activate or deactivate call forward directly from your phones without the need to configure on the Web GUI. For example, the IP PBX requires prefix 9 to call outbound, and the number you want to forward the calls to is 86547096.

Activate always call forward: Dial *71986547096, press 1 to confirm.

Deactivate always call forward: Dial *071.

Activate call forward on busy: Dial *72986547096, press 1 to confirm.

Deactivate call forward on busy: Dial *072.

Activate call forward no answer: Dial *73986547096, press 1 to confirm.

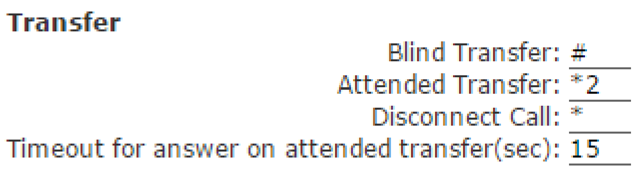
Deactivate call forward no answer: Dial *073.

3.5.9.3 Call Transfer

Call Transfer is used to transfer a call in progress to some other destination. There are two types of call transfer.

- Attended call transfer - Where the call is placed on hold, a call is placed to another party, and a conversation can take place privately before the caller on hold is connected to the new destination. It is also called "Supervised Call Transfer".
- Blind call transfer - Where the call is transferred to the other destination with no intervention (the other destination could ring out and not be answered for instance).

Navigate to web menu *Advanced->Feature Codes*. You'll see the feature code for call transfer as below:

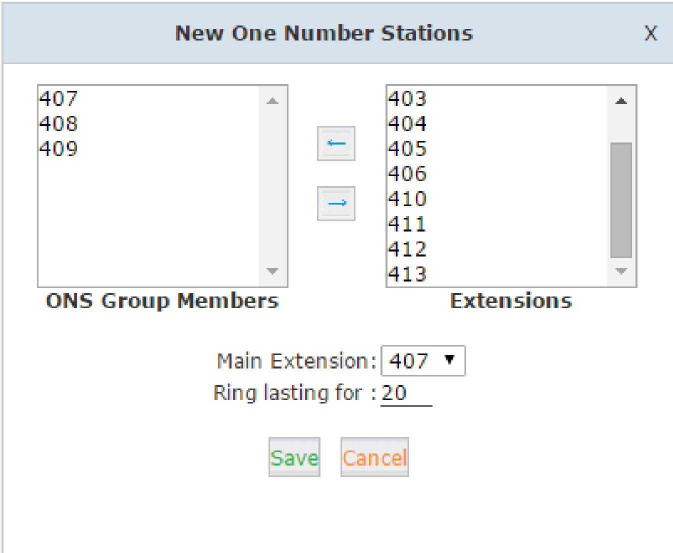


Item	Explanation
Blind Transfer	In a live call, extension user can press # key and the IP PBX system prompts "Transfer", then you enter the number to be transferred to. This call will be transferred instantly and the user can hang up. If the transferred number didn't answer this call it will ring back to the extension user.
Attended Transfer	In a live call, extension user can press *2 and the IPPBX system prompts "Transfer", then you enter the number to be transferred to. After he/she answered your call, you can introduce this call and hang up, and then the call is transferred.
Disconnect Call	In an attended transfer if the other side doesn't want to take the call to be transferred, you can press * to disconnect with him/her and get back to the caller.
Timeout for answer on attended transfer(sec)	In an attended transfer if the third party rings for 15 seconds without answering, the extension user will go back to the caller and the transfer will be terminated.

3.5.10 One Number Stations

One number stations is an innovative IPPBX feature provided by Planet only. With one number stations feature, you can have the same extension number in several different locations.
One number stations feature can put several extension numbers in the same "group", a main number can be selected from the members, when there's an incoming call to the main number it will ring all the member extensions including the main number. Any extension call other extensions will display only the main number.

Navigate to web menu *Advanced->One Number Stations*. Click "New One Number Stations" button to create a one number stations group.



Select the extensions from the "Extensions" column to the "ONS Group Members" column. In the "Main Extension" dropdown list select an extension to be the main extension number. And click on "Save" you'll have a new one number stations group.

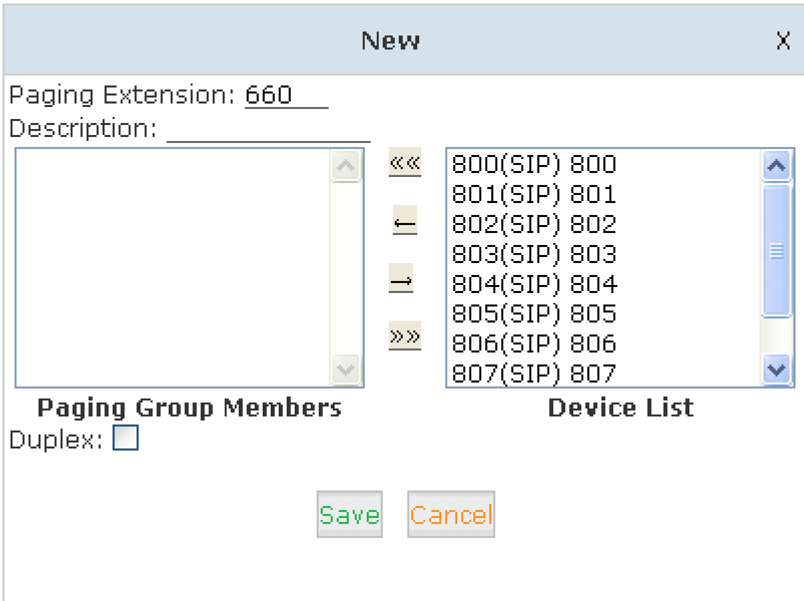
In this case, no matter what extension -- 407, 408 or 409, if they call other extensions, others only see it is extension 407 calling. Others call 407, all these 3 extensions will ring.
As you can see on this page there's a feature code Switch Station available.

This feature code is used to switch extension during a phone call. For example, an inbound call called extension 407, the one number stations member 408 answered this call, you can press *1 from extension 407 or 409 to switch this live call to 407 or 409, 408 will be disconnected.

3.5.11 Paging and Intercom

The Paging and Intercom feature allows you to use your phone system as an intercom system, provided that your endpoints (phone devices) support this functionality. The Paging and Intercom feature allows you to define a number (just like an extension or Ring Group number) that will simultaneously page a group of devices. For example, in a small office, you might define a paging group that allows any user to dial 699, allowing them to page the entire office. You can also use the feature code *50/*51 to page/intercom a single extension, by dialing *50/*51 followed by the extension number..

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



Item	Explanation
Paging Extension	The extension number for this paging group, by calling this extension number you can reach the group members.
Description	Provide a descriptive title for this Page Group.
Paging Group Members	Selected device(s) on this page
Device List	Select Device(s) to page.
Duplex	If enabled the group members can talk to the caller. By calling the paging extension number, all the group member phones will auto answer in speaker mode (requires the IP phones support auto answer feature), the caller can now make a brief announcement to the group members.

Note	For Paging/Intercom function extension (IP phone), enable **Auto Answer**.

3.5.12 Web Extensions

Web Extensions is simply understanding of WebRTC. You can use your web browser to register an extension number to the IP PBX system without any plugins.

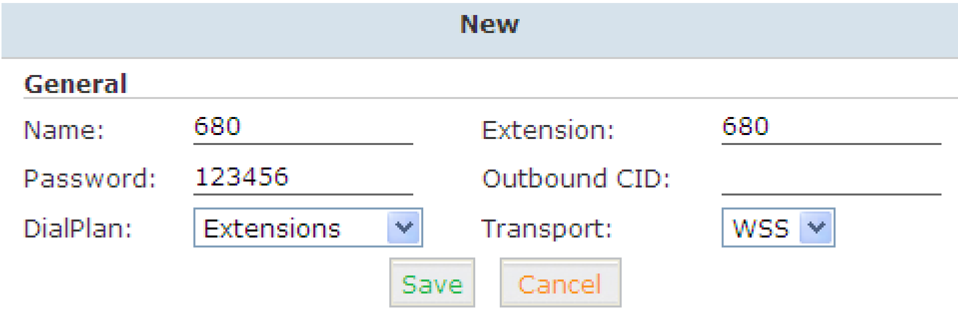
Click on the “New User” button to add a new web extension.

To register the first Web extensions, please follow the steps below:

Step 1:

Create a Web Extension

To create a web extension, navigate to web menu *Advanced->Web Extensions*. Click on the “New User” button to add a new web extension.



Item	Explanation
Name	User name of this web extension.
Extension	Extension number of this web extension.
Password	Password for registration of this web extension.
Outbound CID	Only works if the call was placed out through VoIP trunks.
DialPlan	Defines which type of numbers the web extension can dial.
Transport	WS or WSS
WS	WS (WebSocket) Protocol is an independent TCP-based protocol providing full-duplex communication channels over a single TCP connection. The WebSocket protocol was standardized by the IETF as RFC 6455 in 2011, and the WebSocket API in Web IDL is being standardized by the W3C.
WSS	WSS (WebSockets over SSL/TLS), like HTTPS, WSS is encrypted and we strongly recommend the secure wss:// protocol over the insecure ws:// transport. A variety of attacks against WebSockets are almost impossible if the transport is secured.

Step 2:

Upgrade Web extension patch

As you can see, web extensions use different protocols for signaling and media (WS/WSS) and they are not ordinary SIP/IAX2 extension that can use IP phones or softphones to register so must be treated differently.

Step 3:

Register a Web Extension

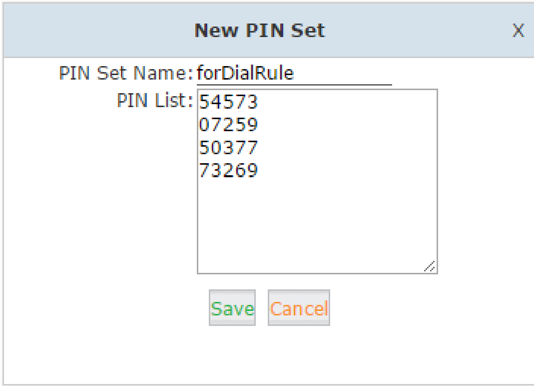
After completing the upgrade process you can access the WebRTC extension register interface. Open your web browser and enter URL https://192.168.1.197/webrtc (192.168.1.197 should be your IP PBX IP address) you will see the web extension register interface. Please complete the register credentials as shown below:

Next, press Enter and the web extension will be registered and is ready for phone calls just like any other standard extension. WebRTC can even be adapted to the enterprise website which can help an enterprise serve their customers with direct voice communication via their website.

3.5.13 PIN Set

Pin sets can be used to secure your IP PBX system phone services. For example outbound dial rules and DISA.

.Click【Advanced】---【PIN Sets】, Click on the “New Pin Set” button to create a collection of PIN codes.

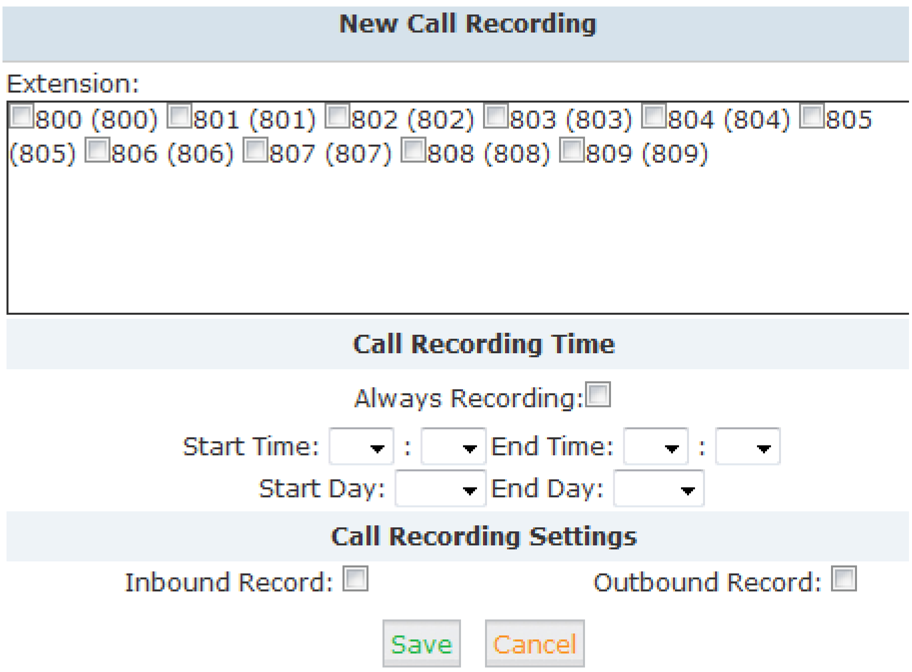


Each line is a PIN code. Press Enter to write down the next PIN code without any symbols.

3.5.14 Call Recording

IPPBX system has built-in ability to record calls. No additional software is required for recording calls. When IP PBX system records a call, both sides of the call are recorded and written out to a file for playback on a computer. Call recording can be used to ensure call quality, or to keep calls for later review. The IP PBX provides the ability to record all of the calls, or to selectively record calls.

Click【Advanced】->【Call Recording】->【New Call Recording】:



Reference:

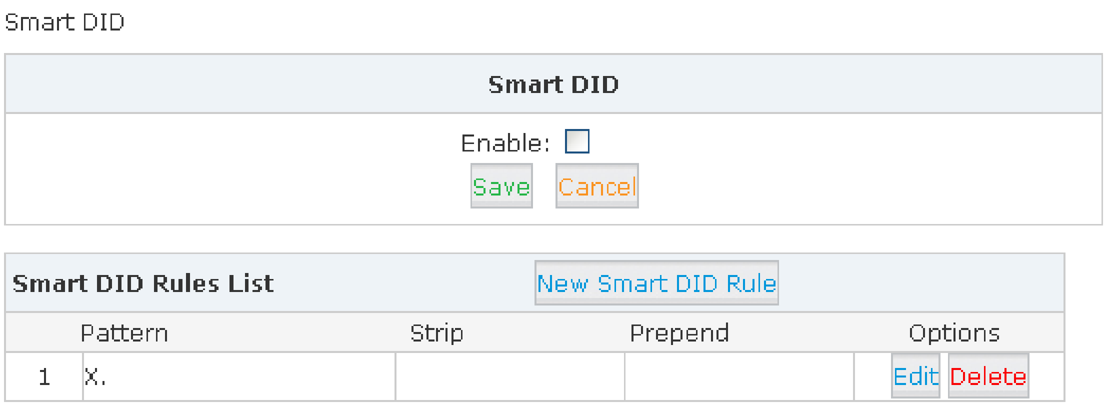
Item	Explanation
Extension	Select the extensions which you want all their calls to be recorded.
Always Recording	If enabled all calls of the above selected extension will be recorded no matter when the calls have been made and received.
Start Time, End Time, Start Day, End Day	If Always Recording is unnecessary you can specify which time durations in a week to record all calls of the above selected extensions.
Inbound Record	Enable to record all inbound calls.
Outbound Record	Enable to record all outbound calls.

Note: The recordings can be searched out on *Report->Record List->Call Recording* page.

3.5.15 Smart DID

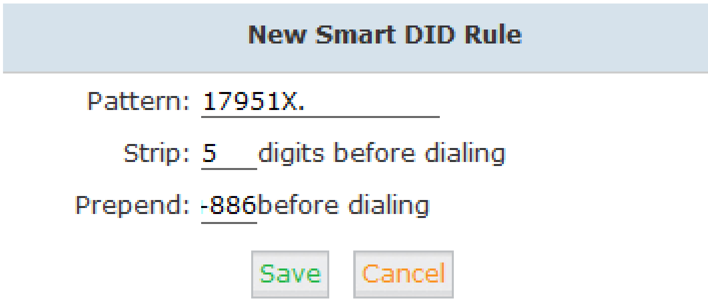
The IPPBX system has the ability to route an inbound call directly to an extension if previously the extension called this number without answering. It is convenient for the called party to make a call back and finds the extension user directly without going through the IVR menu or any other improper call destination.

Click【Advanced】->【Smart DID】:



There's a default Smart DID rule, which enables all outbound calls to be monitored by Smart DID feature. If the call is not answered by the called party, then the called number will be stored into Asterisk database with the extension number which made this call. While the called party is calling back, the IP PBX system can automatically direct this call to the extension number directly.

If you don't want all outbound calls to be monitored by Smart DID, you can modify the existing rule or click "New Smart DID Rule" to add you custom rule/rules. An example is shown below:



Item	Explanation
Pattern	Defines the number format which would be dialed.
Strip	Remove some digits from the front of the dialed number.
Prepend	Prepend some digits in front of the dialed number after being manipulated by the "Strip" option.

The numbers to be dialed will start with prefix 17951 and if they call back, the expected numbers will have +886 in front of them instead of the 5-digit prefix 17951. In such a situation, the outbound and inbound numbers are not the same, you'll need the "Strip" and "Prepend" options to manipulate the dialed numbers to make sure it can match the "same" number when it calls back. If the numbers to be called and the numbers to be received are the same, then you don't have to configure these 2 options. Or you can configure only one of these 2 options, it will all depend on the real applications.

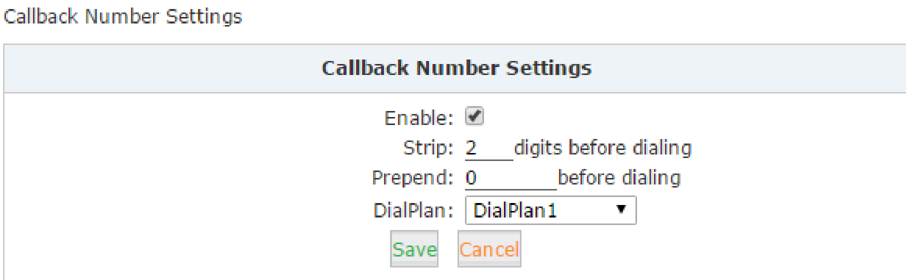
For example the extension user 401 wants to call 86547096, and the carrier requires a prefix 17951 so the rate is much cheaper. The user would dial 1795186547096 to place this call. If the called party missed this call, IPPBX system will store this number +88686547096 with extension number 401 into its database. Later on, if the called party tried to call back, the IPPBX system gets +88686547096 as the caller ID and matches from it database, once successfully matched, this call will be automatically directed to extension 401.

Note	1. The records of Smart DID functionality in the system database will be erased every day at midnight. Which means this is a dynamic effective feature. 2. In the "Pattern" field, patterns that can be used are the same as the patterns used to manipulate dialed number in the dial rules. 3. Due to the mechanism of how asterisk works. For now Smart DID only works

with VoIP trunks but not with FXO or GSM trunks.

3.5.16 Call Back

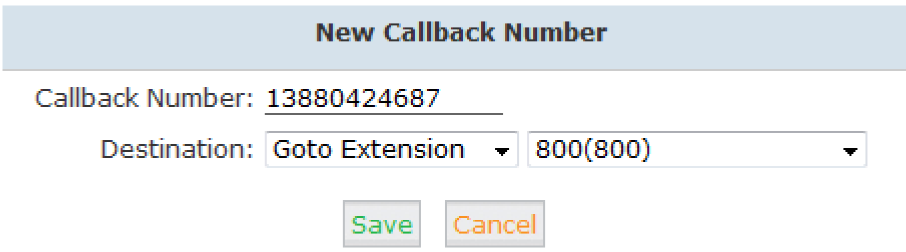
Call Back is a basic service on an IPPBX system for saving on international calls and reducing company phone costs. Ideal for SMB and Corporate business, this PBX feature is designed for users who are making calls from any international destination back to their home country. Please configure it as shown below:



Item	Explanation
Enable	Check the checkbox to enable call back feature.
Strip	The receive caller ID might have some additional digits in front of it and it’s improper for you to call back directly, you can specify here to remove some digits before calling back.
Prepend	After the number had been manipulated by the “Strip” option, you can still add some extra digits in front of it before calling back.
DialPlan	Choose a proper dial plan to make sure the IPPBX system has the permissions for outbound calling.

Click【Advanced】->【Callback】:

At first, enable this function. Select Dial Plan, and define the callback rule (strip digits or prepend prefix). Click【New Callback Number】to add callback number.



Input callback number and define the destination.

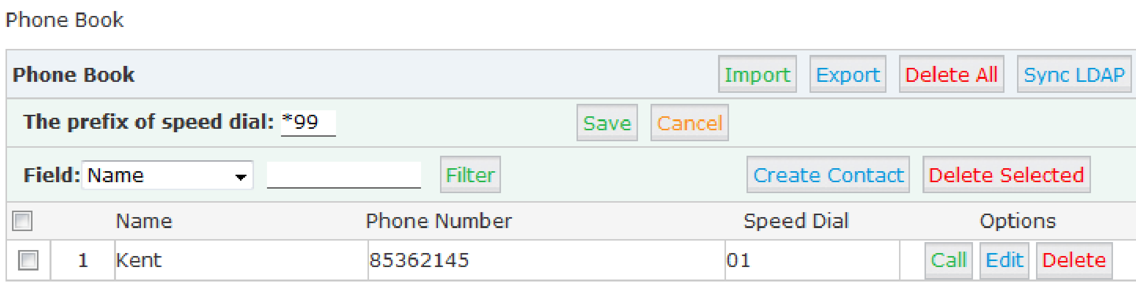
Item	Explanation
Callback Number	The number which calling in to the IPPBX system will be handled by Callback.
Destination	An extension or another call destination which will be used to call the callback number.

Here in this case, if the caller 13880424687 calling in the IPPBX system, IPPBX will disconnect this call and make a call back to this number using extension 800.

3.5.17 Phone Book

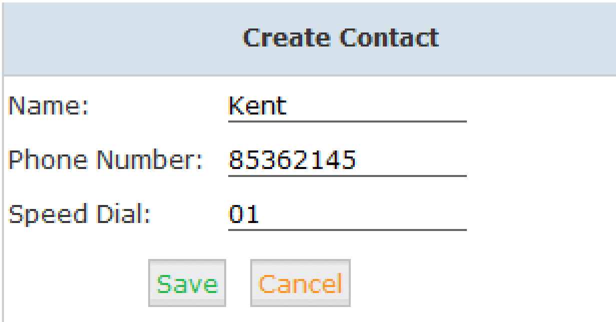
When incoming call matches the number in the phone book, the name of the matched number will be displayed. Please configure it as shown below:

Click【Advanced】->【Phone Book】:



Item	Explanation
Import	You can import contact list from .txt or .csv files.
Export	Export the current contact list as .csv file.
Delete All	Delete all contacts.
Sync LDAP	Synchronize the contacts to the LDAP server.
The prefix for speed dial	Using this feature code with the speed dial code of a contact you can call the contact without knowing the exact number.
Filter	Search contacts by contact name, phone number or speed dial code.
Create Contact	Create a new contact record.
Delete Selected	Delete the selected contacts.
Call	Assign an extension to call this contact.
Edit	Edit the information of this contact.
Delete	Delete this contact.

Click【Create Contact】to see the following diagram:



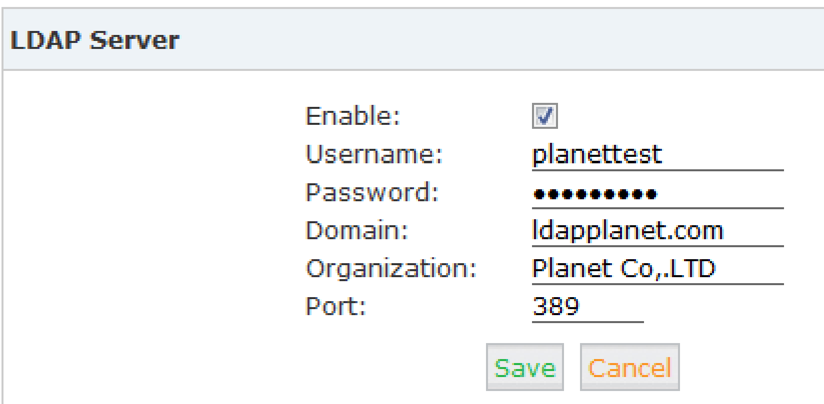
Item	Explanation
Name	Input contact's name. (Letter or figure only).
Phone Number	Input Phone Number of contact.
Speed Dial	Speed dial number which can be used to call this contact from the extensions. After the contacts have been created they will be listed here on this page.

