# 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: 1.1 (Nov., 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

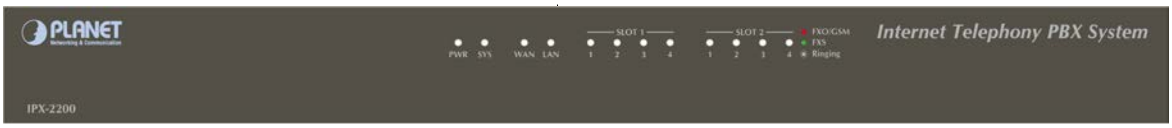
If any of the above items are damaged or missing, please contact your dealer immediately.

### 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<br>Off | PBX Power ON<br>PBX Power OFF |
| **SYS** | Blinking Green<br>On<br>Off | System is working<br>System doesn't boot<br>System failure |

| Front Panel LED | Status | Description |
|---|---|---|
| **WAN** | Blinking Green<br>On<br>Off | Data transfer<br>PBX network connection is established<br>Waiting for network connection |
| **LAN** | Blinking Green<br>On<br>Off | Data transfer<br>PBX network connection is established<br>Waiting for network connection |
| **FXO** | Steady Red<br>Flashing<br>Off | Ready/Standby<br>Ringing<br>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 10/100/1000BASE-T 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<br><br>**-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<br><br>**-GSM module** is connected to Global System for Mobile Communications (GSM) with SIM card |

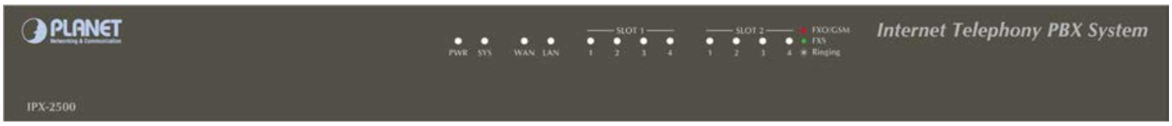
| Note | Supporting 2 slots, user can buy expansion module like IPX-21FO (4FXO) or IPX-21GS (4GSM) for extending port service. |
|---|---|

## 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<br>Off | PBX Power ON<br>PBX Power OFF |
| **SYS** | Blinking Green<br>On<br>Off | System is working<br>System doesn't boot<br>System failure |
| **WAN** | Blinking Green<br>On<br>Off | Data transfer<br>PBX network connection is established<br>Waiting for network connection |
| **LAN** | Blinking Green<br>On<br>Off | Data transfer<br>PBX network connection is established<br>Waiting for network connection |
| **FXO** | Steady Red<br>Flashing<br>Off | Ready/Standby<br>Ringing<br>Module not available |
| **GSM** | Steady Red<br>Flashing<br>Off | Ready/Standby (SIM card inserted)<br>Ringing<br>No SIM card inserted |
| **FXS** | Steady Green<br>Flashing<br>Off | Ready/Standby<br>Ringing<br>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 support auto negotiating Fast Ethernet 10/100/1000BASE-T 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<br><br>-**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<br><br>-**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. |

## 1.3 Specifications

| | | |
|---|---|---|
| **Product** | IPX-2200<br>Internet Telephony PBX system<br>(200 SIP Users registrations) | IPX-2500<br>Internet Telephony PBX system<br>(500 SIP Users registrations) |
| **Hardware Specifications** | | |
| WAN | 1 x 1000BASE-T RJ45 for WAN, connecting to broadband modem or a WAN router | |
| LAN | 1 x 1000BASE-T 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<br>SYS: 1, LNK/Off<br>WAN: 1, LNK/Off<br>LAN: 1, LNK/Off<br>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<br>RFC 826 ARP<br>RFC 1034, 1035 DNS<br>RFC 2068 HTTP<br>RFC 2131 DHCP<br>RFC 2516 PPPoE<br>RFC 3261, RFC 3311, RFC 3515<br>RFC 3265, RFC 3892, RFC 3361<br>RFC 3842, RFC 3389, RFC 3489<br>RFC 3428, RFC 2327, RFC 2833<br>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)<br>Note: T.38 support is dependent on fax machine, SIP provider and network/transport resilience. | |
| Voice Processing | DTMF detection and generation<br>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<br>• Up to 200 IP phone registers/extensions<br>• Recording and Voicemail (GSM/default): 1500 hours Wav: 150 hours | • 100 concurrent call legs<br>• Up to 500 IP phone registers/extensions<br>• 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<br>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)**