3.5.18 LDAP Server

3.5.18.1 LDAP Server Settings

LDAP (Lightweight Directory Access Protocol) is an open, vendor-neutral, industry standard application protocol for accessing and maintaining distributed directory information services over an IP network. LDAP server has been embedded to IP PBX which is mainly used to centralize manage the phonebook. LDAP server has generated the phonebook based on the created extensions by default.

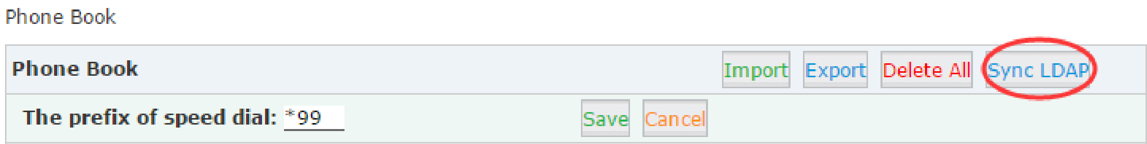
LDAP Server



Item	Explanation
Enable	Enable/Disable LDAP Service.
Username	Define the username of the server administrator (e.g.: manager). This setting will be used on the IP Phone.
Password	Define the password of the server administrator. This setting will be used on the IP Phone.
Domain	Define a domain for the LDAP server (e.g.: ldapdomain.com). This setting will be used on the IP Phone.
Organization	Define an organization to describe the members recorded by LDAP (e.g.: planet.ltd). This setting will be used on the IP Phone.
Port	LDAP service port, default number 389.

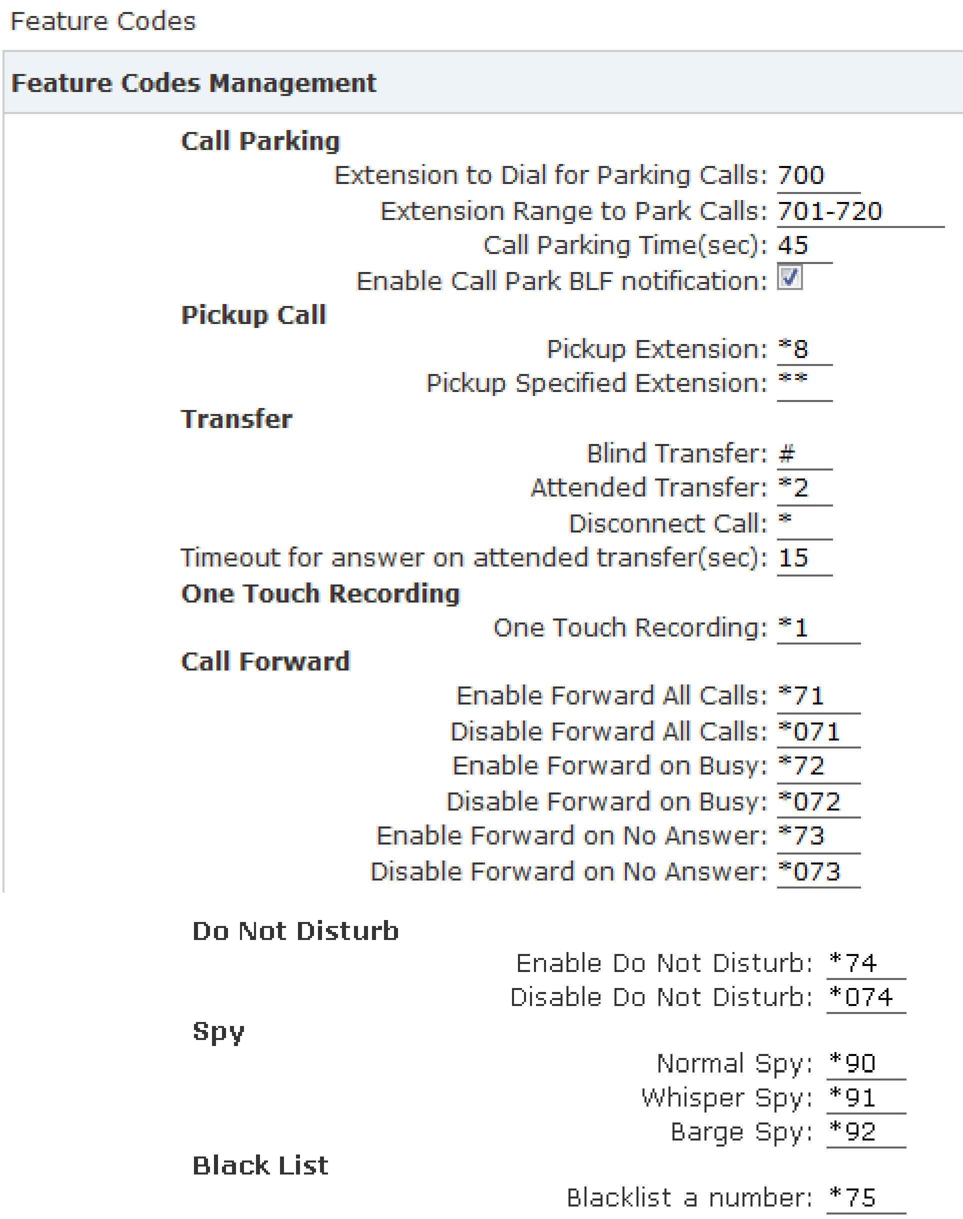
3.5.18.2 Synchronize Contacts with LDAP Server

Navigate to web menu *Advanced->Phone Book*. Click on the "Sync LDAP" button to synchronize contacts with LDAP server.



3.5.19 Feature Codes

Click【Feature Codes】to display the dialog as shown below. You can define relevant parameter.



Item	Explanation
Extension to Dial for Parking Calls	Define an extension for parking calls.
Extension Range to Park Calls	Define the extension range for parking calls. (e.g. 701-720)
Call Parking Time (sec)	Define the time for parking calls. Planet IP PBX will call the extension again if parking is over time.
Pickup Extension	Define an extension for pickup.

Item	Explanation
Pickup Specified Extension	Pick up the specified extension. Default: Dial**+extension number to pick up the specified extension
Blind Transfer	Allow unattended or blind transfers. It works like this: While on a conversation with A, you dial the blind transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number (B's number) and A is put through to B immediately. Your line is off. The caller ID displayed to B is exactly the same as the caller ID presented to you.
Attended Transfer	Allow attended transfer or supervised transfer. It works like this: While on conversation with A, you dial the Attended Transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number (B's number) and talk with B to introduce the call, then you can hang up and A will be connected with B. In case B does not want to answer the call, he/she simply hangs up and you will be back to your original conversation.
Disconnect Call	Disconnect the current transfer call (for Attended transfer).
Timeout for answer on attended transfer（sec）	Set the timeout value
One Touch Recording	Configure the function key for One Touch Recording
Call Forward	Enable/Disable Call Forward and the settings of function keys for different forward modes.
Do Not Disturb	Enable/Disable “Do Not Disturb”
Spy	Configure the function keys for spy modes.
Blacklist	Add/Delete blacklisted number.
Voicemail	Configure the function keys for entering voicemail and check extension voicemail.
Invite Participant	In conference, the administrator can invite people into the conference by dialing “0”. After pressing “0”, you will get dial tone, and you can dial to invite people. After the call is connected, please press ** to direct the people into the conference, or *# to hang up the current call and return to the conference.
Create Conference	During the call, you can dial *0 to forward to the conference with the callee.
Return to conference with participant	In conference, the administrator can dial “0” to invite people into the conference. After pressing “0”, you will get dial tone, and you can dial to invite the participant; when the call is connected, dial “**” to return

Item	Explanation
	to the conference with invited participant.
Return to conference without participant	In conference, the administrator can dial “0” to invite people into the conference. After pressing “0”, you will get dial tone, and you can dial to invite the participant. When the call is connected, you can dial “*#” to hang up and return the conference yourself.
Pause Queue Member Extension	Pause the agent, and the agent cannot receive the call.
Unpause Queue Member Extension	Unpause the agent, and the agent can receive the call.
Others	Function key for Intercom/ Paging/ Directory

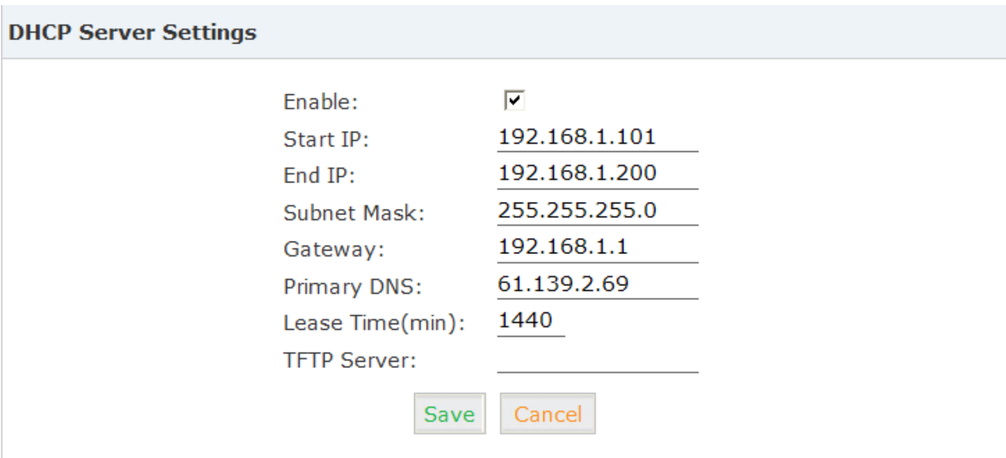
3.5.20 Phone Provision

When you need many IP Phones, please record the MAC, extension number, and user name of each phone according to the format (please take reference of the auto provision script file model for details). Then import the format file. Once the phone is connected to the local network, it will get the extension number and password automatically.

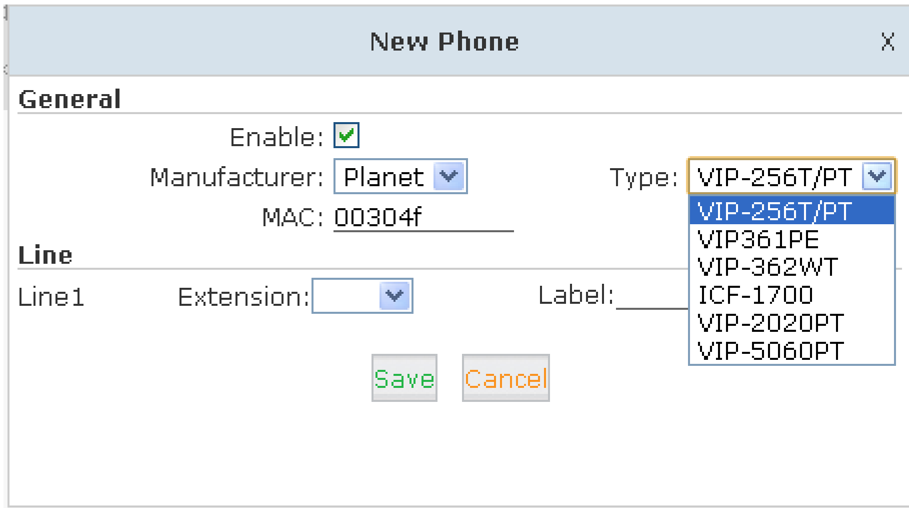
There are two operation methods to fulfill this function. Please see details shown below:

Enable DHCP service

Click【Network Settings】->【DHCP Server】, enable DHCP Server in the dialog as shown below:



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



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

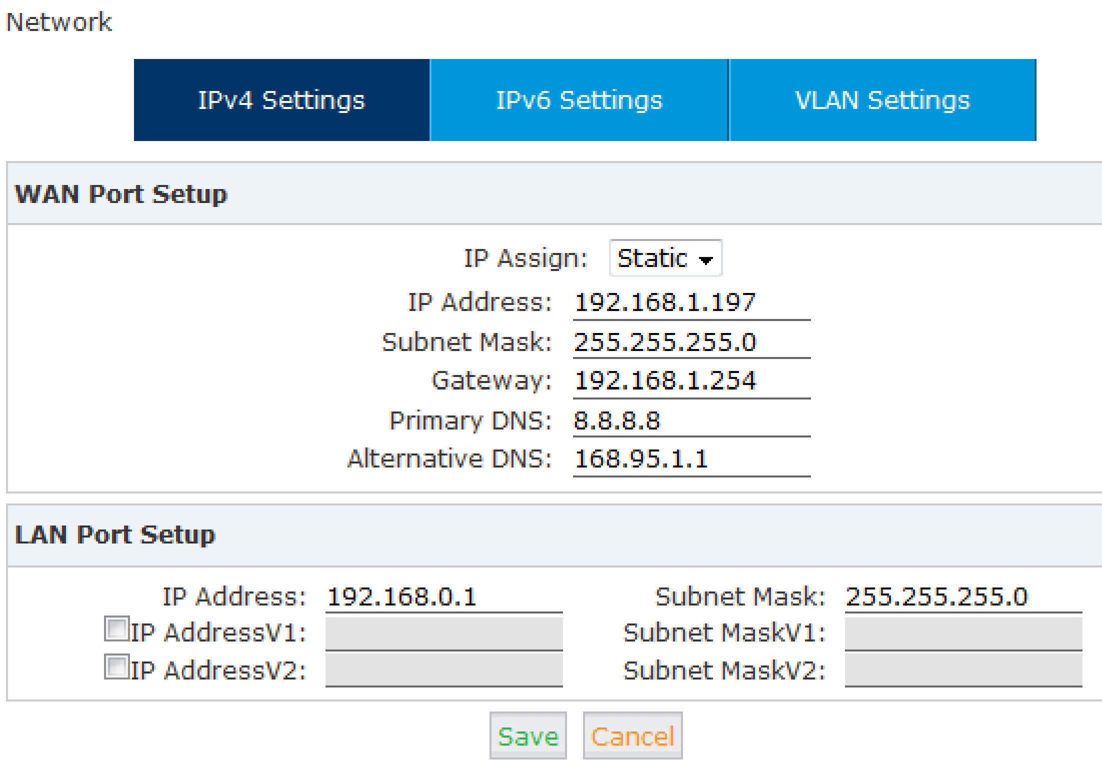
Chapter 4. Network Settings

4.1 Network

IPPBX system supports static IP, DHCP and PPPoE as WAN connection options, and on LAN port only static IP is supported. If you are configuring WAN connection as static IP or DHCP, make sure WAN and LAN IP addresses are not in the same network.

4.1.1 IPv4 Settings

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



Reference

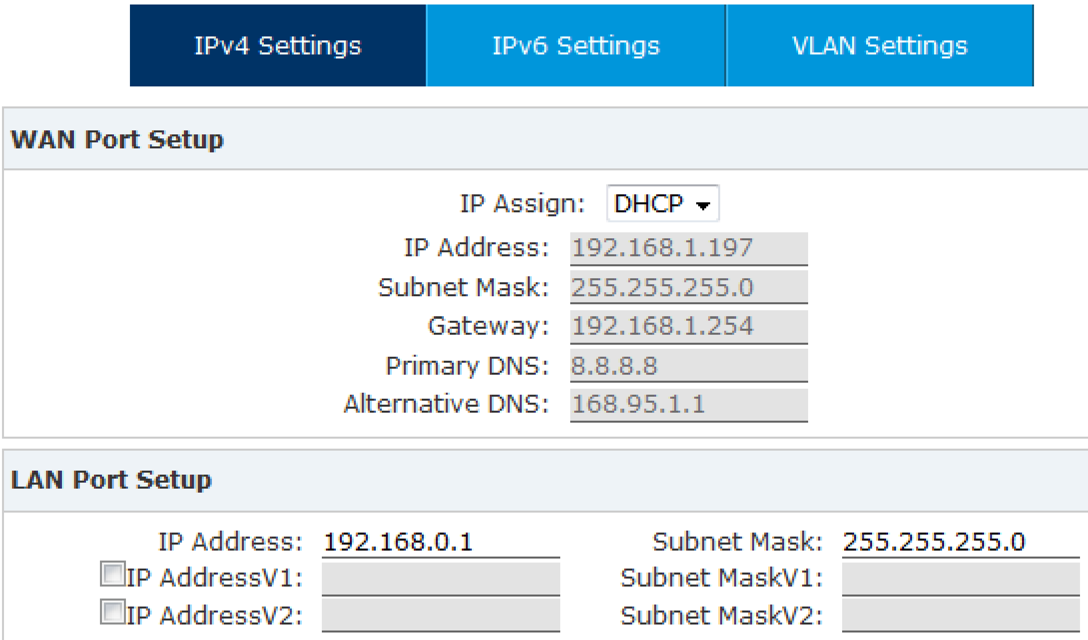
Item	Explanation
IP Assign	Static/ DHCP/PPOE supported.
LAN Interface	Define the LAN interface.

By default IP PBX has been preconfigured with static IP 172.16.0.1 and 192.168.0.1 on WAN and LAN interfaces. If you want to use a static IP, just configure here with the address, netmask, gateway and DNS given to be the ISP or the network admin.

And the LAN interface you can specify 2 additional virtual IP addresses. It can be used to access some other networks from the LAN port.

4.1.1.1 DHCP

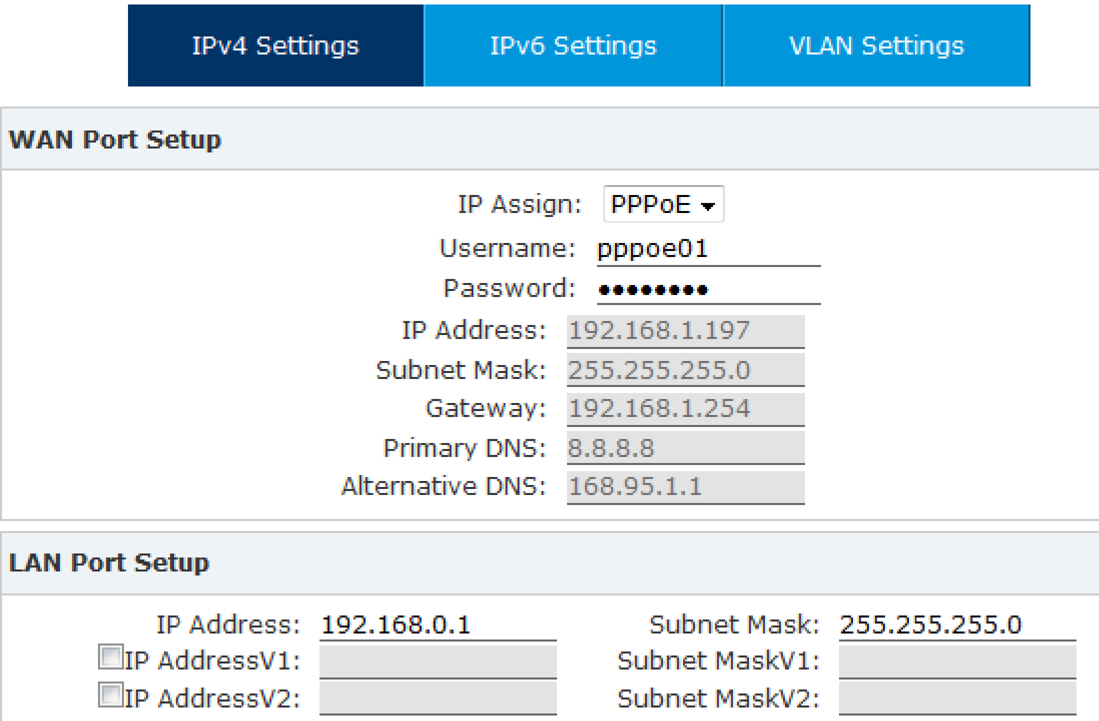
If your Internet connection automatically provides you with a usable IP address, you can select "DHCP" on WAN interface.



If DHCP is selected, WAN interface will not be configurable; it obtains all network parameters from the DHCP server. DHCP should be used cautiously. If all the IP extensions subscribe to the IPPBX system through WAN, you'd better make sure WAN gets a Static DHCP.

4.1.1.2 PPPoE

If PPPoE IPPBX will be connected to the network via ADSL modem by means of Point-to-Point Protocol over Ethernet (PPPoE) dial-up. In such a situation, extensions will subscribe to the IPPBX system through LAN, WAN port can be used for remote extensions.



If PPPoE is set, you just have to specify the username and password given by your ISP and the IPPBX system will dial-up to the ISP and you have Internet access on WAN. LAN port connects to your local network for internal IP extensions to register. If needed, you can change LAN IP to fit your local network.

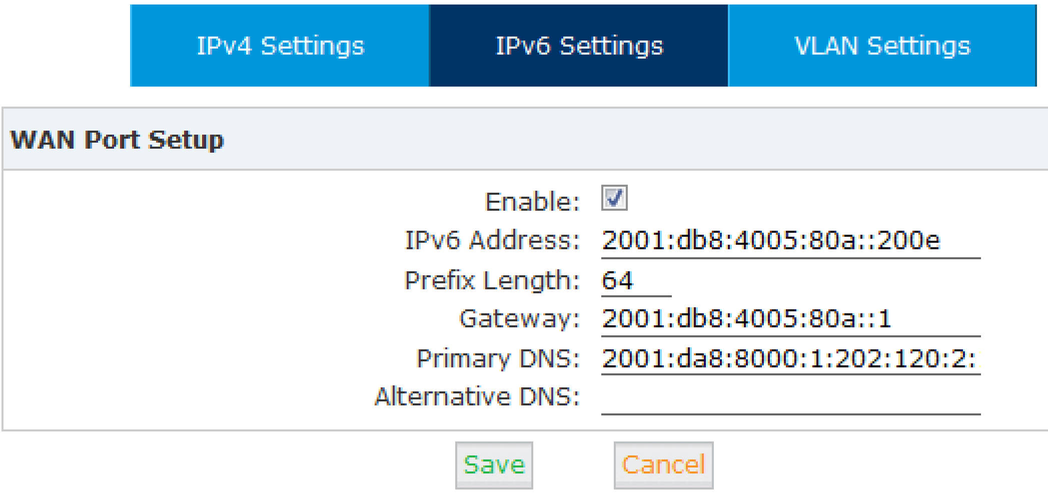
4.1.2 IPv6 Settings

IPv6 (Internet Protocol Version 6) has been in development for nearly two decades. Now the next-generation protocol is ready to replace IPv4 and assume its place as the backbone of the Internet.

Today, major Internet service providers (ISPs), home networking equipment manufacturers, and web companies around the world are permanently enabling IPv6 for their products and services. Many organizations, institutions and universities have deployed their own networks on IPv6.

To be able to deliver VoIP calls over IPv6 (SIP over IPv6), you can configure IP PBX system with IPv6 addresses to be able to deploy it in your IPv6 network infrastructure.

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



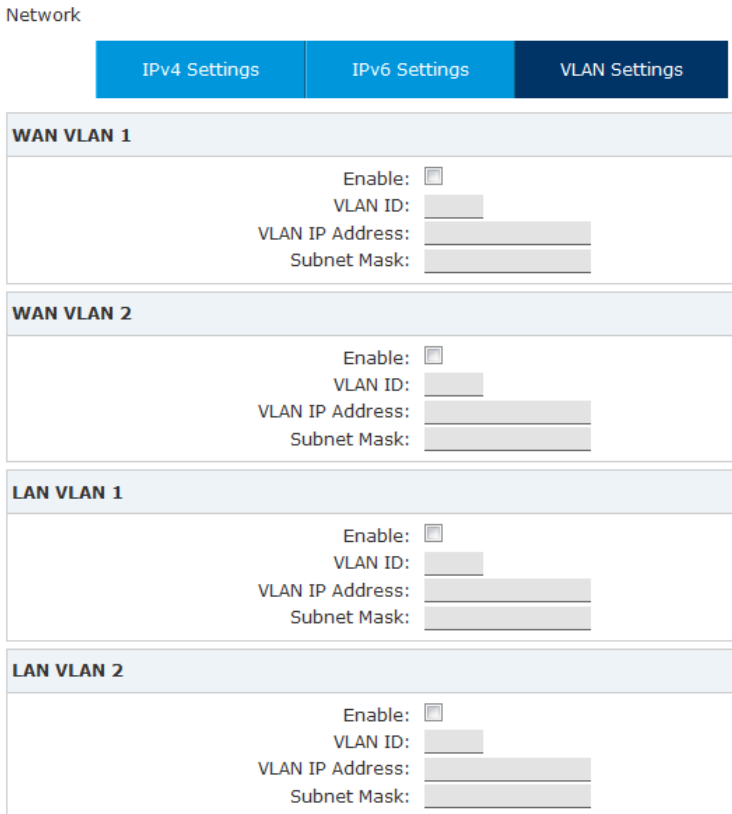
IPv6 Reference:

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

4.1.3 VLAN Settings

With a layer-3 switch you can configure VLAN on IP PBX system to divide the VoIP and data traffic. Voice VLAN can keep the phones working even when the data network is congested. You can see here on this page. You are able to configure 4 VLANs, 2 for each WAN or LAN port.

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



VLAN Reference:

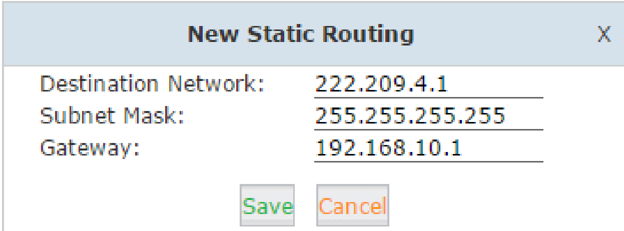
Item	Explanation
Enable	Enable VLAN to define the VLAN address and VLAN ID.

Note	Make sure VLAN IPs for VLAN1 and VLAN2 of WAN and LAN interfaces are in several different network segments.

4.2 Static Routing

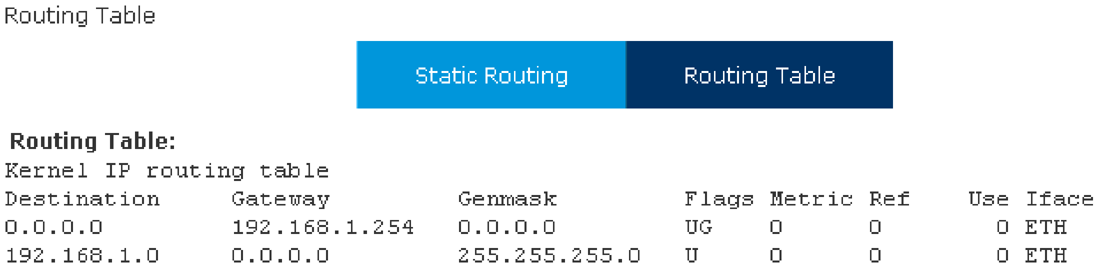
Static Routing is a form of routing that occurs when a router uses a manually-configured routing entry, rather than information from a dynamic routing protocol to forward traffic.

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



Item	Explanation
Destination	Set the IP address of destination host or network address. E.g.222.209.4.1, 192.168.10.0.
Subnet Mask	Set subnet mask of the destination network.
Gateway	Define the gateway accessing the destination network.

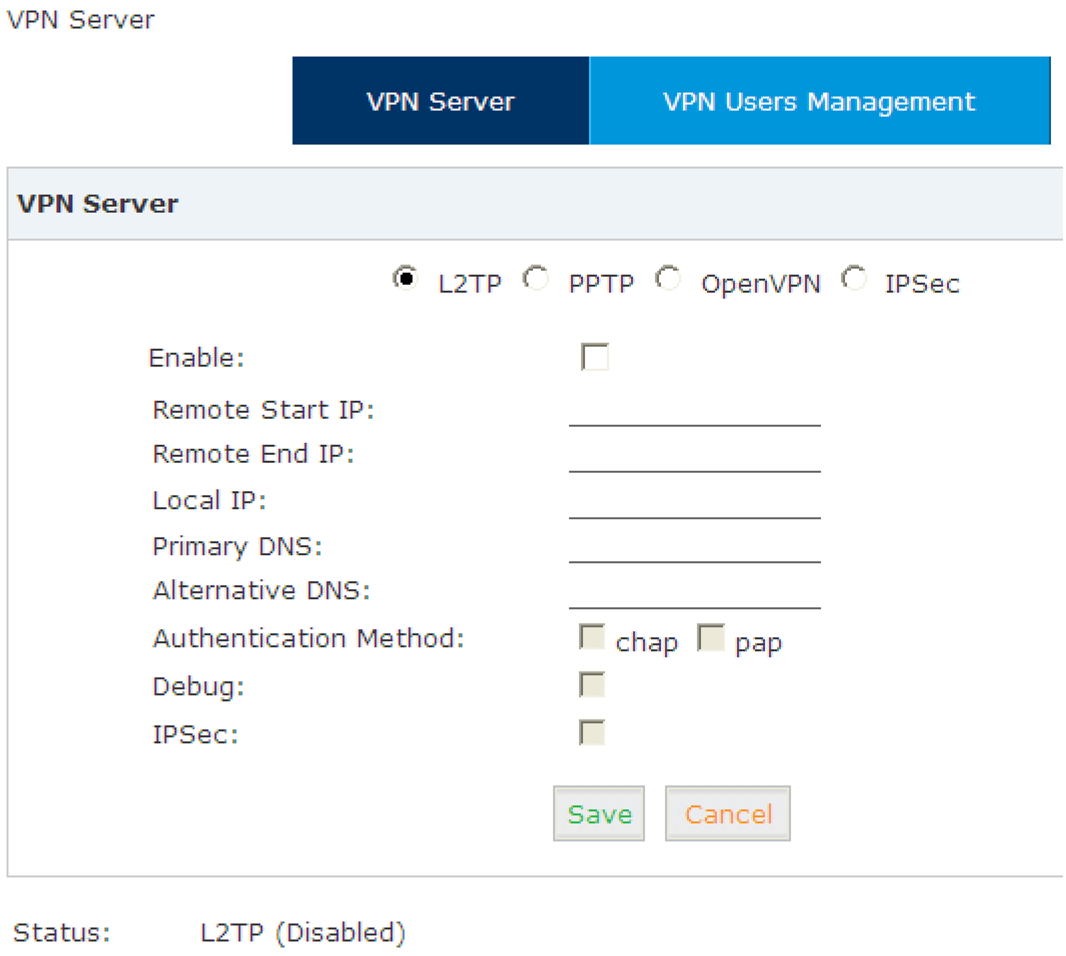
Click【Network Settings】->【Static Routing】->【Routing Table】, and the current routing information will be displayed below:



4.3 VPN Server

VPN (Virtual Private Network) is mostly used for setting up long-distance and/or secured network connections. While it's been used on IP PBX, all the phone calls sending and receiving are encrypted so it secures your remote offices/extensions' phone services. Built-in VPN Server on Planet IP PBX series is an easy way to set up such secured connectivity between other Planet series IP PBXs or IP phones. You don't need to build a dedicated VPN server or buy a VPN router. This is also a workaround to avoid a firewall issue when configuring remote VoIP client as SIP protocol is notoriously to pass through a firewall due to its random numbers to establish connection.

The IP PBX supports four kinds of VPN variety: L2TP/PPTP/OpenVPN/IPSec. Click【Network Settings】->【VPN Server】:



4.3.1 L2TP VPN

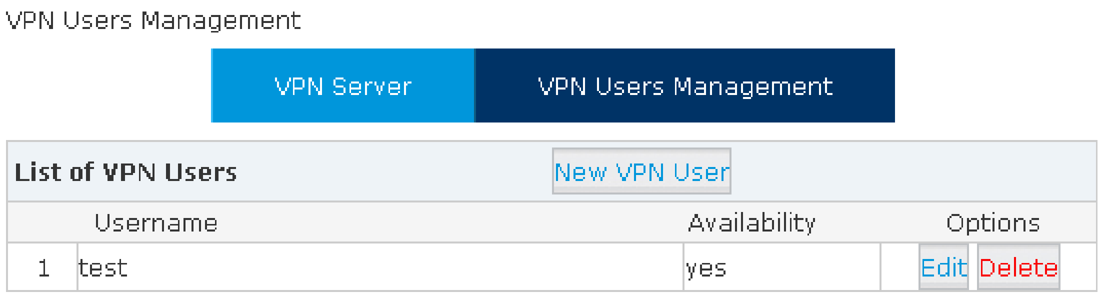
Reference:

Item	Explanation
Enable	Tick the checkbox to enable L2TP VPN server.
Tick the checkbox to enable L2TP VPN server.	L2TP VPN remote network IP range, between start IP and end IP there must be less than 10 available IP addresses.
Local IP	L2TP VPN local server IP address.
Primary DNS	Primary DNS for VPN connection.
Alternate DNS	Alternative DNS for VPN connection.
Authentication Method	Select the authentication method: chap or pap. pap: Password Authenticate Protocol PAP works like a standard login procedure; it uses static user name and password to authenticate the remote system. chap: Challenge Handshake Authentication Protocol CHAP takes a more sophisticated and secure approach to authentication by creating a unique challenge phrase (a randomly generated string) for each authentication.
Debug	Tick to enable debug for L2TP VPN connection, debug info will be

Item	Explanation
	written into system logs.
IPSec	Enable IPSec encryption for L2TP VPN server.
IPSec Local IP	IP PBX WAN IP which can access Internet.
IPSec Password	Define a password for IPSec VPN client to authenticate.

Note	If the IP PBX system is behind NAT, you need to open ports 500, 4500 and 1701 on the router/firewall.

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



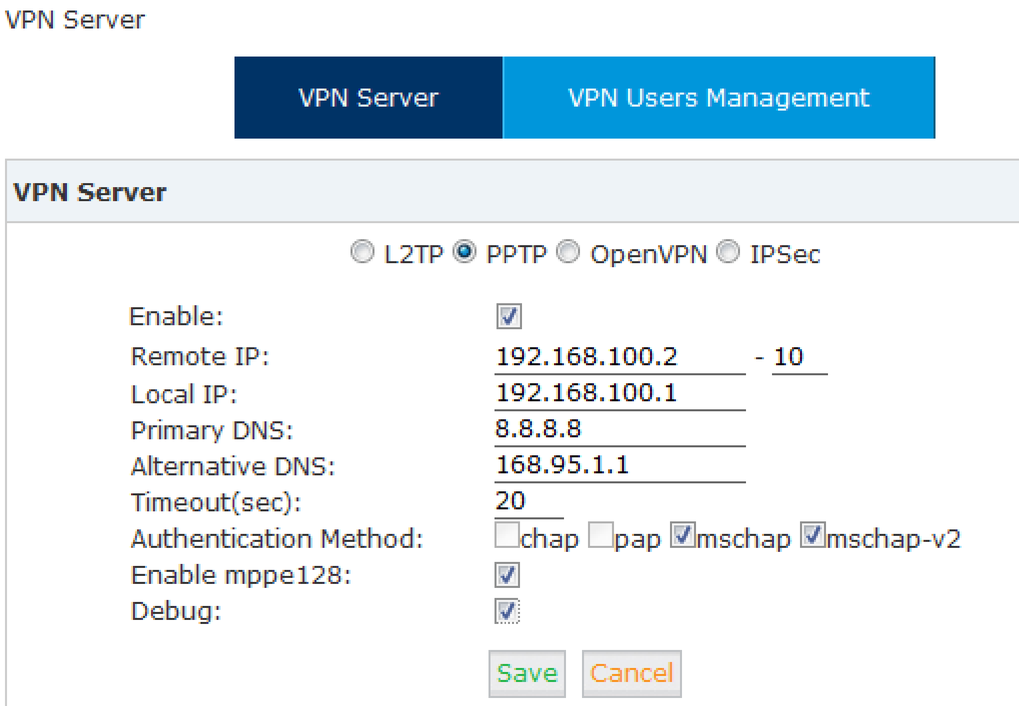
This page is used for management of VPN user name and password.

4.3.2 PPTP VPN

The Point-to-Point Tunneling Protocol (PPTP) uses a control channel over TCP and a GRE tunnel operating to encapsulate PPP packets. The intended use of this protocol is to provide security levels and remote access levels comparable with typical VPN products.

4.3.2.1 PPTP VPN Server

Navigate to web menu *Network Settings->VPN Server*. Check the radio button of PPTP to configure PPTP VPN server.



Item	Explanation
Enable	Tick the checkbox to enable PPTP VPN server.
Remote IP	PPTP VPN remote network IP range, between start IP and end IP there must be less than 10 available IP addresses.
Local IP	PPTP VPN local server IP address.
Primary DNS	Primary DNS for VPN connection.
Alternative DNS	Secondary DNS for VPN connection.
Timeout (sec)	Session timeout for PPTP tunnels.
Authentication Method	Choose method/methods for the authentication of the VPN clients. ● **chap**: Challenge Handshake Authentication Protocol CHAP takes a more sophisticated and secure approach to authentication by creating a unique challenge phrase (a randomly generated string) for each authentication. ● **pap**: Password Authenticate Protocol PAP works like a standard login procedure; it uses static user name and password to authenticate the remote system. ● **mschap**: MS-CHAP is the Microsoft version of the Challenge-Handshake Authentication Protocol.

Item	Explanation
	● **mschap-v2**: Microsoft Challenge Handshake Authentication Protocol version 2 (MS-CHAP v2), it provides stronger security for remote access connections.
Enable mppe128	Microsoft Point-to-Point Encryption (MPPE) encrypts data in Point-to-Point Protocol (PPP)-based dial-up connections or Point-to-Point Tunneling Protocol (PPTP) virtual private network (VPN) connections with 128-bit key.
Debug	Tick to enable debug for PPTP VPN connection, debug info will be written into system logs.

For the VPN client to connect you'll need to create a VPN user account. Click the "VPN User Management" tab and click the "New VPN User" button to add a VPN user account.

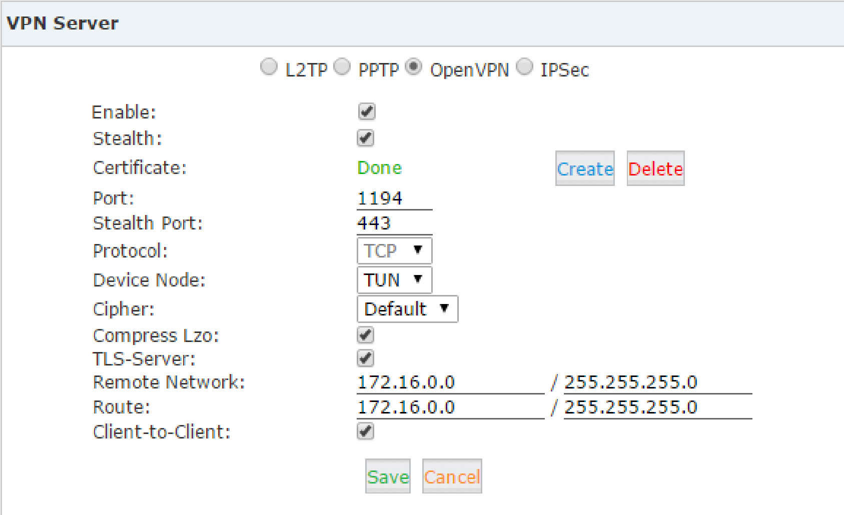
Note	If the IPPBX system is behind NAT, you need to open ports 1723 on the router/firewall.

4.3.3 OpenVPN

OpenVPN is an open-source software application that implements virtual private network (VPN) techniques for creating secure point-to-point or site-to-site connections in routed or bridged configurations and remote access facilities. It uses a custom security protocol [3] that utilizes SSL/TLS for key exchange. It is capable of traversing network address translators (NATs) and firewalls. It was written by James Yonan and is published under the GNU General Public License (GPL).

OpenVPN allows peers to authenticate each other using a pre-shared secret key, certificates, or username/password. When used in a multiclient-server configuration, it allows the server to release an authentication certificate for every client, using signature and Certificate authority. It uses the OpenSSL encryption library extensively, as well as the SSLv3/TLSv1 protocol, and contains many security and control features.

Check the radio button of OpenVPN to configure OpenVPN server.



Item	Explanation
Enable	Tick to enable OpenVPN server
Stealth	Some deep packet inspection firewalls might not allow OpenVPN traffic, stealth SSL tunneling can disguises your OpenVPN traffic under the HTTPS traffic which is often seen as HTTPS traffic by the DPI.
Certificate	Certificate is one of the client authentication methods of OpenVPN.
Port	OpenVPN service port, default is 1194.
Stealth Port	Stealth service port, default is 443.
Protocol	You can choose from UDP or TCP. As stealth requires TCP only so if with stealth enabled, this options is not configurable and will use TCP by default.
Device Node	TUN or TAP; A TAP device is a virtual Ethernet adapter, while a TUN device is a virtual point-to-point IP link.
Cipher	Cipher (or cypher) is an algorithm for performing encryption or decryption.
Compress LZO	LZO is an efficient data compression library which is suitable for data de-/compression in real time.
TLS-Server	TLS is an excellent choice for the authentication and key exchange mechanism of OpenVPN.
Remote Network	OpenVPN remote network.
Route	The route entries adjust the local routing table, telling it which network to route over the VPN.
Client-to-Client	Client-to-Client can enable the intercommunication between clients.

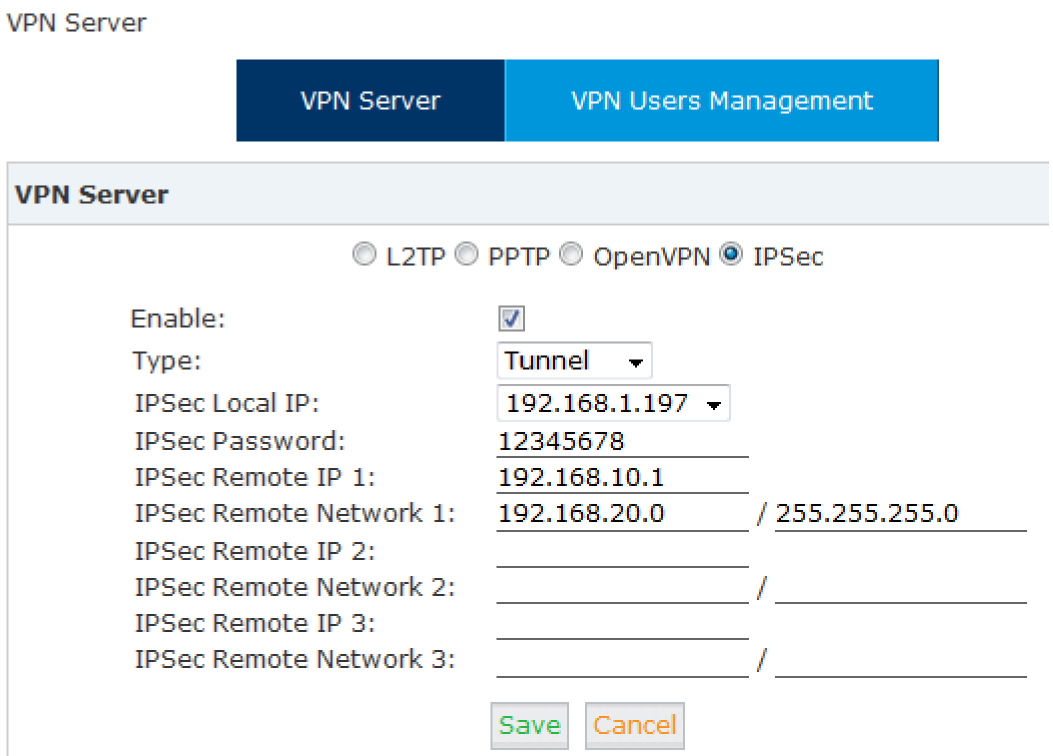
4.3.4 IPSec VPN

Internet Protocol Security (IPsec) is a protocol suite for secure Internet Protocol (IP) communications by authenticating and encrypting each IP packet of a communication session. IPSec can be configured to operate in two different modes, Tunnel and Transport mode. Use of each mode depends on the requirements and implementation of IPSec.

4.3.4.1 IPSec VPN Server (Tunnel mode)

Tunnel mode is used to encrypt all traffic between secure IPSec Gateways, for example two IP PBX's, each acts as an IPSec Gateway for the hosts/IP phones behind it. The WAN ports will be used to connect to each other to establish IPSec VPN connection; the PCs or IP phones on the LAN ports can communicate with each other on both sides via secured IPSec tunnel.

Check the IPSec radio button to configure IPSec VPN server.



Item	Explanation
Enable	Tick the checkbox to enable IPSec VPN server.
Type	Default Tunnel mode.
IPSec Local IP	IP PBX WAN IP, which can be used to connect to the client network.
IPSec Password	Define a password for authentication of the IPSec client.
IPSec Remote IP	IPSec VPN client IP. The client uses this IP to connect to IPSec server.

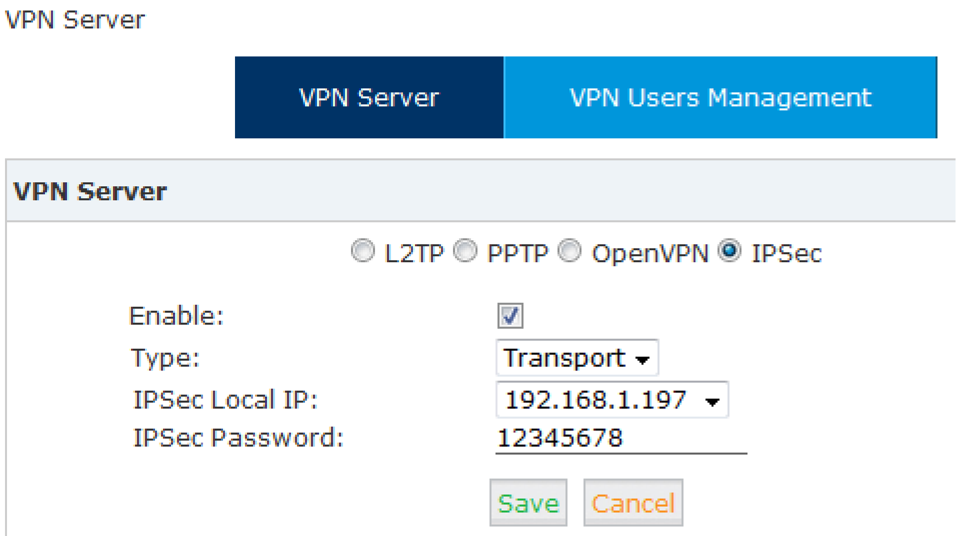
Item	Explanation
IPSec Remote Network	Specify the IPSec VPN client LAN network address.

Note	
	1. If the IPPBX is behind NAT, ports 500 and 4500 need to be opened on the router/firewall. 2. If the IPPBX connects to Internet via PPPoE, then IPSec Local IP needs to be the IP address assigned by PPPoE. 3. IPSec VPN server can connect 3 IPSec clients.

4.3.4.2 IPSec VPN server (Transport mode)

IPSec Transport mode is used for end-to-end communications, NAT traversal is not supported with the transport mode. So if two IP PBX's are connected via IPSec transport mode, IPSec only encrypts the communication service ports, not like Tunnel mode which encrypts the whole LAN subnet.

Check the IPSec radio button.