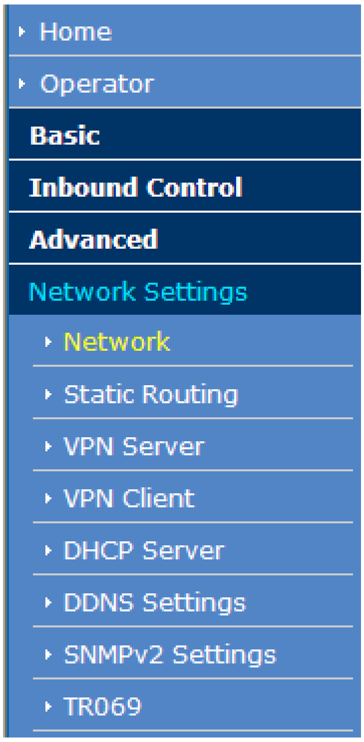
| |
|---|
| Default LAN IP: https://**192.168.0.1** |
| Default WAN IP: https://**172.16.0.1** |
| Default User Name: **admin** |
| Default Password: **admin** |

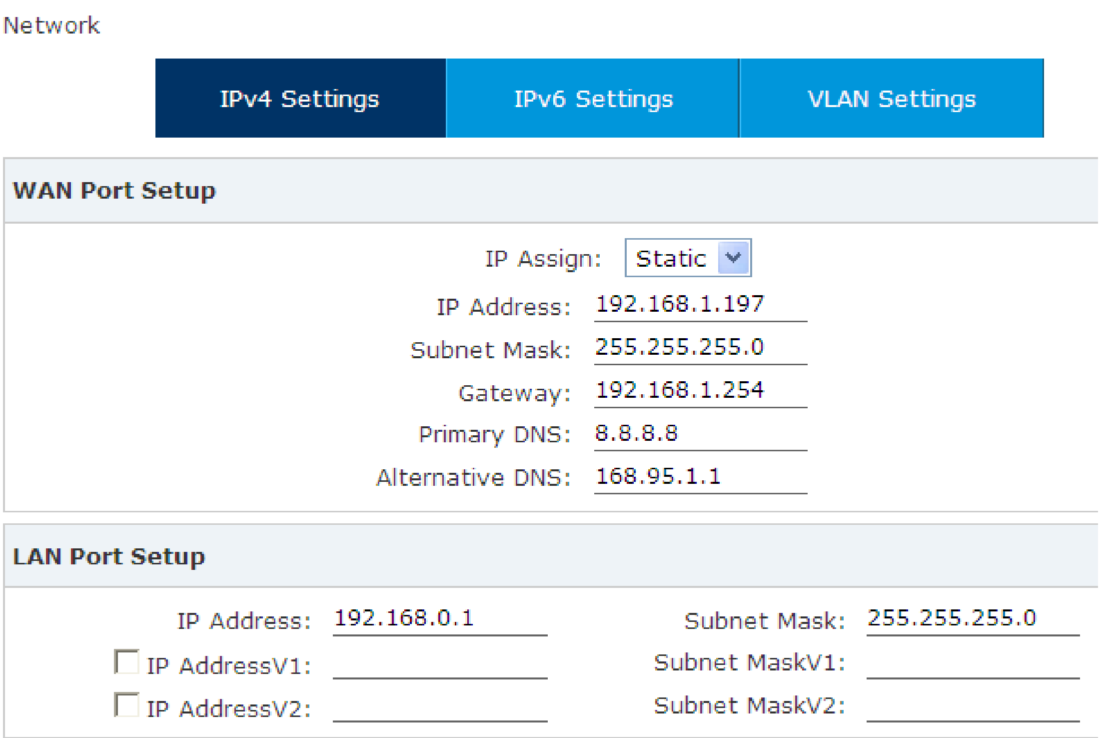

**Figure 2-1. Login page of the IPX-2200/IPX-2500**

| 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.