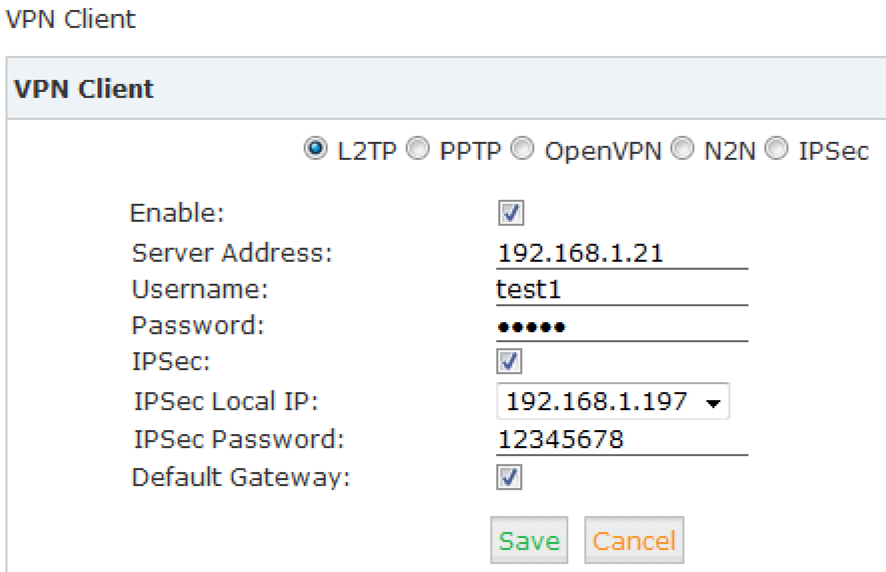
Item	Explanation
Enable	Tick the checkbox to enable IPSec VPN server.
Type	Select Transport mode.
IPSec Local IP	IPPBX WAN IP.(Same as configuring Tunnel mode)
IPSec Password	Define a password for authentication of the IPSec client.

4.4 VPN Client

Planet IP PBX supports four kinds of VPN Clients: L2TP, PPTP, OpenVPN and N2N.

Click【Network Settings】->【VPN Client】:

4.4.1 L2TP VPN Client

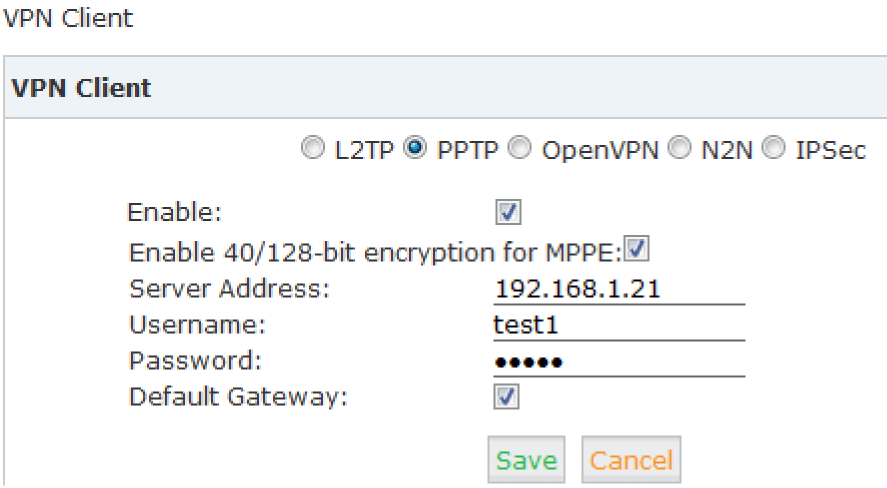


Reference:

Item	Explanation
Enable	Tick to enable L2TP VPN client
Server Address	L2TP server public IP.
Username	L2TP VPN user name given by the VPN server.
Password	L2TP VPN user password given by the VPN server.
IPSec	Enable IPSec support.
IPSec Local IP	IPPBX WAN IP which can access Internet.
IPSec Password	Accordingly as the password specified on the server.
Default Gateway	All traffic goes through the L2TP VPN connection.

4.4.2 PPTP VPN Client

On the branch office site, check the radio button of PPTP to configure PPTP VPN client.

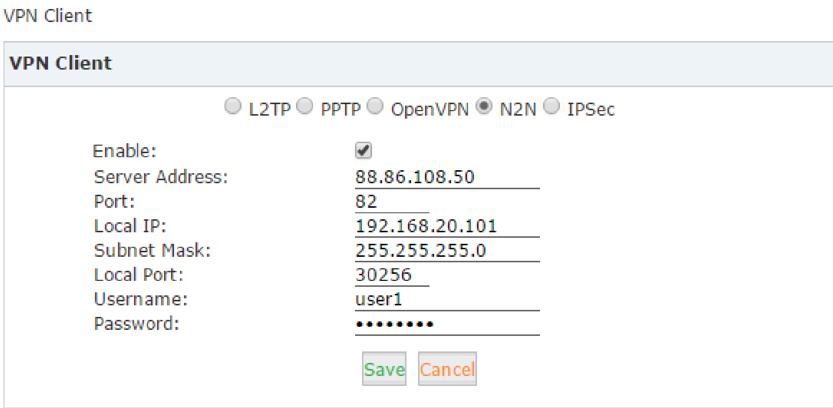


Item	Explanation
Enable	Tick to enable PPTP VPN client.
Enable 40/148-bit encryption for MPPE	Tick to enable 40-bit key (standard) or 128-bit key (strong) MPPE encryption schemes.
Server Address	PPTP VPN server public IP.
Username	PPTP VPN user name given by the VPN server.
Password	PPTP VPN user password given by the VPN server.
Default Gateway	All traffic goes through the L2TP VPN connection.

4.4.3 N2N VPN Client

N2N is an open source Layer 2 over Layer 3 VPN application which utilizes a peer-to-peer architecture for network membership and routing.

On IP PBX system we support N2N VPN client. Check the radio button of N2N VPN and configure the client info.

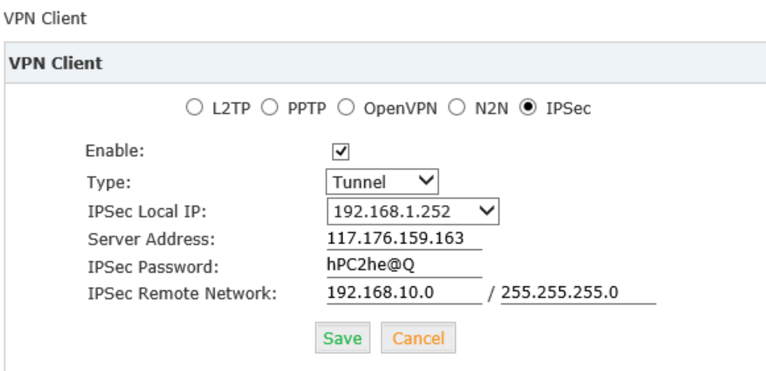


Item	Explanation
Enable	Tick this checkbox to enable N2N VPN client
Server Address	N2N server(supernode) IP address.
Port	N2N service port number. 82 by default.
Local IP	VPN local IP.
Subnet Mask	Netmask of the VPN network.
Local Port	N2N local service port.
Username/Password	Used for the N2N server to authorize the connection.

4.4.4 IPSec VPN Client (Tunnel mode)

On the remote site, open the web GUI of another Planet IPPBX system and navigate to web menu *Network Settings->VPN Client*.

On the VPN Client page, choose IPSec and tick "Enable" option to enable IPSec client.



Item	Explanation
Enable	Tick the checkbox to enable IPSec client.
Type	Accordingly as the IPSec server.

Item	Explanation
IPSec Local IP	WAN port IP which can connect to the IPSec server.
Server Address	Specify the IPSec server IP.
IPSec Password	Specify the IPSec VPN password defined previously on the server.
IPSec Remote Network	The IPSec VPN server LAN network address.

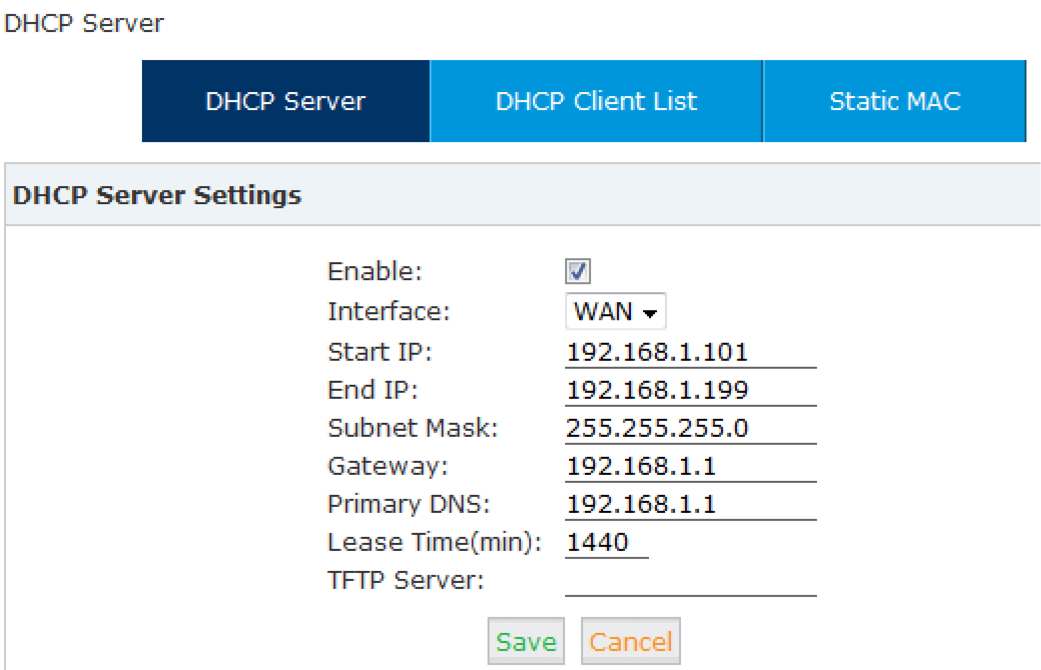
4.5 DHCP server

DHCP (Dynamic Host Configuration Protocol) is a standardized network protocol used on Internet Protocol (IP) networks for dynamically distributing network configuration parameters, such as IP addresses for interfaces and services.

With DHCP, computers/IP phones request IP addresses and networking parameters automatically from IP PBX WAN/LAN port; it saves a lot of time for administrator to configure these settings manually.

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

4.5.1 DHCP Service



Item	Explanation
Enable	Enable DHCP service.

Item	Explanation
Interface	Choose the network port to implement DHCP service.
Start IP, End IP	Specify the DHCP IP address pool.
Subnet Mask	Netmask to be assigned to the client devices.
Gateway	Gateway address to be assigned to the client devices
Primary DNS	DNS to be assigned to the client devices.
Lease Time(min)	DHCP server leases an address to a new device for a period of time. When the lease expires, the DHCP server might assign the IP address to a different device. Default value is 1440 minutes.
TFTP Server	Point out the TFTP server address which may be used to auto provision the IP phones.

4.5.2 DHCP Client List

You'll have all the devices that are getting IP address from the IP PBX system.

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

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

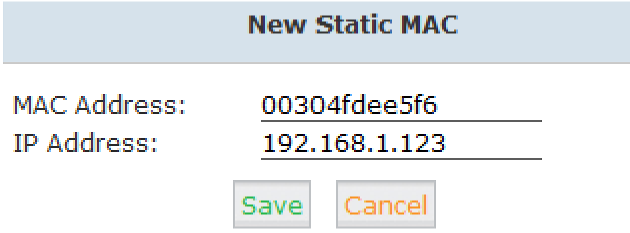
This page is used to display DHCP Client address and related information.

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

4.5.3 Static MAC

Static MAC is a useful feature which makes the DHCP service on IP PBX always assigns the same IP address to a specific computer or IP phone on your LAN. To be more specifically, the DHCP service assigns this static IP to a unique MAC address assigned to each NIC on your LAN.

Click "New Static MAC" to add a record to the IP PBX system.

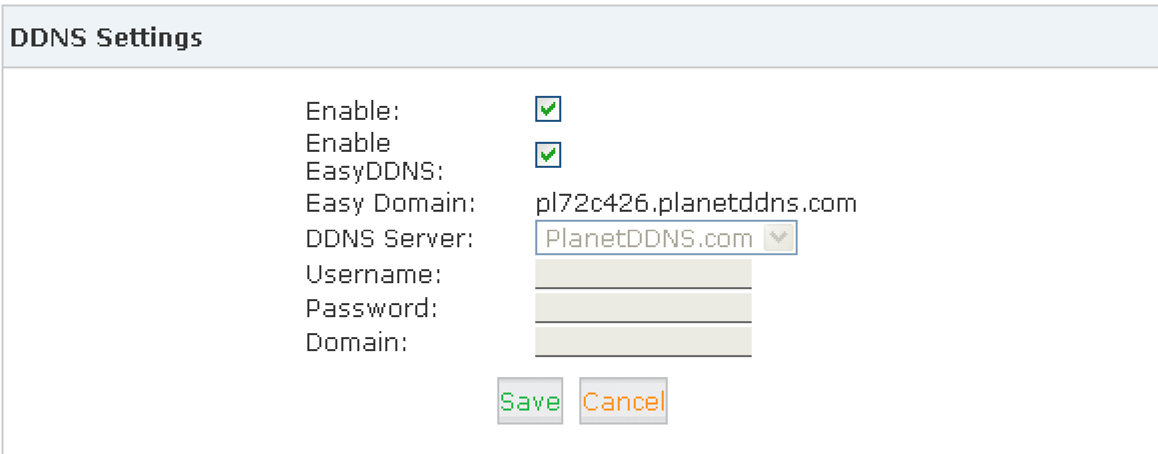


4.6 DDNS Settings

Unlike DNS that only works with static IP addresses, DDNS (Dynamic Domain Name Server) is designed to also support dynamic IP addresses, such as those assigned by a DHCP server. Built-in DDNS feature on IP PBX system only needs a simply signs up with a Dynamic DNS provider, with the domain name they gave which maps your IP address on the Internet, you can access IP PBX and also other services within your LAN via the domain name without getting to know Dynamic public IP.

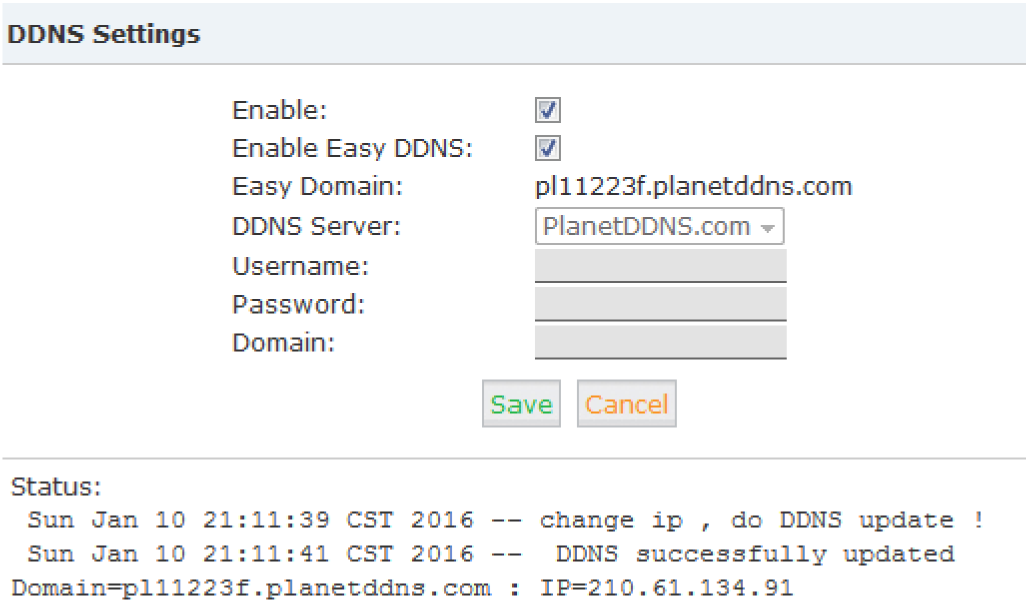
After setting DDNS, IP PBX phone services can be accessed from remote site via the domain name which DDNS provider gave. Also remote management is possible even without a static public IP.

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

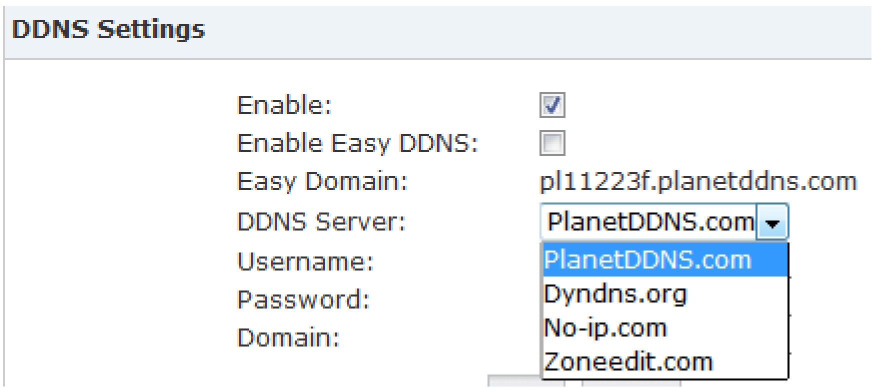


Item	Explanation
Enable	Tick to enable DDNS service
DDNS Server	Select the DDNS service provider which you subscribed the DDNS service.
Username	Username you subscribed to the service provider.
Password	Password you used to sign up to the service provider.

Item	Explanation
Domain	Your domain name.



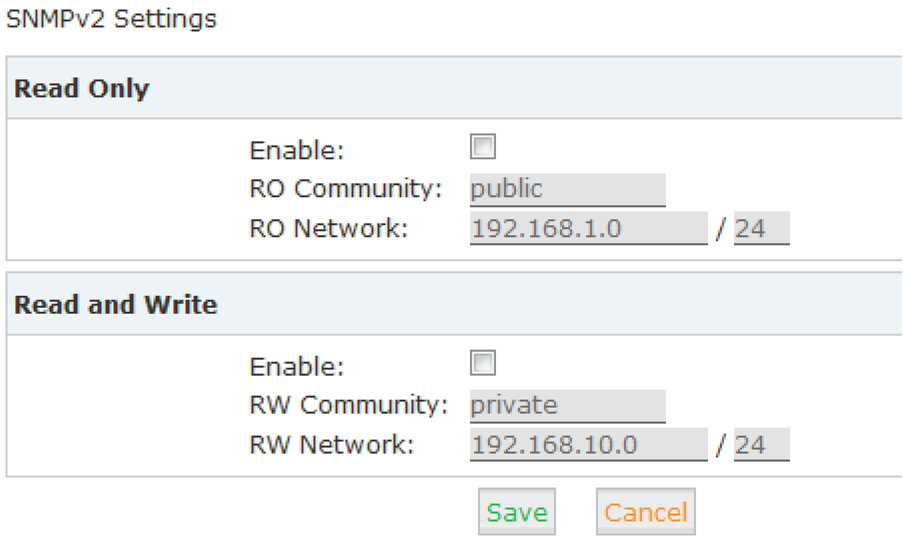
IP PBX supports DDNS provided by Planet DDNS, Dyndns.org, No-ip.com and zoneedit.com.



4.7 SNMPv2 Settings

SNMP (Simple Network Management Protocol) is used for remote management.

Click【Network Settings】->【SNMPv2 Settings】:

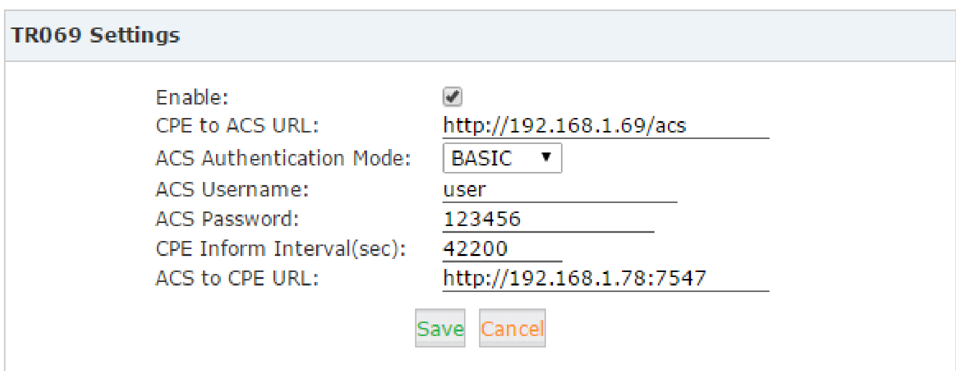


Reference

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

4.8 TR069

TR069 (Technical Report 069) is a Broadband Forum (formerly known as DSL Forum) technical specification entitled CPE WAN Management Protocol (CWMP). It defines an application layer protocol for remote management of end-user devices.



Item	Explanation
Enable	Enable TR069 service.
CPE to ACS URL	Input URL to visit ACS, which is used by PBX to connect ACS via CPE WAN management protocol (CWMP).
ACS Authentication Mode	Select ACS Authentication Mode: NONE/BASIC/DIGEST.
ACS Username	When PBX sends request to ACS, ACS will provide username to the authorized PBX.
ACS Password	When PBX sends request to ACS, ACS will provide password to the authorized PBX.
CPE Inform Interval (sec)	Interval for CPE to connect ACS.
ACS to CPE URL	Input URL to visit CPE. Format: http://IP:port(7547).

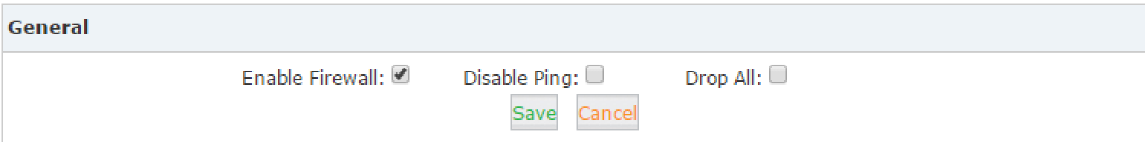
Chapter 5. Security

This chapter will introduce you how to configure the Security of PLANET IP PBX.

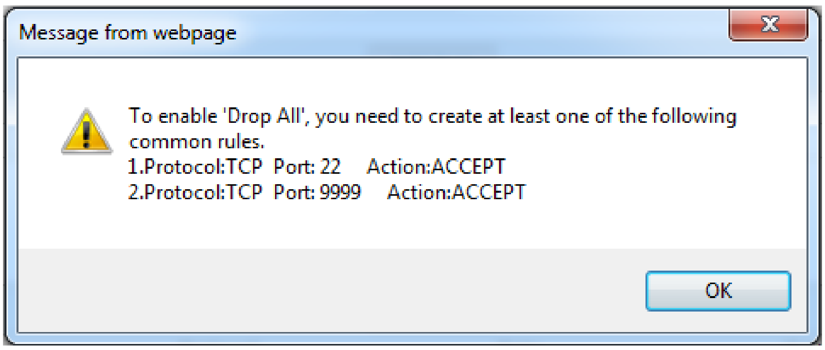
5.1 Firewall

The IP PBX system has been preconfigured with a built-in firewall which prevents your IP phone system from unauthorized accessing, phone calls and some other attacks.

General



Item	Explanation
Enable Firewall	By default, firewall is enabled. You may disable the built-in firewall by unchecking “Enable Firewall” checkbox, only if your IP PBX is behind a router/firewall without port forwarding to the Internet.
Disable Ping	Ignore ping request. If enabled, you cannot ping the IPPBX system.
Drop All	Drop all packets sent to the IP PBX system, this will cause IP PBX system blocking all communication with the outside world. So the system will prompt to add at least one grant rule on port 22 (SSH) or 9999 (Web) to make sure the IPPBX system is totally unreachable.

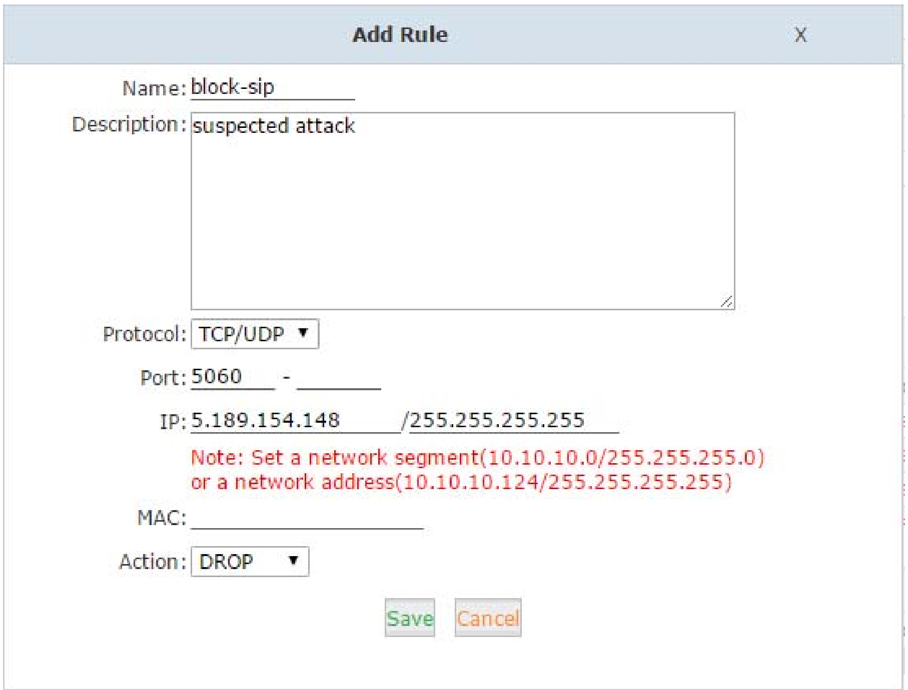


The rule/rules can be created in the “Common Rules” section.

Common Rules

In Common Rules section, you can configure the firewall to grant or deny an IP address or a network from communicating with the IPPBX system. Even the service port number can be specified so it can grant or deny a specific IP or network to access a specific service.

By clicking the “Add Rule” button you can add a custom rule for rejecting or accepting an IP address or network address.



Item	Explanation
Name	A name for this rule.
Description	Optional, you may describe why this rule is created.
Protocol	Transmission protocol, UDP, TCP or UDP with TCP.
Port	Service port number.
IP	Can be an IP address or a network address.
MAC	Action to be taken according to the Mac address of a device instead of IP. Only works with the devices within the same local network because Mac address cannot transport on Internet.
Action	Select “Drop” to block and “Accept” to grant.

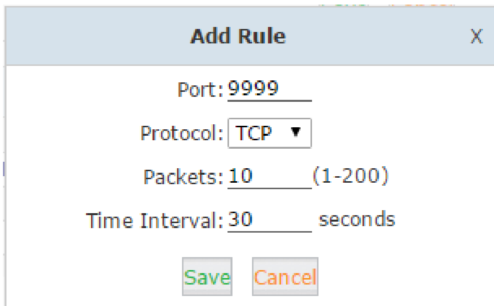
Auto Defense

The IPPBX system uses Fail2Ban to do intrusion detection, iptables is used for blocking the attack attempts. Fail2Ban is an intrusion prevention framework written in the Python programming language. It works by reading Asterisk logs and some other logs in the IP PBX system, and uses iptables profiles to block brute-force attempts.

In Auto Defense section you can define some custom rules to help the IP PBX system determine brute-force attempts.

Auto Defense	Add Rule		
Port	Protocol	Rate	Options
5060	UDP	120/30s	Edit Delete
5060	UDP	40/2s	Edit Delete
5061	TCP	80/2s	Edit Delete
22	TCP	10/60s	Edit Delete

Click the "Add Rule" button to add a new custom rule.



In this case, it will block the IP which will send more than 10 packets to the port 9999 within 30 seconds. This rule will prevent brute-force attempts of the web login.

Rejected IP

Any IP address that is banned will be shown in the table of "Rejected IP". The table will show the IP address of the banned host, as well as what kind of service was detected as to the intrusion.

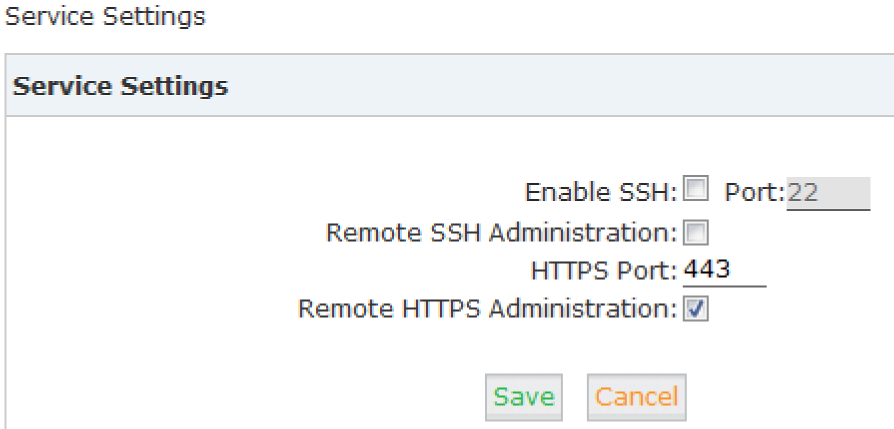
Rejected IP		
Type	IP	Options
VOIP	212.83.154.178	Delete
VOIP	173.249.158.227	Delete
VOIP	5.189.154.148	Delete

If a host appears incorrectly in the list of rejected IP, you can click on the "Delete" button to remove it from the list.

5.2 Service

As we can see here on this page, you are able to configure the SSH and HTTPS services.

Click【Security】->【Service】:



Item	Explanation
Enable SSH	With this option you can enable or disable SSH access to the IPPBX system. It's not enabled (unchecked) by default.
Port	By default SSH service port number is 22. You can change it to any other available port number.
Remote SSH Administration	If this option is checked, remote SSH access will be enabled.
HTTPS Port	Web GUI service port number, by default, is 9999. You can change to any other port number if needed.
Remote HTTPS Administration	If this option is checked, remote web access will be enabled.

Note

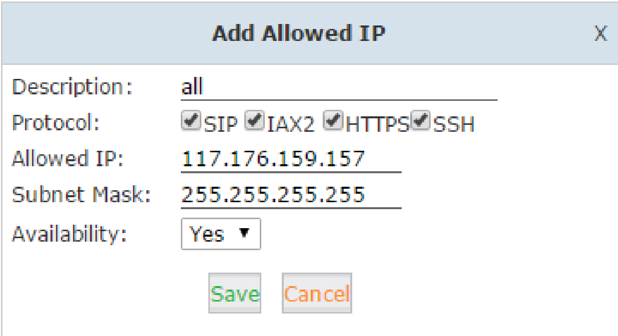
If you want remote access to SSH and web GUI of the IPPBX system, you can forward the corresponding ports on your router. Before doing this please make sure you have set strong password words for root user and web admin user.

5.3 Fail2Ban

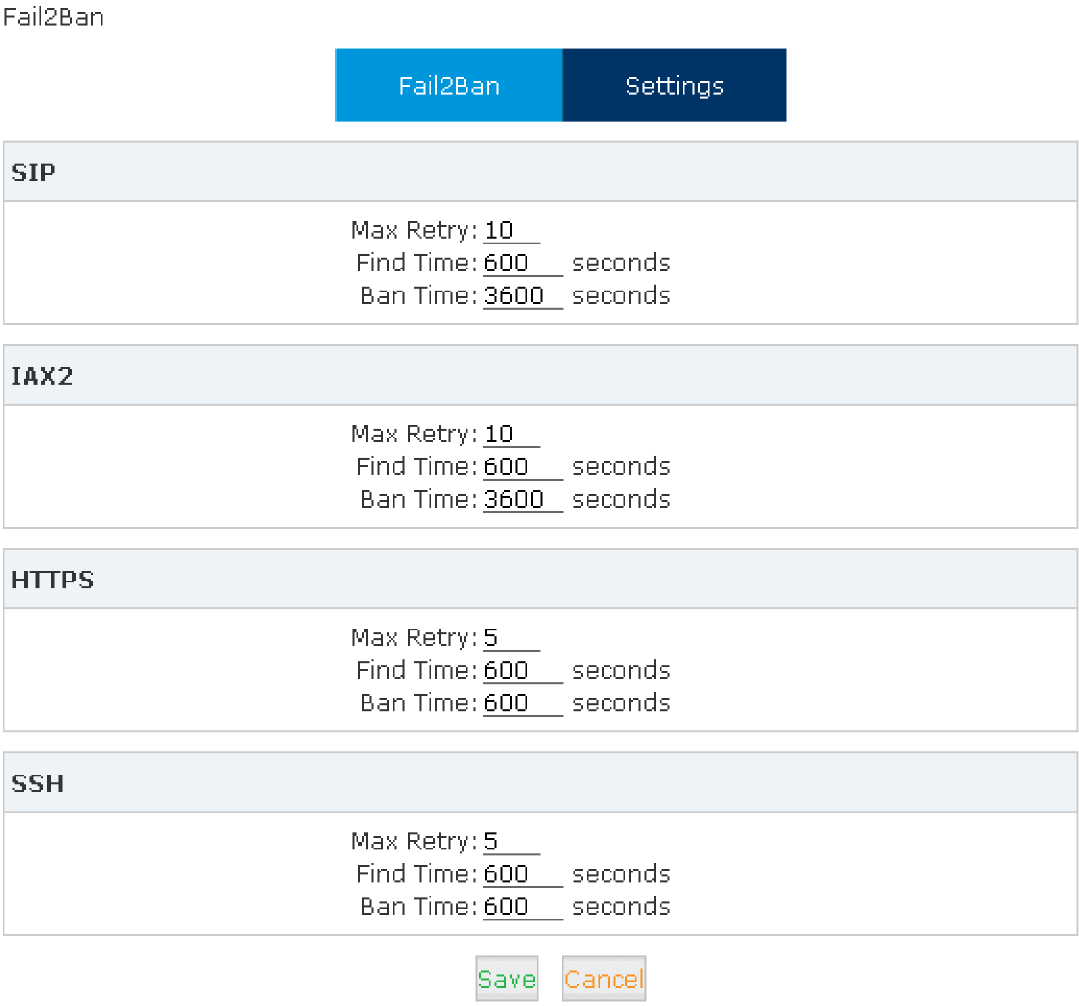
Planet IPPBX system uses Fail2Ban to perform intrusion detection; iptables is used for blocking any attack attempts. Fail2Ban is an intrusion prevention framework written in the Python programming language. It works by reading Asterisk logs and some other logs in the IPPBX system, and uses iptables profiles to block brute-force attempts. In the Auto Defense section you can define some custom rules to help the IPPBX system determine brute-force attempts.

Allowed address allows you to add IP addresses and network addresses to the IPPBX system as a whitelist. The IPs in the whitelist will be always treated as trusted IP and will not be filtered by the firewall rules.

Click the “Add New IP” button to add a trusted IP or network to the system IP whitelist.



Item	Explanation
Description	A name for this entry.
Protocol	Select protocols this IP/network can access.
Allowed IP	IP address or network to be trusted.
Subnet Mask	Netmask for this IP or network.
Availability	Choose “Yes” to activate this entry; choose “No” to deactivate.



These options are actually for Fail2Ban, the “Max Retry” limits the authentication attempts. “Find Time” defines the time duration from the first attempt to the last attempt which reaches the “Max Retry” limitation. “Ban Time” is the time in seconds the IPPBX system will block the IP which exceeded max retry. These settings also don’t effect on the allowed addresses.

Chapter 6. Report

6.1 Record Status

On register status page you are able to check the extension and SIP/IAX2 trunk status intuitively. You can see from which IP the extension is registered and also you can see the connection state, for example how much delay is there between the IPPBX system and the end point.

6.1.1 SIP User Status

Register Status

SIP Users Status	IAX2 Users Status	SIP Trunks Status	IAX2 Trunks Status

SIP Users Status

Name	Extension	IP	NAT	ACL	Port	Status
800	800	N/A	No	No	N/A	Unregistered
801	801	N/A	No	No	N/A	Unregistered
802	802	N/A	No	No	N/A	Unregistered
803	803	N/A	No	No	N/A	Unregistered
804	804	N/A	No	No	N/A	Unregistered
805	805	N/A	No	No	N/A	Unregistered
806	806	N/A	No	No	N/A	Unregistered
807	807	N/A	No	No	N/A	Unregistered
808	808	N/A	No	No	N/A	Unregistered
809	809	N/A	No	No	N/A	Unregistered

Here on this page you can see the SIP/IAX2 extensions, web extensions and also the register status of the trunk users. Only the trunk is configured as peer mode which will be listed here.

Status and Description

- Registered: Registration success.
- Unregistered: Registration failure or unapplied.
- Unreachable: Network delay.
- Timeout: Network timeout.

6.1.2 IAX2 User Status

IAX2 Users Status

Name	Extension	IP	Port	Reachability
412	412	192.168.7.32	4569	Registered (2 ms)
413	413	N/A	N/A	Unregistered

Status and Description

- Registered: Registration success.
- Unregistered: Registration failure or unapplied.
- Unreachable: Network delay.
- Timeout: Network timeout.

6.1.3 SIP Trunk Status

SIP Trunks Status		
Username	Hostname/IP	Status
5252742452	gw1.sip.us:5060	Registered
61921248	183.62.205.209:5060	Registered

Here you can see all your outbound SIP trunks' status.

Status and Description

- Registered: Successfully registered to the service provider and ready for phone calls.
- Request Sent: If this status, it's most probably the network is totally unreachable to the SIP server. Please make sure network setting on the IPPBX system is correct.
- Waiting for Authentication: If "Waiting for Authentication" then most probably the register request has already been received by the server side but cannot authenticate the register request due to credentials incorrect. Please double check your inputted credentials.
- Failed: After trying to register within certain time period without success, you get "Failed" on the trunk status.

6.1.4 IAX2 Trunk Status

IAX2 Trunks Status		
Username	Hostname/IP	Status
asterisk	192.168.7.146:4569	Registered

Here you can see all your outbound IAX2 trunks' status.

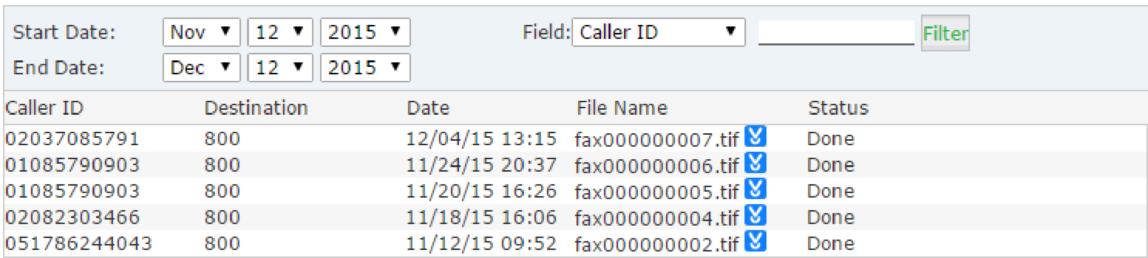
Status and Description

- Registered: Successfully registered to the service provider and ready for phone calls.
- Request Sent: If this status, it's most probably the network is totally unreachable to the service provider. Please make sure network setting on the IPPBX system is correct.
- Waiting for Authentication: If "Waiting for Authentication" then most probably the register request has already been received by the server side but cannot authenticate the register request due to credentials incorrect. Please double check your inputted credentials.
- Failed: After trying to register within certain time period without success, you get "Failed" on the trunk status.

6.2 Fax List

You can search any fax received by the IPPBX system.

Fax List



In the “Start Date” and “End Date” fields specify a time duration, and click “Filter” to get all faxes received during this period. If you specify a “Caller ID” or “Destination ID” in the field, you can get the fax sent/received by a specific number in this period.

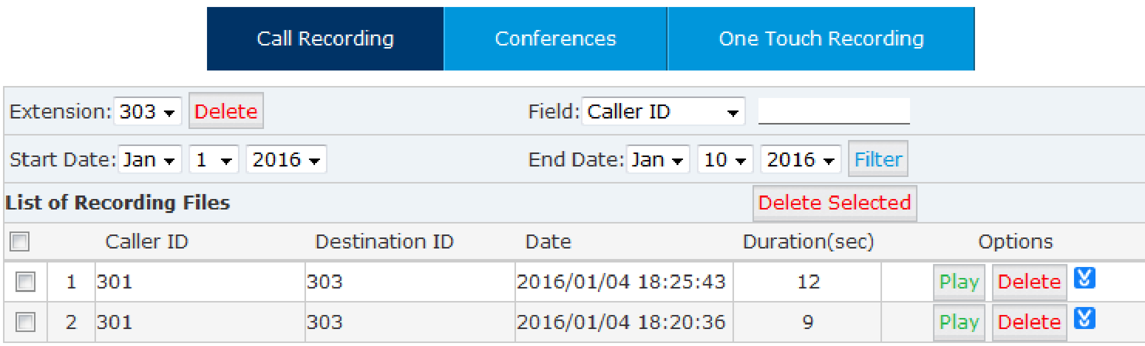
The faxes can be downloaded to your PC hard drive by clicking the button.

6.3 Record List

6.3.1 Call Recording

You are able to search all recorded call conversations if you have configured the extension always to be recorded.

Call Recording

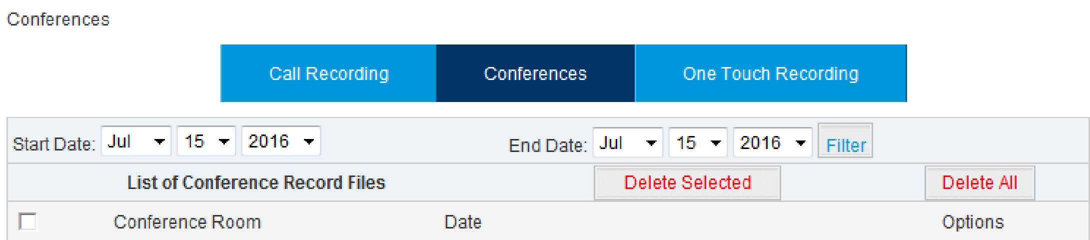


Item	Explanation
Extension	Select an extension number to search the recordings of this extension.
Delete	Delete all recordings of the selected extension number.

Item	Explanation
Field	Filter the recordings by specifying caller ID or destination ID. For example, if you select “Caller ID” and specify number 301, you get the recordings of the calls made by extension 301; if you select “Destination ID” and specify number 301, you get the recordings of the calls which called extension 301.
Start Date/End Date	Search the recordings during this time period.
Delete Selected	Delete the select recording items.
Caller ID	Caller ID of this recorded call.
Destination ID	The number the caller called.
Date	Exact time when this call recording began.
Duration (sec)	Duration of the recording.
Options	Playback, delete and download options of the recording files.
Play	You can play back the recordings directly on the web page or play back on a specific phone.

6.3.2 Conferences

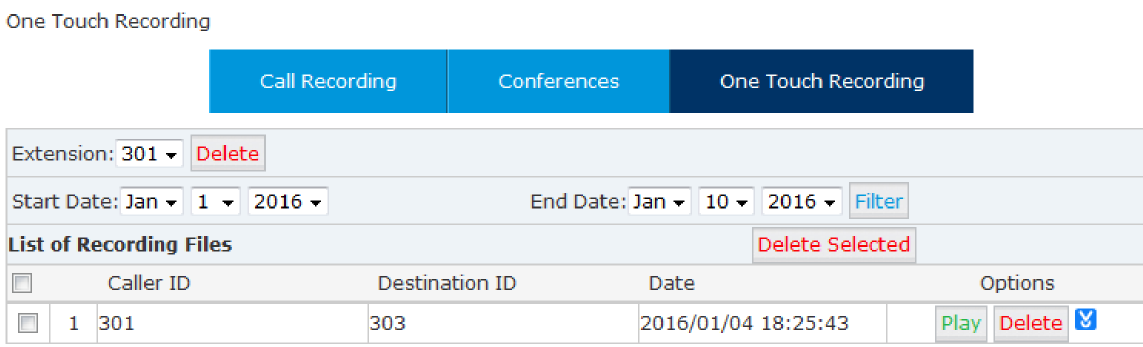
All recorded conferences can be found here on the Report->Record List->Conference page.



Item	Explanation
Start Date/End Date	Specify a time duration to search the recorded conferences.
Delete Selected	Delete the selected searched results.
Delete All	Delete all searched results.
Conference Room	The number of the recorded conference.
Date	Exact time when the conference began.
Options	Playback, delete or download the recording file.
Play	Playback the recordings directly on the web page or playback on a specific phone.
Delete	Delete the recorded audio file.

6.3.3 One Touch Recording

Call recordings recorded by one touch recording feature code *1 can be found on the *Report-> Record List->One Touch Recording* page.



Item	Explanation
Extension	Extensions that use one touch recording to record calls would be listed here.
Delete	Delete all recordings of the selected extension number.
Start Date/End Date	Search the recordings during this time period.
Delete Selected	Delete the select recording items.
Caller ID	Caller ID of this recorded call.
Destination ID	The number the caller called.
Date	The time when exactly this call began.
Play	Playback, delete and download options of the recording files.
Delete	Delete the recorded audio file.

6.3.4 Call Recording Playback

On IP PBX system, you have two ways to play back the recording files.

- Play back on the web interface
- Play back on a specific phone

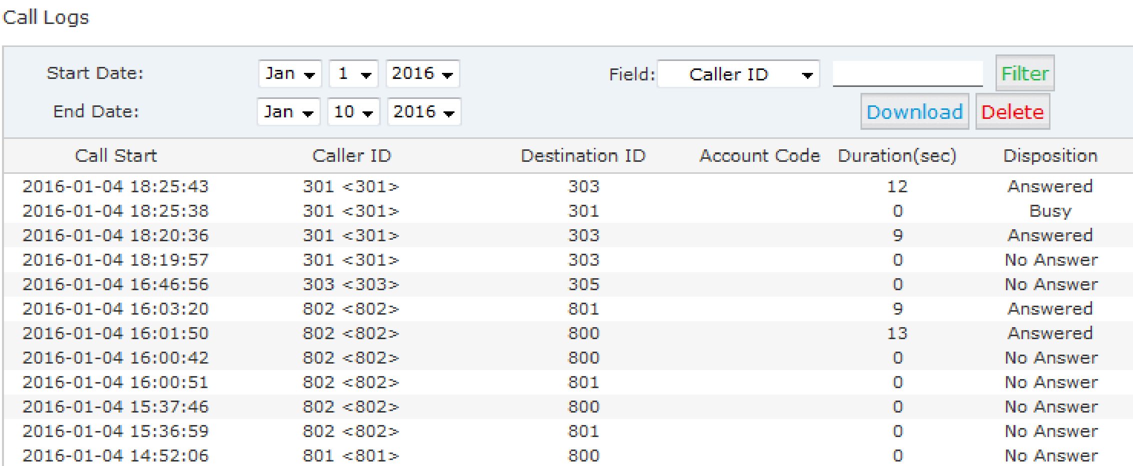
By clicking the “Play” button on a call recording file, you’ll see a dialog like below:

With “Type 1”, you can click the button to play back the recording directly on the web interface.

With “Type 2”, you can specify an extension number and click on “Play” to enable the extension to ring and the extension is answered that will play on the phone.

6.4 Call Logs

Call log is also known as CDR (Call Detailed Records). On the call logs page you can check any call records that go through the IPPBX system. Navigate to web menu *Report->Call Logs*. By specifying the time duration and/or Caller ID/Destination ID/Account, you can find out the logs you want.