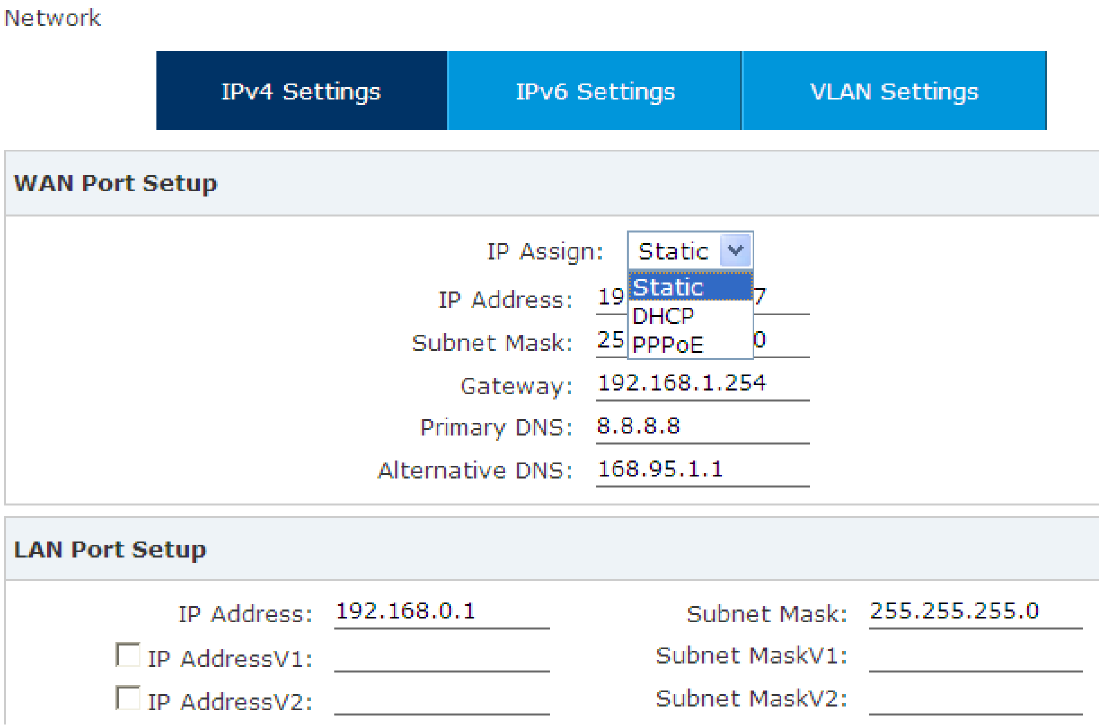



**Figure 2-4. Selection of IP Connection Type**

# 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**

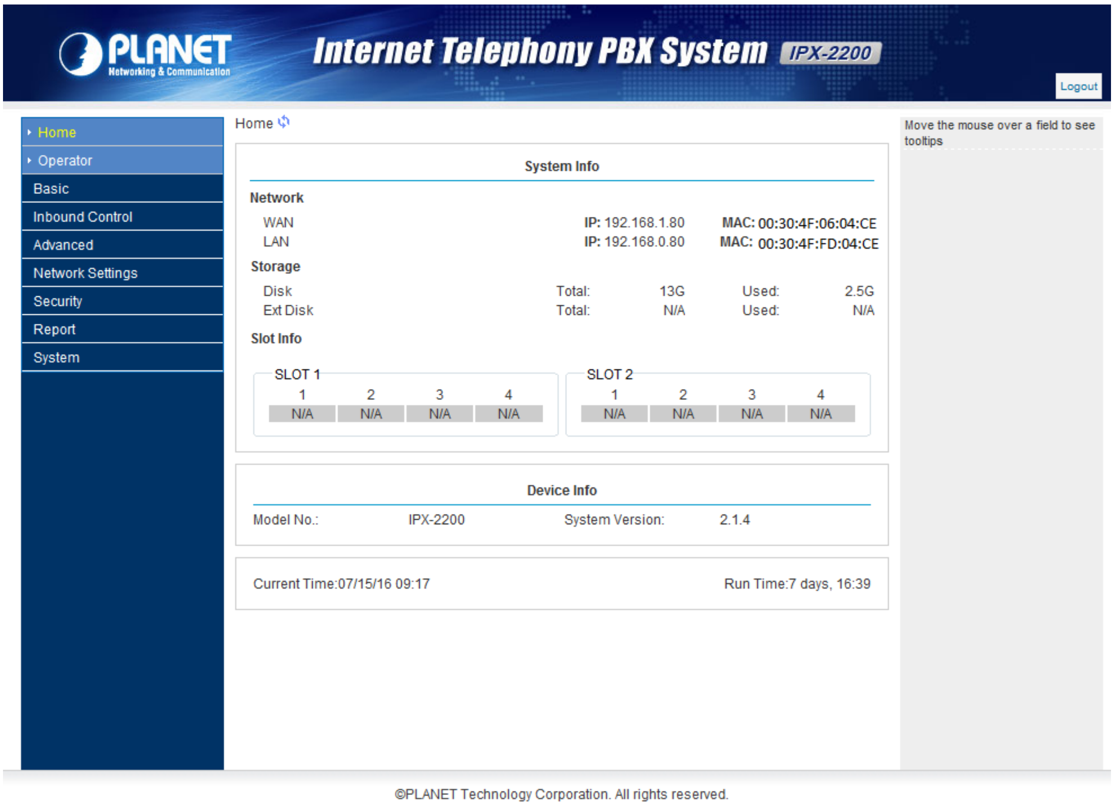
Default User Name: **admin**

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.<br>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.<br>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:

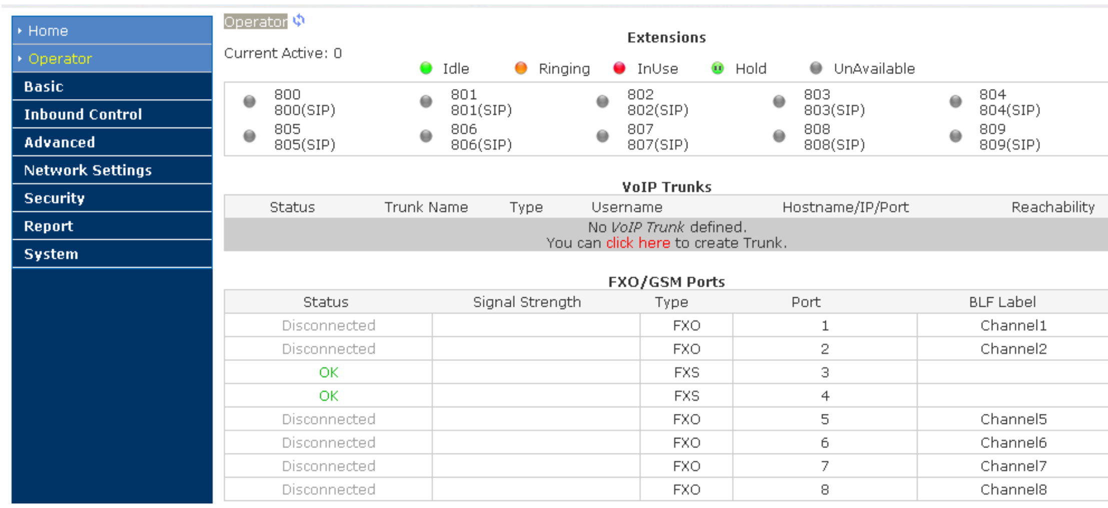
| | | |
|---|---|---|
| **1** | **Logout** | Logout the Web panel |
| **2** | **Activate Change** | Activate the changes for your current configuration |

### 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



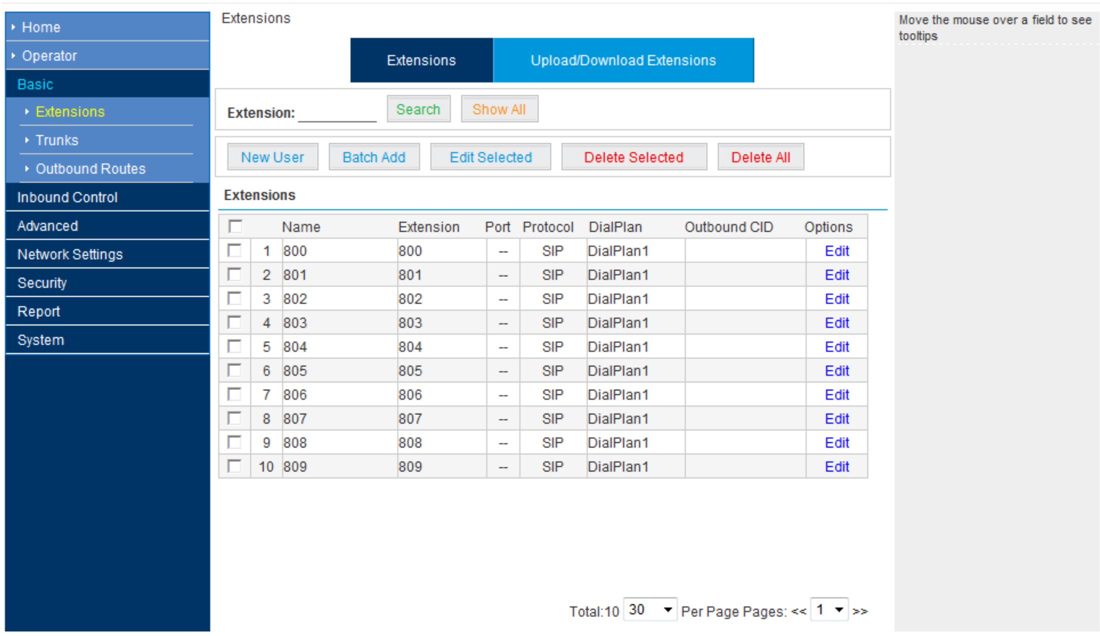
Display all the Extension, VoIP Trunk and Slot information.