Call Start	Caller ID	Destination ID	Account Code	Duration(sec)	Disposition
2016-01-04 18:25:43	301 <301>	303		12	Answered
2016-01-04 18:25:38	301 <301>	301		0	Busy
2016-01-04 18:20:36	301 <301>	303		9	Answered
2016-01-04 18:19:57	301 <301>	303		0	No Answer
2016-01-04 16:46:56	303 <303>	305		0	No Answer
2016-01-04 16:03:20	802 <802>	801		9	Answered
2016-01-04 16:01:50	802 <802>	800		13	Answered
2016-01-04 16:00:42	802 <802>	800		0	No Answer
2016-01-04 16:00:51	802 <802>	801		0	No Answer
2016-01-04 15:37:46	802 <802>	800		0	No Answer
2016-01-04 15:36:59	802 <802>	801		0	No Answer
2016-01-04 14:52:06	801 <801>	800		0	No Answer

Item	Explanation
Start Date/End Date	Define the searching time period by “Start Date” and “End Date”.
Field	Search criteria.

Item	Explanation
Caller ID	Searching by the caller number.
Destination ID	Searching by the called number.
Account Code	Searching with the pin code had been used for outbound dialing.
Download	Download the searching results.
Delete	Delete the searching results.
Call Start	The time when exactly this call began.
Caller ID	The number of the caller.(By clicking on the number you can add this number to the IPPBX system phone book.)
Destination ID	The number which has been called. (By clicking on the number you can add this number to the IPPBX system phone book.)
Account Code	The pin code had been used for outbound dialing.
Duration	The duration of this phone call.
Disposition	How the calls been handled. Answered, No Answer and Failed.

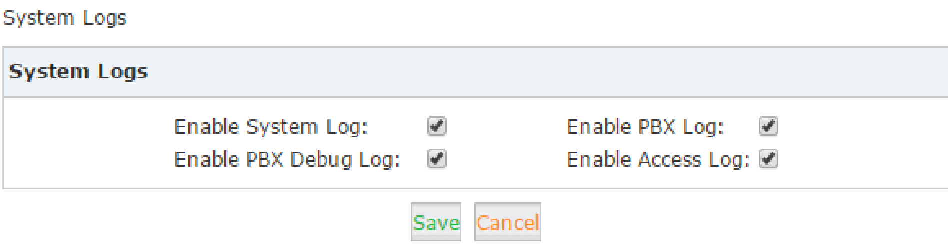
6.5 System logs

These logs are IPPBX journals which store all system activities. They can be used for debug purpose if the system is running into exception. Please do not enable these logs if the system is functioning properly, because there is a lot of data being generated and wrote into the logs files about every details of the system activities.

In the IP PBX system, there are 4 kinds of log files.

Item	Explanation
System Log	System Logs store all the system events.
PBX Log	PBX Logs store all the Asterisk events.
PBX Debug Log	Asterisk debug logs.
Access Log	Web and SSH access logs.

To enable these logs for the IPPBX system, please Navigate to web menu *Report->System Logs*. And enable the logs by ticking the corresponding checkboxes.



After checking the checkboxes, please click “Save”. The log files will be generated.

List of Logs Download Selected Delete Selected

		Name	Type	Options
	1	debug20151221.log	Debug Log	Delete Download
	2	login201512.log	Login Log	Delete Download
	3	pbx20151221.log	PBX Log	Delete Download
	4	sys20151221.log	System Log	Delete Download

Each day there will be a new log file generated for each of the log types. Enable them only if you are familiar with these logs for troubleshooting.

Chapter 7. System

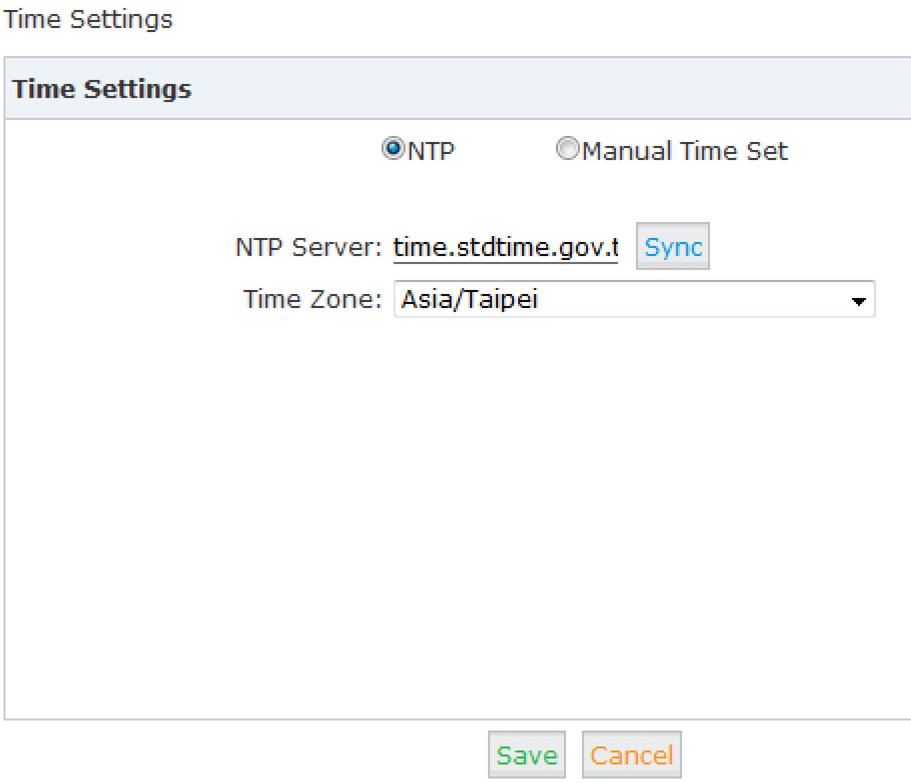
7.1 Time Settings

System time is very important for the IP PBX system. If the IP PBX system handles the inbound phone calls using time rule, then only the system time will correct the calls that can be handled properly. Beside call logs and debug logs, they record the system events using system time as well. The IPPBX system supports NTP (Network Time Protocol) and manual time set.

7.1.1 NTP

Navigate to web menu *System->Time Settings*.

By default, IP PBX system uses NTP to obtain time from Internet time servers. All you have to do is tell the IP PBX system where to find the server by specifying its domain or IP address. And also don't forget to select the correct time zone you are in.

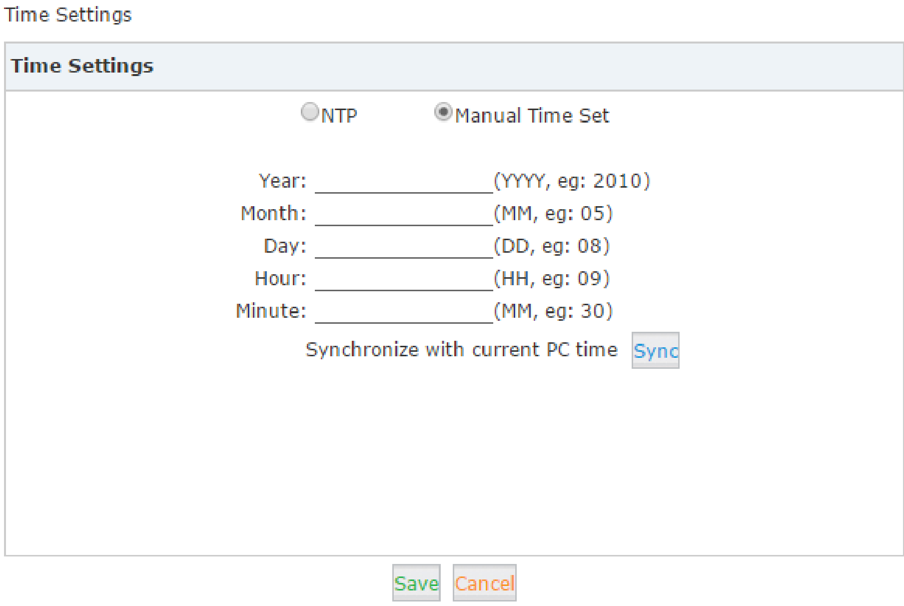


Once done, click the "Sync" button to enable IPPBX system to try to synchronize the current time from the Internet. It might take a while depending on the network conditions. After the process is done, you'll get notice "Sync Failed!" or "Sync Success!". If failed please check if the

IPPBX can access Internet or please change an NTP server and try again.

7.1.2 Manual Time Set

If you want to manually set time for the IP PBX system or for some special reasons, the IP PBX cannot access Internet. You can choose to manually set the system time by checking the "Manual Time Set" radio button.



There are two ways to manually set a time to the system.

1. Manually write down the time and date info and click "Save".
2. Synchronize the IP PBX system time with your PC time by clicking the "Sync" button and then click on the "Save" button.

Once "Save" is clicked the time manually written or synchronized from the PC will be stored into the hardware clock chip on board the IP PBX motherboard.

7.2 Module Settings

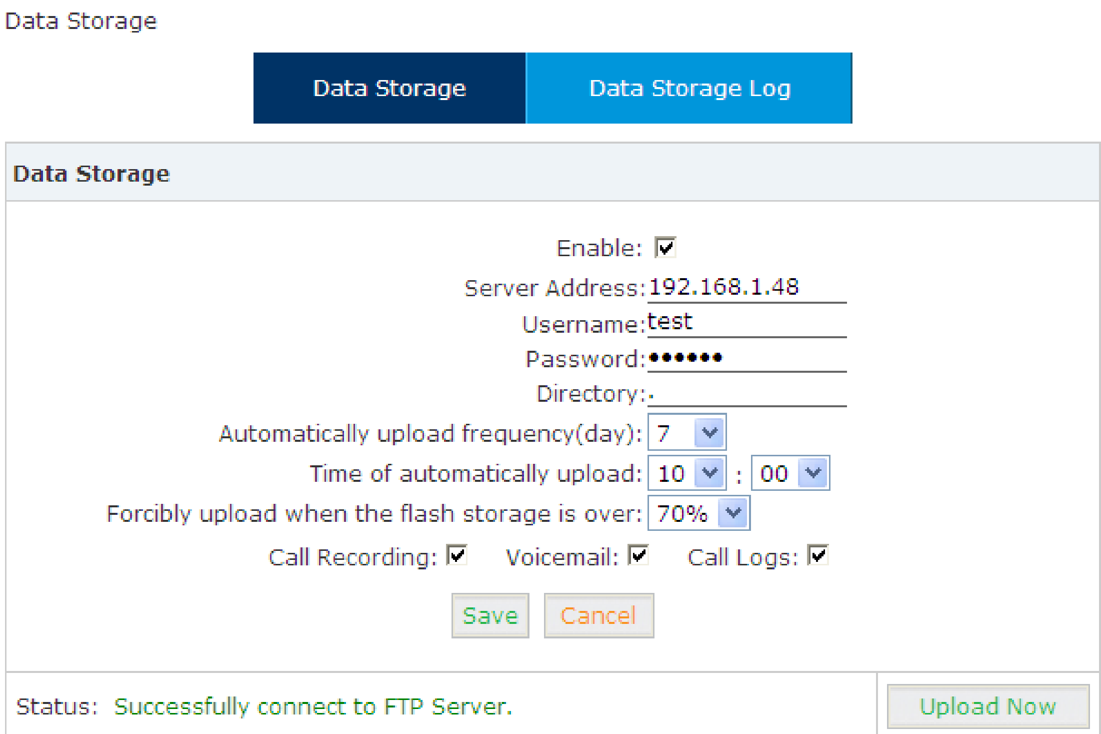
Planet IPX-2200 and IPX-2500 IPPBX systems need proper module settings to load correct drivers and configure files to drive the E1 and BRI telephony modules. Default module settings are with module types FXS/FXO/GSM on both telephony module slots. So if you don't have E1 and BRI modules installed, you don't have to configure module settings.

7.3 Data Storage

Data storage allows you to upload the recorded files, log files and voicemail messages to an FTP server through the Ethernet.

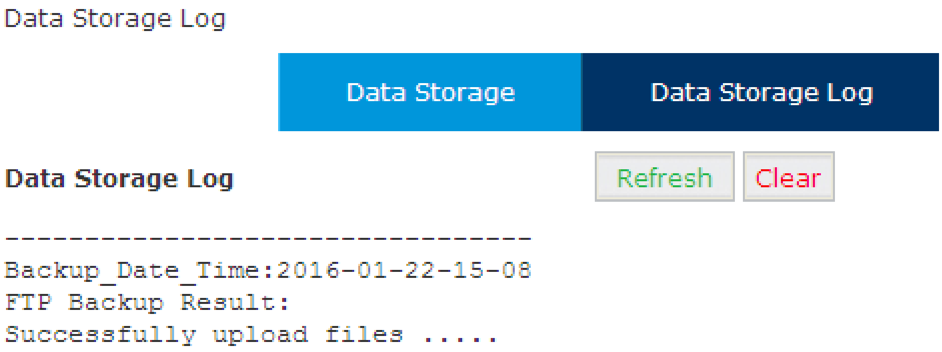
7.3.1 Data Storage

With your existing FTP server you can configure the IP PBX to upload the call recordings, voicemails and call log files to your FTP server. If you don't have one you can even use your Windows PC to set up an FTP server for the IPPBX system to connect. Just make sure your PC is always turned on or at least by the time IPPBX is going to upload files you have to turn on your Windows PC.

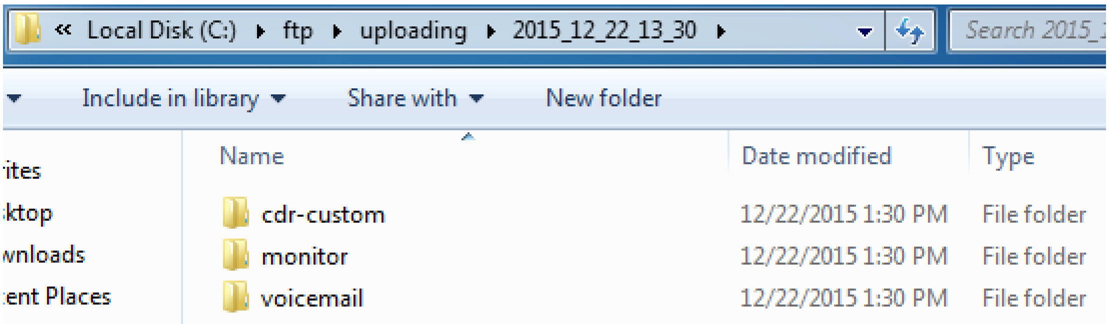


After these settings click "Save" to see the status "Successfully connect to FTP Server." You can click "Upload Now" to instantly upload a data.

Click on the “Data Storage Log” tab to enable the logs of each automatic data to upload as shown below.



After each upload, you’ll get a new folder on your FTP server directory named by the date and time of this uploading.

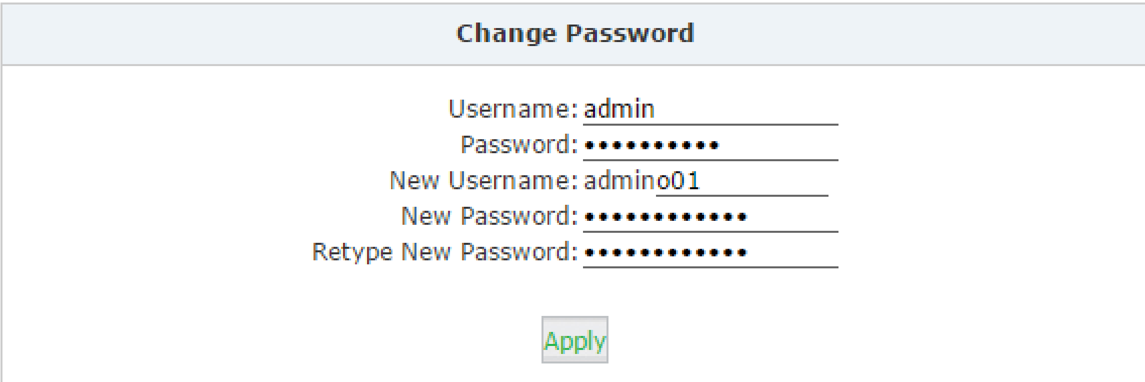


Note	After each upload except the call logs (Master.csv inside cdr-custom folder), other files will be removed from the IP PBX system, including the call recordings (files inside monitor folder) and voice messages (files inside voicemail folder). So after each upload, you will get only the newly-generated audio files.

7.4 Management

7.4.1 Change Password

In the “Change Password” section, you are able to change admin password, also admin username can be changed by adding some extra letters following name string “admin”.



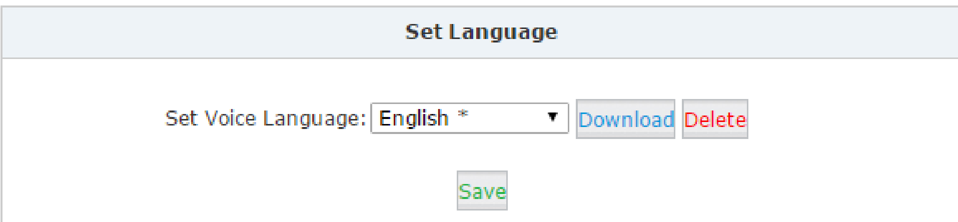
Once completed, click “Apply” to automatically get you logged out and redirected to the login page; now you are able to login with the new username and password.

7.4.2 Set System Voice Prompts

What's system voice prompts?

System voice prompts guide the callers to how to place a call or how to use the IPPBX system functionalities. For example, while checking voicemail the system voice prompts indicate the user to enter voicemail password and if nobody is answering a call, system voice will indicate leaving a message.

In the “Set Language” section you can set the language you want.



For now, IP PBX system supports 22 different languages as the system voice prompts. They are English, English (Australia), Chinese, French, French (Canada), Spanish, Spanish (Mexico), Portuguese, Portuguese (Brazil), Italian, Persian, Arabic, Turkish, Thai, Russian, Polish, Dutch, Korea, Hungary, Vietnamese, Hebrew, Greek and German.

The items with * mean these languages are already existing in the system; others can be downloaded here by clicking the “Download” button.

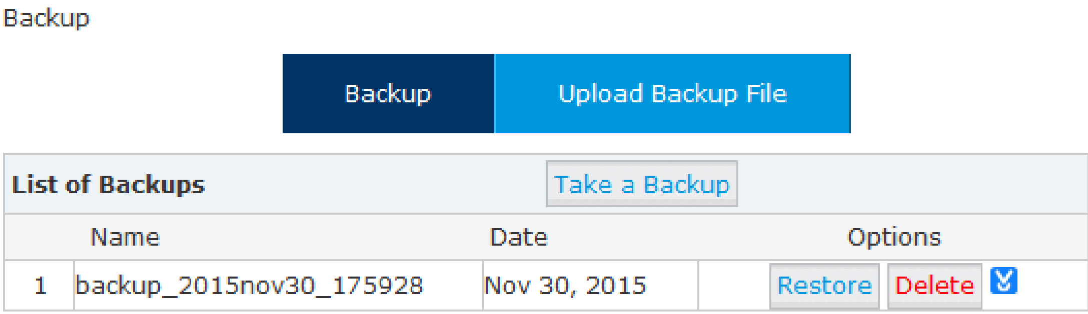
7.5 Backup

7.5.1 Take a Backup

Taking a backup on IP PBX system is the same as you create a recovery point on your Windows system. By restoring the backup you can recover the IP PBX system configurations to the time point when it’s still functioning well.

Normally the first backup should be taken when you finish configuring the IPPBX to work for the very first time. And maybe later you’ll apply new changes to the configurations in which you can take new backups as well.

Navigate to web menu *System->Backup*. Click the “Take a Backup” button to create a backup file which will contain all current system configurations.

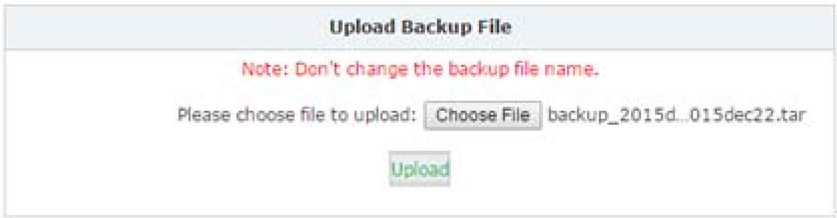


Once done, you get the backup file listed on this page. And the file is stored in the file system. Any time, by clicking the “Restore” button you can restore the configurations. By clicking the “Delete” button you can delete this backup. And you can also download the backup to your computer hard disk drive by clicking the button.

Note	If you are downloading the backup to your computer hard drive, please keep this file confidential, because this file contains web admin password, user extension password and many other sensitive information which may compromise your IP PBX system.

7.5.2 Upload Backup File

Click on the “Upload Backup File” tab to enable to upload a backup file from your computer hard drive.



Note	If you are uploading a backup from another IP PBX system, please make sure they have the exact same hardware configurations. It’s not recommended to upload backup files to different IP PBX systems, unless you are pretty comprehensive with IP PBX systems

7.6 Troubleshooting

Troubleshooting includes two tools for you to check the network reachability, ping and traceroute. With these tools you'll get an outside view of your network response time and network topology, which allows you to track down possible errors more easily

Click【Network Settings】->【Troubleshooting】:

7.6.1 Ping

The ping command is a very common method for troubleshooting the accessibility of devices. It uses a series of Internet Control Message Protocol (ICMP) Echo messages to determine:

- Whether a remote host is active or inactive.
- The round-trip delay in communicating with the host.
- Packet loss.

Ping 192.168.1.254 Packets: 4 Run Stop

```
PING 192.168.1.254 (192.168.1.254): 56 data bytes
64 bytes from 192.168.1.254: seq=0 ttl=64 time=5.773 ms
64 bytes from 192.168.1.254: seq=1 ttl=64 time=12.411 ms
64 bytes from 192.168.1.254: seq=2 ttl=64 time=3.637 ms
64 bytes from 192.168.1.254: seq=3 ttl=64 time=2.461 ms

--- 192.168.1.254 ping statistics ---
4 packets transmitted, 4 packets received, 0% packet loss
round-trip min/avg/max = 2.461/6.070/12.411 ms
```

By specifying the domain or IP of the host and how many packets to be sent, click the “Run” button to enable the command to begin the process. You'll get results indicating the reachability of the destination.

7.6.2 Traceroute

The traceroute command is used to discover the routes that packets actually take when traveling to their destination. Click the “Traceroute” tab and specify the domain or IP address you want to look up and then click the “Run” button to start the process.

Troubleshooting

Ping	Traceroute	Tcpdump	Channel Monitor

Traceroute 8.8.8.8 Run Stop

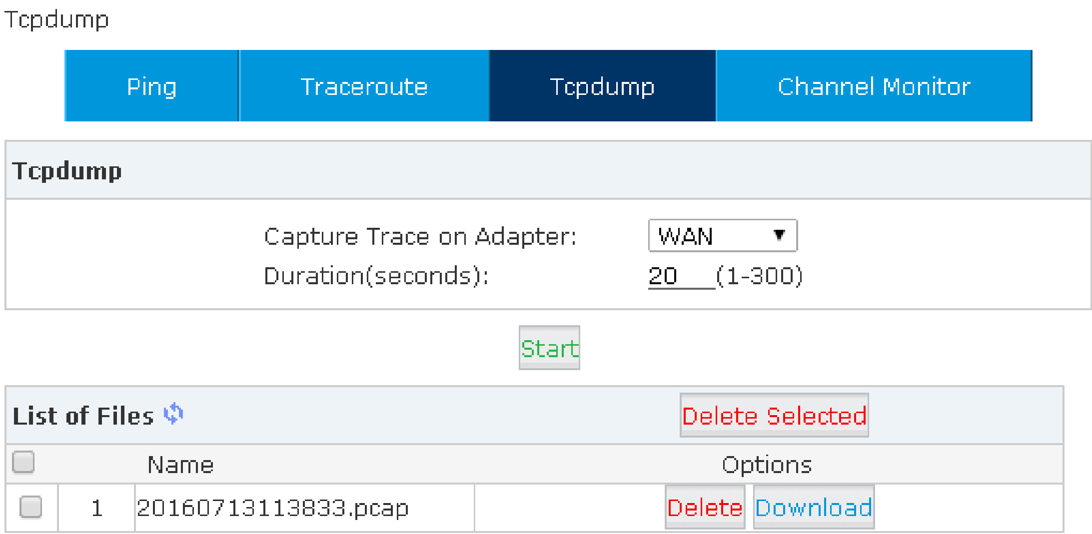
```
traceroute to 8.8.8.8 (8.8.8.8), 30 hops max, 60 byte packets
 1  192.168.1.254 (192.168.1.254)  0.523 ms  0.317 ms  0.810 ms
 2  210-61-134-254.HINET-IP.hinet.net (210.61.134.254)  16.735 ms  16.68
 3  tpe4-3302.hinet.net (168.95.229.86)  16.911 ms  16.864 ms  16.827 ms
 4  211-22-226-1.HINET-IP.hinet.net (211.22.226.1)  100.742 ms  100.679 r
 5  209.85.243.30 (209.85.243.30)  33.506 ms  33.457 ms 72.14.233.20 (72.
 6  209.85.242.163 (209.85.242.163)  23.604 ms 209.85.252.161 (209.85.25
 7  209.85.243.23 (209.85.243.23)  24.911 ms 209.85.247.57 (209.85.247.5
 8  * * *
 9  google-public-dns-a.google.com (8.8.8.8)  25.697 ms  25.660 ms  21.5
```

After the process, system will notice “Trace Complete” and you can see which routes the packets being taken before reaching the final destination.

7.6.3 Tcpdump

TCPDUMP is a common pocket analyzer that allows users to capture TCP/IP and other packets being transmitted or received over a network to which the Planet IPPBX is attached. The captured packets can be downloaded from the IPPBX system and been analyzed on your Windows PC to display the SIP traffic details. It can be used to debug a VoIP call problem.

On the 【System】->【Troubleshooting】->【TCPDUMP】page, you can do a capture on one of Planet IPPBX Ethernet interfaces.

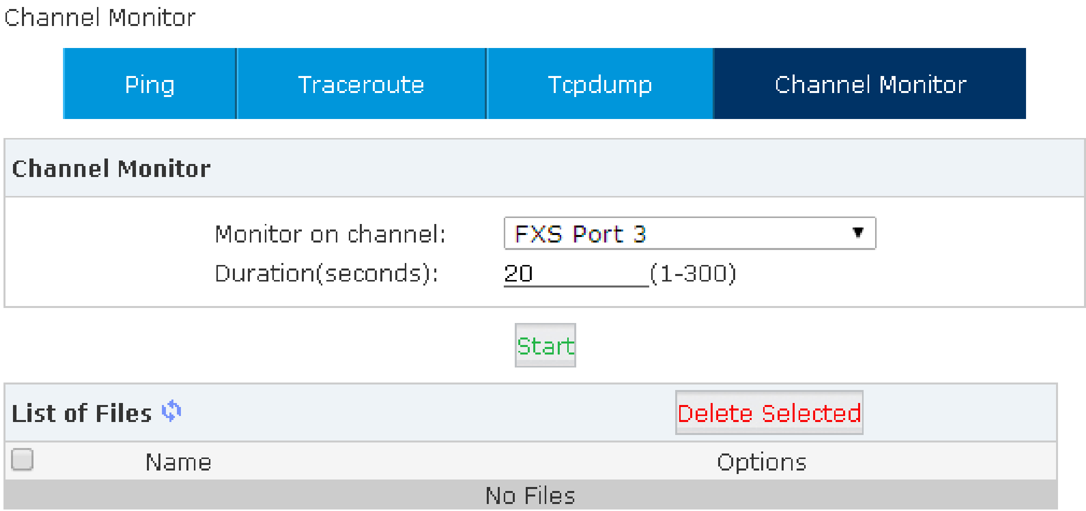


Select an interface and specify the duration of this capture and then click on “Start”. The process will begin and now you can make a call to recur the problem. Once the time is up, the captured packets will be displayed in the “List of Files” section. You can download it to analyze the SIP packets for troubleshooting purposes.

7.6.4 Channel Monitor

Channel Monitor is technically a DAHDI Monitor that allows you to monitor signal level on analog channel and record the output to a file. Recorded audio files are by default raw signed linear PCM. You can play it to the speaker to listen to the phone call signaling on the analog channel. Or you can use a sound editor to visually display the audio level at both the Rx (audio Received by Asterisk) and Tx (audio Transmitted by Asterisk).

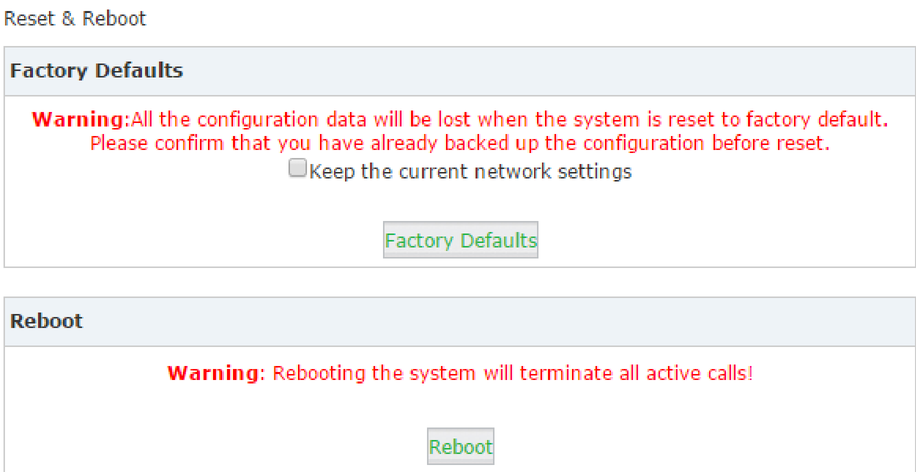
Usually Channel Monitor can be used to capture the caller ID signaling of an FXO channel. If you are experiencing caller ID problem, you can perform channel monitor on the FXO port and then analyze the captured packets.



In the “Monitor on channel” field, you should select a channel to be monitored. And then you have to specify the duration to monitor. Then click on “Start” to enable the capture to begin. Now you should make a call in from this channel (port). After the capture is done, you’ll get the file listed in the “List of Files” section.

7.7 Reset & Reboot

Navigate to web menu *System->Reset & Reboot*.



As you can see here on this page, you are able to reset and reboot the IPPBX system directly via web GUI.

7.7.1 Reset

By clicking the “Factory Defaults” button, you can reset all configurations of the IP PBX system. Except the configurations to be reset, the recording files, voicemail messages and call logs will also be erased. So please make sure you have backed up the files you need before resetting. The whole resetting process will be done in 2 minutes. If you choose to reset network settings also, then you need to login with the default URL https://172.16.0.1 on WAN. Username and password will all be reset to **admin**.

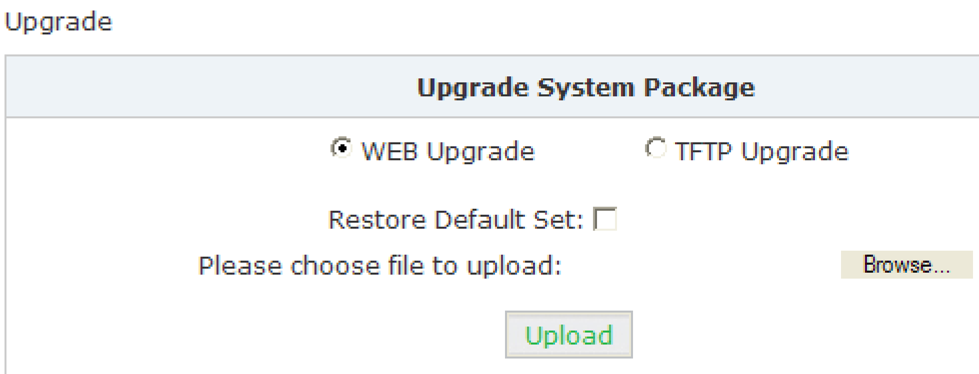
7.7.2 Reboot

By clicking “Reboot” you can restart the IPPBX system. The whole process will be done in 2 minutes.

7.8 Upgrade

Planet will update the IPPBX firmware irregularly for new features and bug fixes. You can visit our office website www.planet.com.tw to check the updates for your IP PBX system. The downloaded firmware package should be in .zip format. Please extract the package first and upgrade with the uImage-md5.xxx file to upgrade your IP PBX system. You can see there are two methods – Web upgrade and TFTP upgrade -- to upgrade the IPPBX firmware.

7.8.1 Web Upgrade



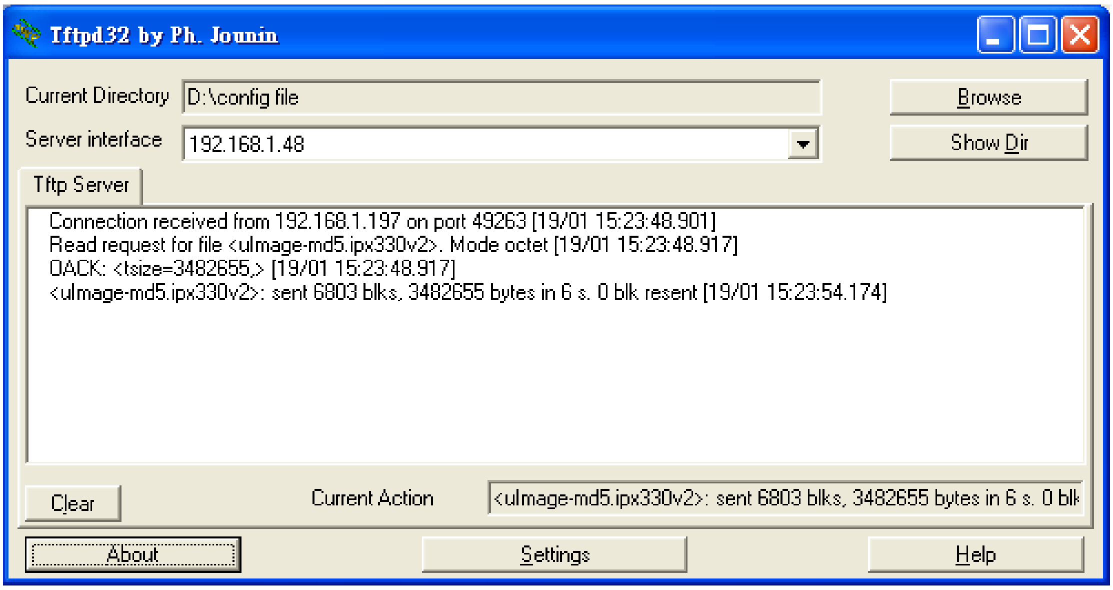
Check the “Web Upgrade” radio button and click the “Browse” button to locate the new firmware in your PC hard drive. Click “Upload” and it will ask you to confirm if restarting the IP PBX system to complete the upgrading process. You can click “Yes” to continue upgrading.

Note	The “Restore Default Set” option is used to reset the IP PBX system configurations while upgrading; if not otherwise specified you don’t have to enable this option to

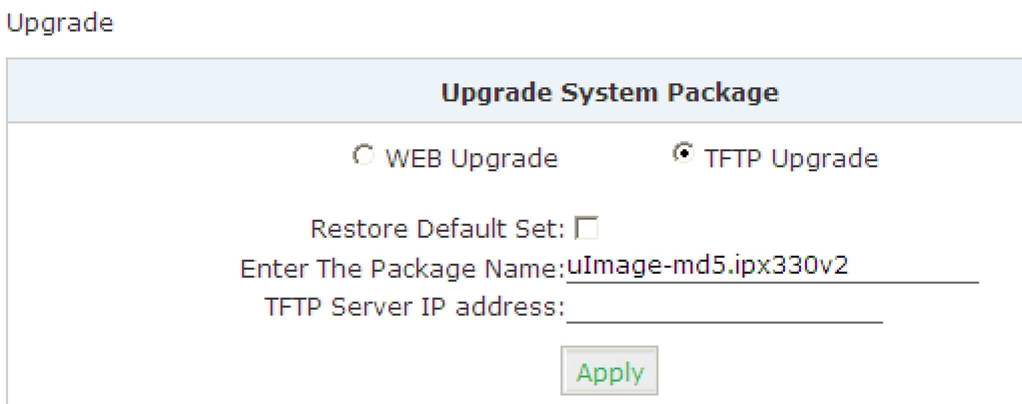
	reset the IP PBX system. If you want to reset, it will reset all system configurations including the network profiles.

7.8.2 TFTP Upgrade

If you don’t have a TFTP server, you can Google tftpd32 and download this application to set up a lightweight TFTP server on your Windows.



Please click “Browse” on the TFTP application window to locate the new firmware. And in the “Server Interface” dropdown list, it’s a list of your PC network interfaces. Please select a correct interface (in the same network) which can access the IP PBX system. On the IPPBX web GUI please check the “TFTP Upgrade” radio button, and specify the exact firmware file name on the “Enter The Package Name” blank, and in the “TFTP Server IP address” blank please specify the IP address displayed on the TFTP application window.



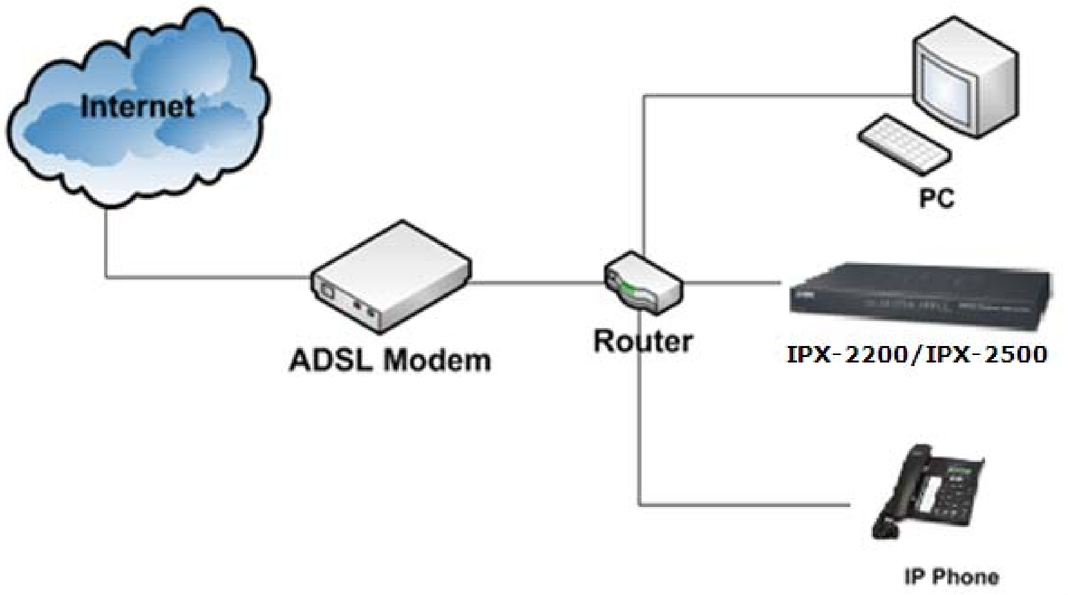
Please double-check the file name and TFTP server IP address and then click “Apply” to enable to upgrade the firmware just like web upgrade.

Chapter 8. Operating Instructions

This chapter will introduce you how to use PLANET IP PBX by example.

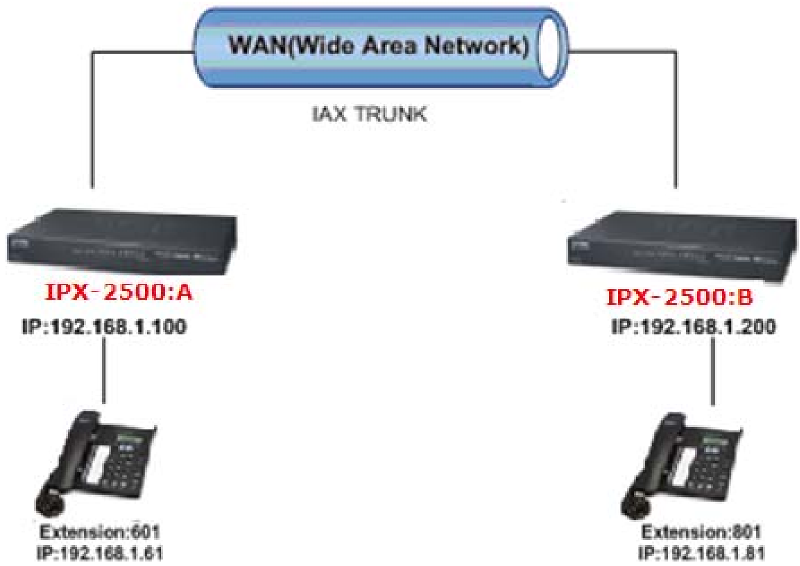
8.1 How to connect the IP PBX to the Internet

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



8.2 How to combine two IP PBXs in a different network

Normally, two sets of the IPX-2500 are located in different places with different IP addresses for Internet access.



For external line configuration, you must use public IP address.
Take the following instructions as an example:

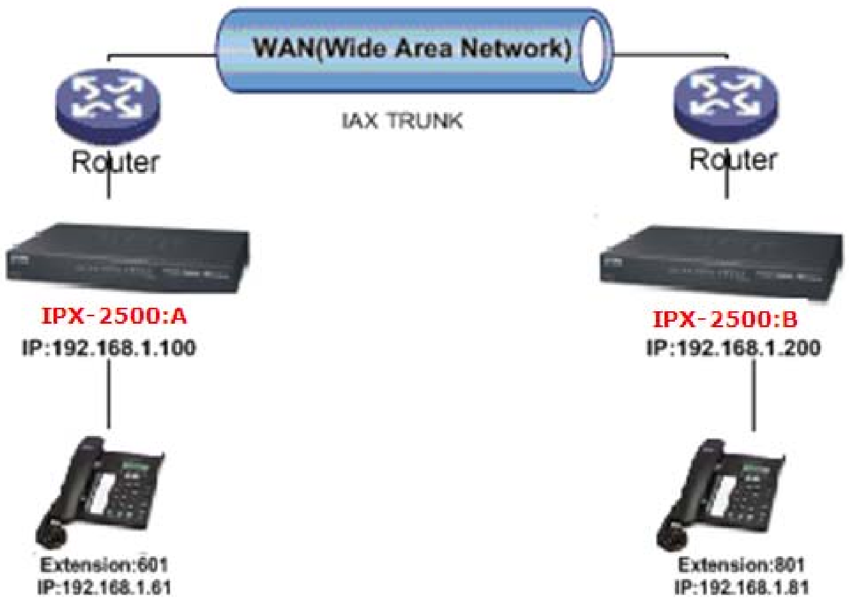
Register IPX-2500-B IP to a trunk of IPX-2500-A with authentication.
Configuration Rule:

1. IP Phone registers on IPX-2500-A as extension 601.
    1. Another IP Phone registers on IPX-2500-B as extension 801.
    2. IPX-2500-A IP: 192.168.1.100.
    3. IPX-2500-B IP: 192.168.1.200.
    4. Extension format of IPX-2500-A: 6XX.
    5. Extension format of IPX-2500-B: 8XX
    6. Create an extension 888 with password 123456 on IPX-2500-B.
    7. All extensions on IPX-2500-A can call extensions on IPX-2500-B with format 8XX.
    8. All extensions on IPX-2500-B can call extensions on IPX-2500-A with format 6XX.

For detailed steps, please take Chapter 8.2 as reference.

Two sets of IPX-2500 **behind router**

Sometimes the IPX-2500 doesn't have a public IP address, and you have to configure port mapping for your router.

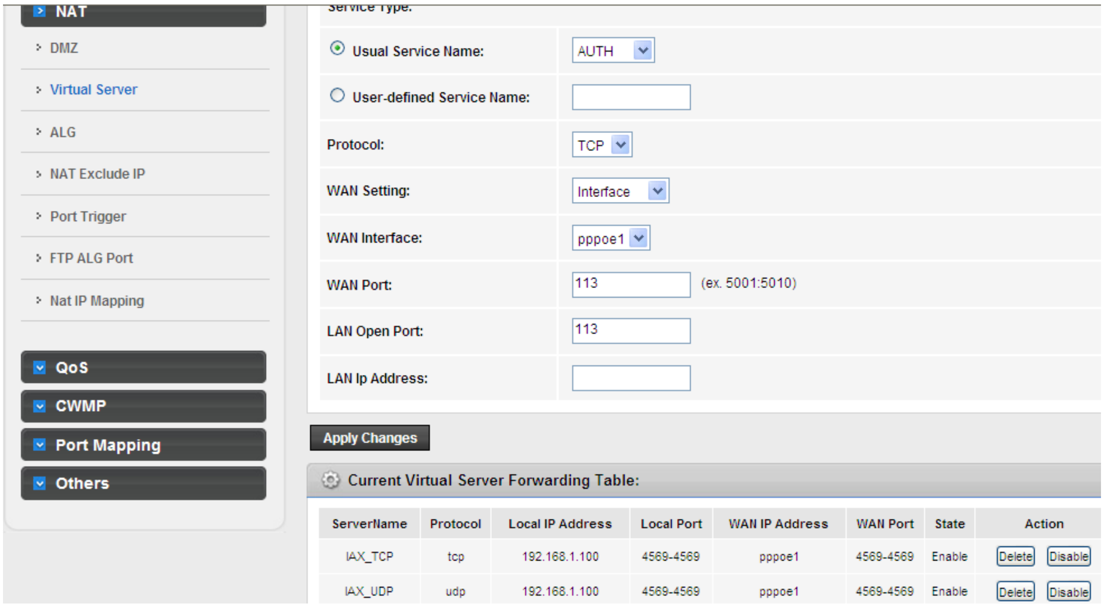


Step 1: Configure the mapping rule of IPX-2500-A on the router.

The IPX-2500-B is connected behind the router, and registers on IPX-2500-A through internet. You need to configure the port mapping of IAX2 port (4569) on the router. Then, all data received from RJ11 port of router (192.168.1.100:4569) will be sent to IPX-2500-A

Now, take the web management panel of ADN-4102 router as an example.

In here both UTP and TCP must open for IP PBX.



Step 2: IPX-2500 Configuration

Configure the trunk and dial plan on IPX-2500-B, and register IPX-2500-B IP to IPX-2500-A. The configuration is the same as the above, but you have to replace the public IP address with the internal IP: 192.168.1.21.

Step 3: Configure port mapping rule of IPX-2500-B on the router

Configure port mapping of IPX-2500-B on the router according to Step 1.

Step 4: Connect two sets of the IPX-2500 and make the call

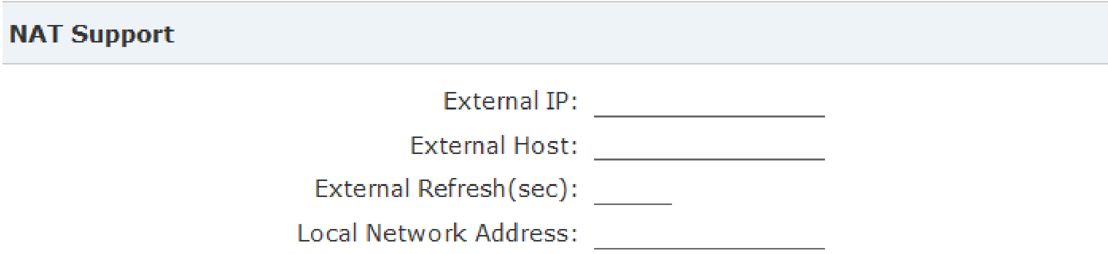
Create extension 601 on IPX-2500-A, extension 801 on IPX-2500-B, and create the correct outbound rule.

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

8.3 How to resolve the problem about hearing one side only

If the IPX-2500 is behind router, to resolve the problem, please set up IP address as shown below:

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



Item	Explanation
External IP	External IP or domain to replace the device IP
External Host	External domain to replace the device IP.
External Refresh(sec)	Refresh time, default is 10 seconds
Local Network Address	IP address and subnet mask needed to be converted. e.g. 192.168.1.100/255.255.255.0

8.4 How to use soft phone in IPX-2200 or IPX-2500

8.4.1 Softphone on Windows PC

The softphones 3CX, Bria, Zoiper and many other softphone Apps all can work with IP PBX. Below is an example of registering Zoiper to IP PBX system as an extension from your Windows PC.

Step 1:

Dowload Zoiper from http://www.zoiper.com/.

Step 2:

Install and run Zoiper on your Windows.

Step 3:

Click menu “Settings” and select “Create a new account” and select “SIP” protocol and click Next.

Step 4:

Fill in the register credentials shown below.

Step 5:

Click Next to complete registering.

8.4.2 Softphone on Android Phone, iPhone or iPad

Most of the softphones mentioned previously have mobile editions for both Android and iOS platforms. You can download to install from your mobile phone App Store.

Below is how you register Zoiper softphone to IP PBX as an extension from your iPhone:

Step 1:

Run Zoiper on your iPhone and tap 

menu.

Step 2:

Tap  Accounts menu.

Step 3:

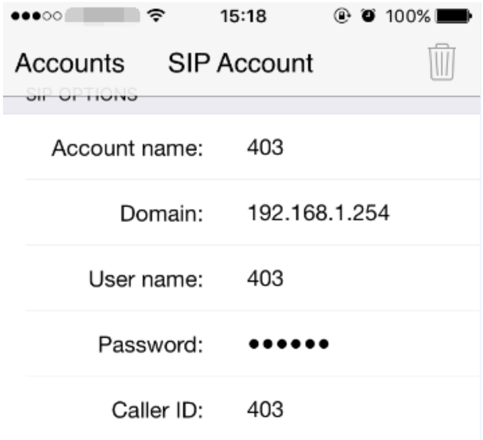
Tap  to create a new account.

Step 4:

It asks “Do you already have an account (username and password)?” Tap “Yes” and then tap “Manual configuration” to continue.

Step 5:

Tap SIP account to configure the account shown below:



Step 6:

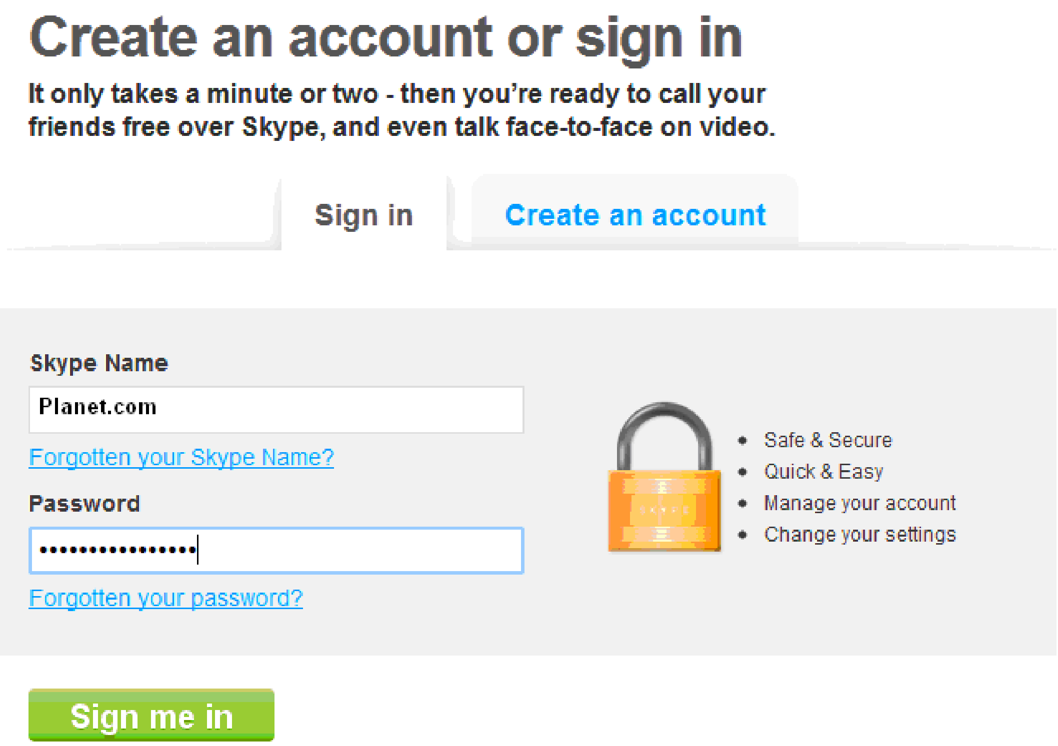
After entering the register credentials, tap Register to register to IP PBX system as an extension.

8.5 How to use Skype account in IPX-2200 or IPX-2500

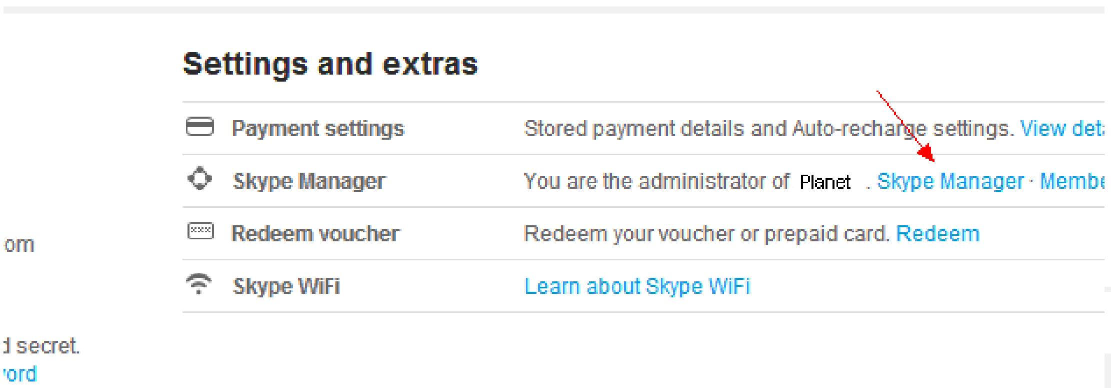
Note	The fee of your business account is much more than **€50** when you use the account for the first time.

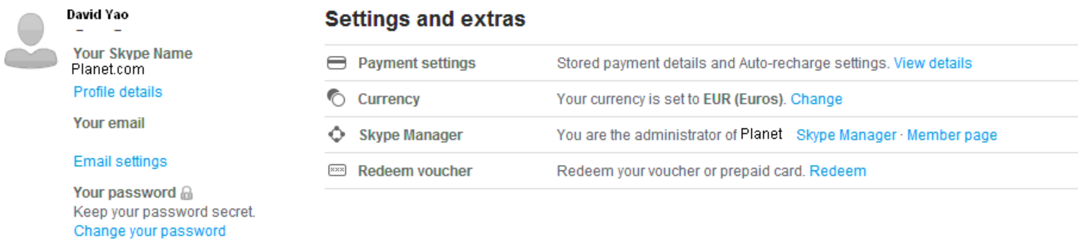
1 https://login.skype.com

Sign in with the business account.

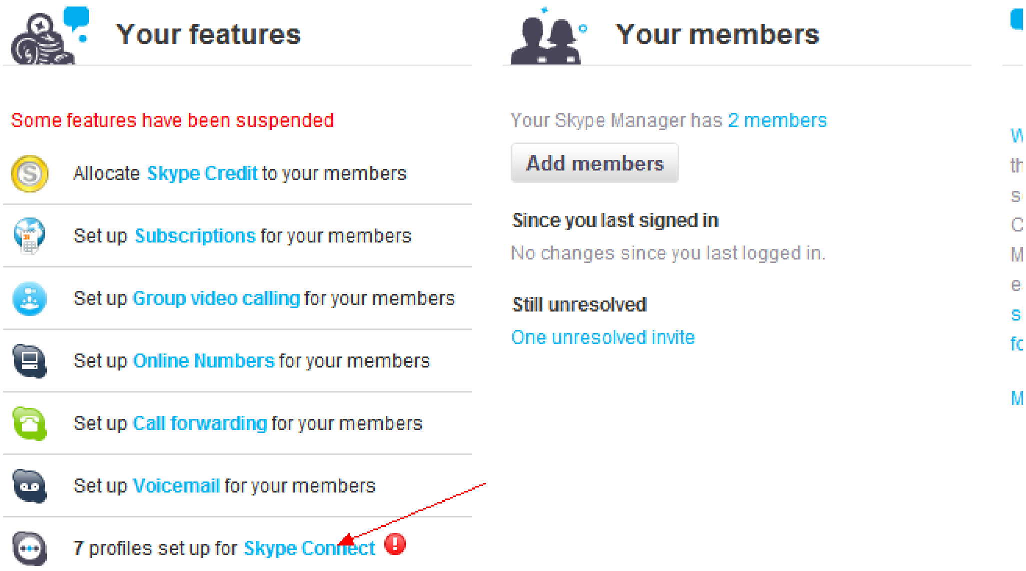


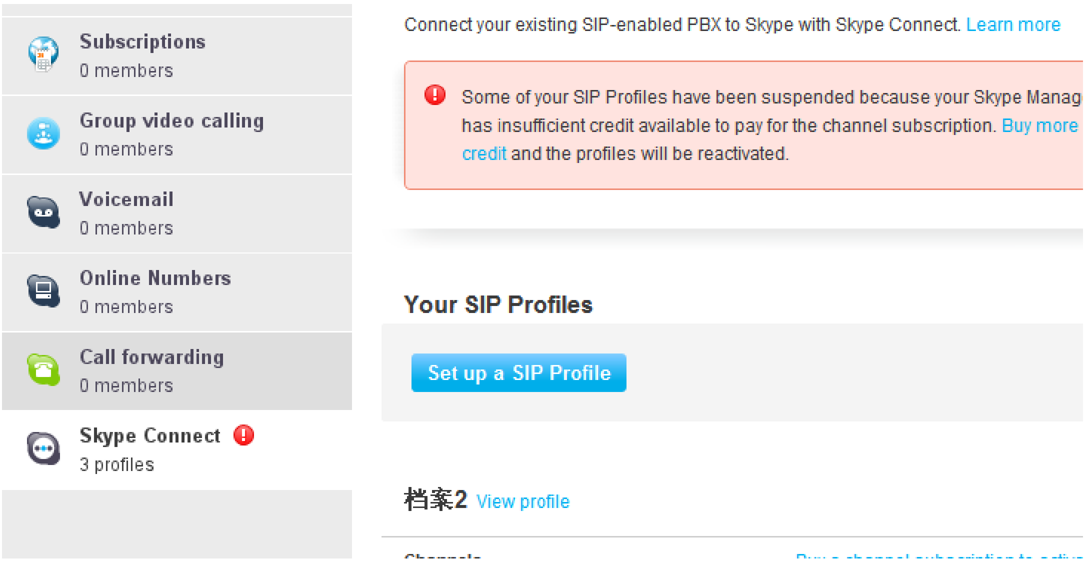
2 When you have signed in, at the end of this page, you will find the **Skype Manager**, Please click it.



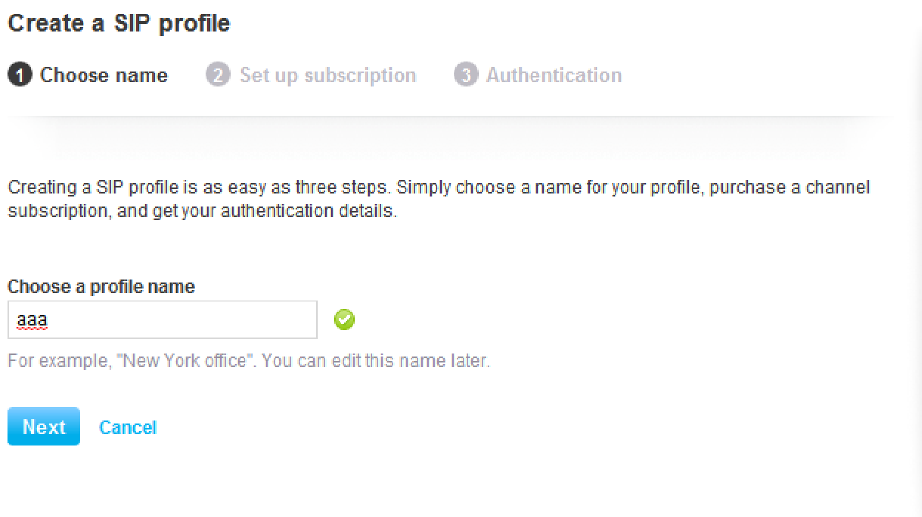


3 Please click the **Skype connect**

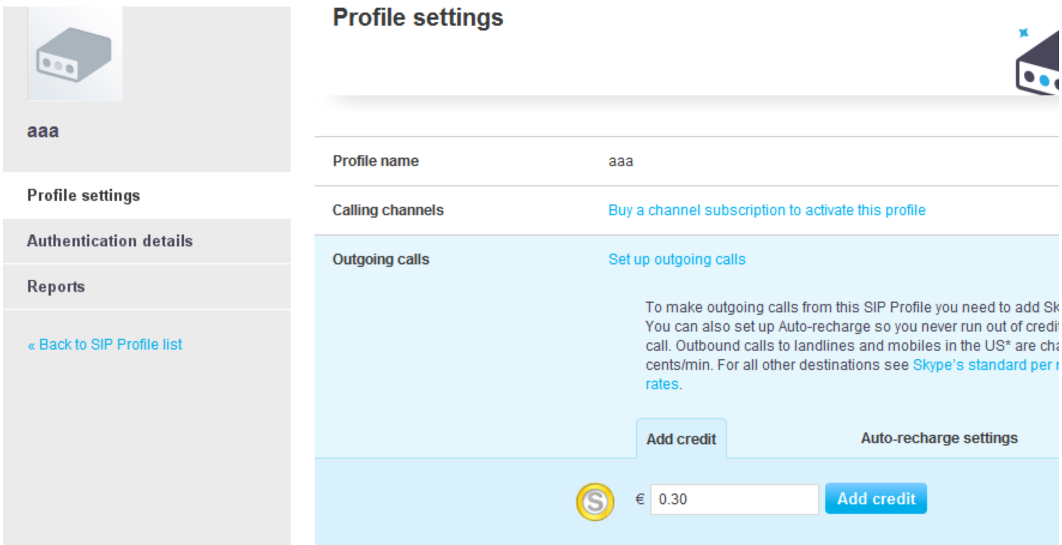




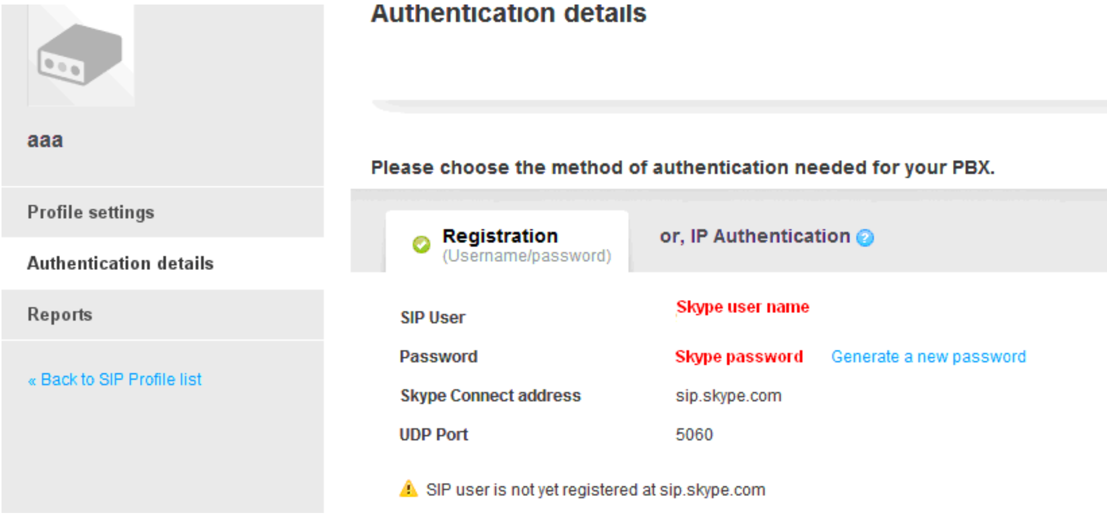
4 Create a SIP profile



Then you can create one SIP account. You need to pay **€ 4.95** for one channel as monthly rent and you need to input the registration information in our VoIP trunk blank. Then you can register with Skype server. And then you need to assign money for **outgoing calls**, and then you can call out.



Then you can see the SIP account information, and please click the **Authentications details.**



5 Settings on IP PBX

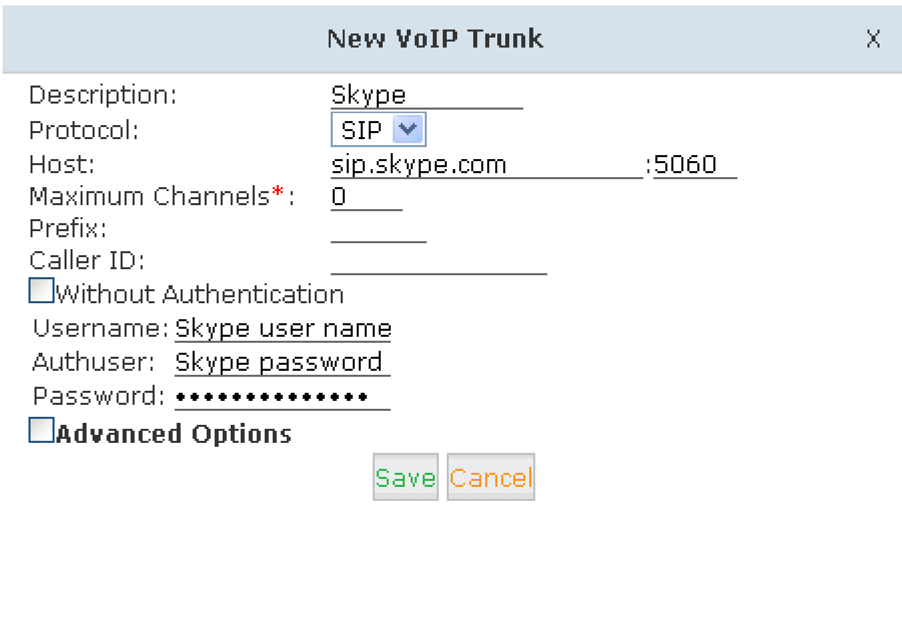
5.1 Build one SIP trunk with Skype for SIP account

Provider Type: Custom Trunk

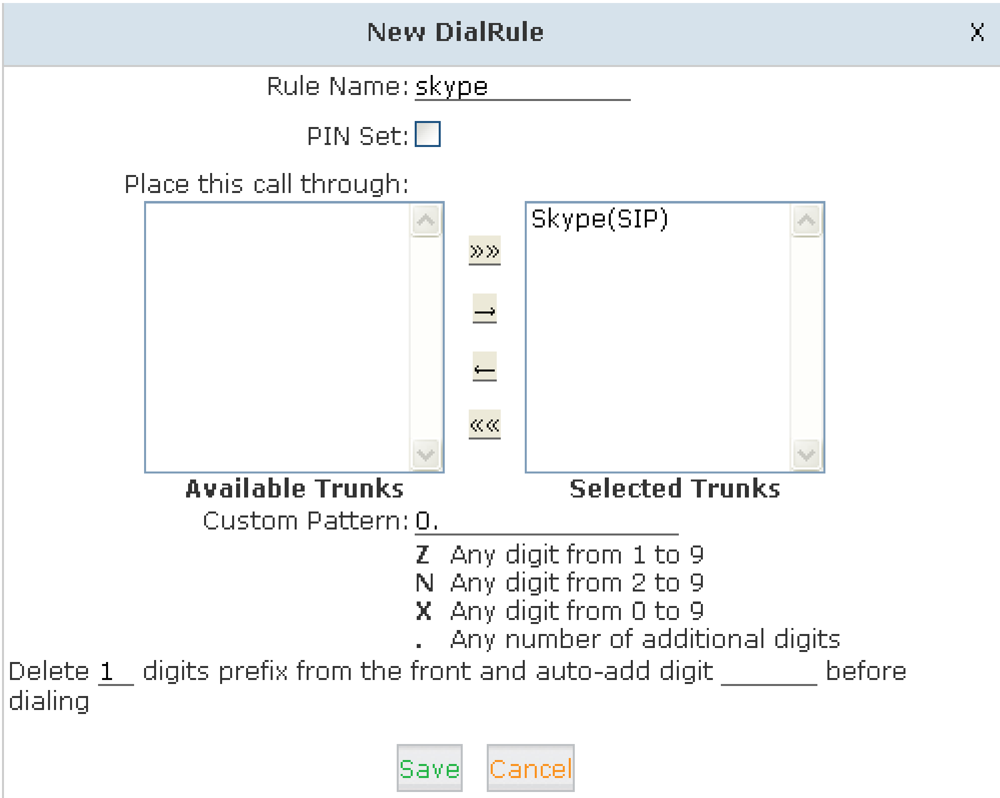
Host: sip.skybe.com

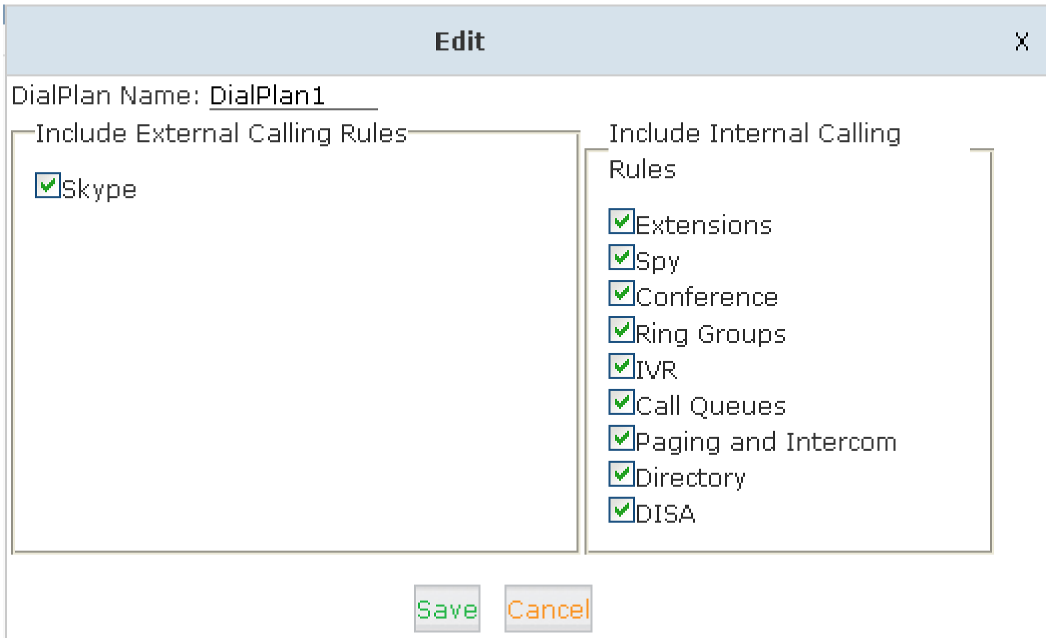
User name: the user name you defined in Authentication detail

Password: the password you defined in Authentication detail



5.2 Set one outbound rule





5.3 Make an outbound call

After we have done the above, in the extension we can dial 00 + Country Code + City Area code + local number to dial out via Skype line

For example, dialing number 00 (outbound prefix number) + 001 (International Code) + 886

(Country code) + 2 (city Area code without 0) + 22199518 (local phone number) will enable you to contact Taiwan Planet Company

5.4 Set inbound rule