About extension:

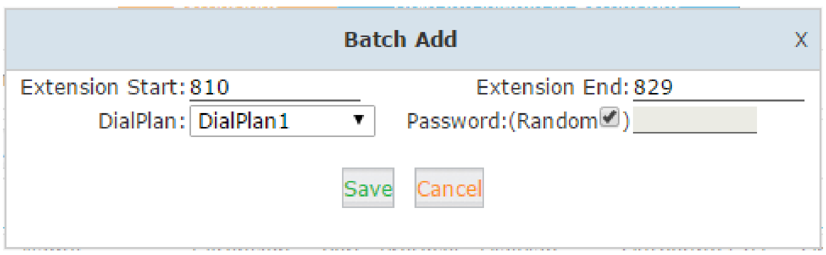
| | | |
|---|---|---|
| **1** | | Idle |
| **2** | | Ringing |
| **3** | | In use |
| **4** | | Hold |
| **5** | | Unavailable |

## 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**

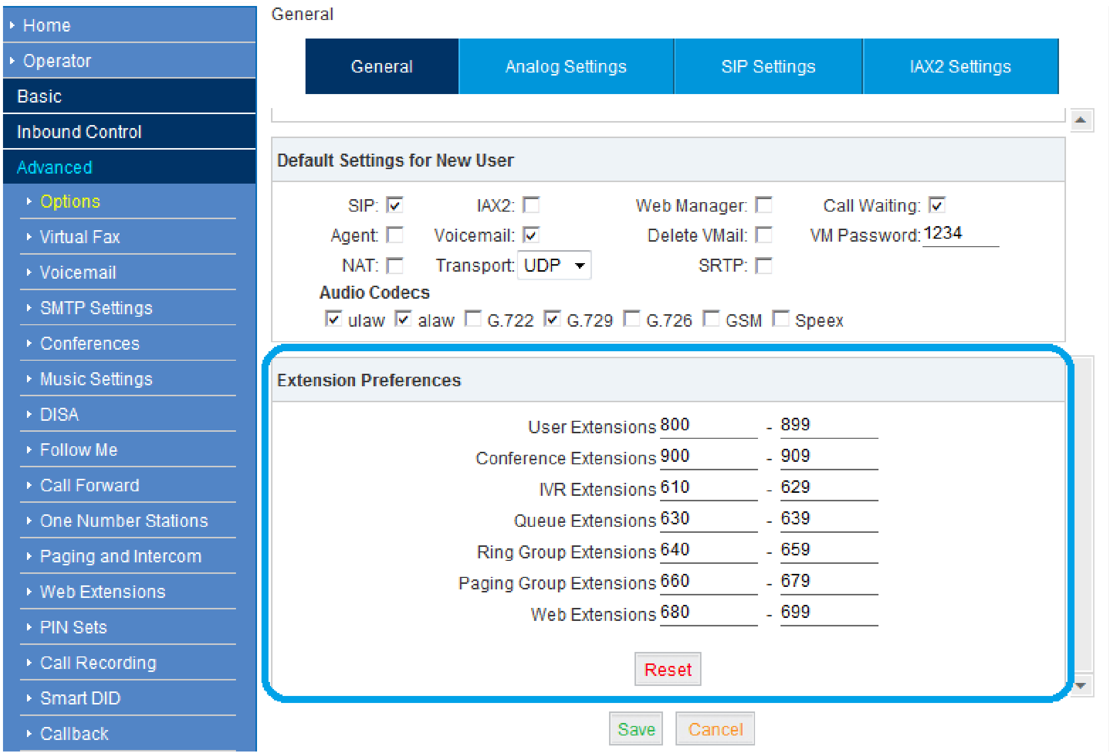
| Item | Explanation |
|---|---|
| Extension Start/Extension End | These two fields define the new extension range to be generated. |
| Dial Plan | Select a same dial plan for these new extensions. |
| Password | Can be different random passwords consisting of numbers, letters and special characters (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.

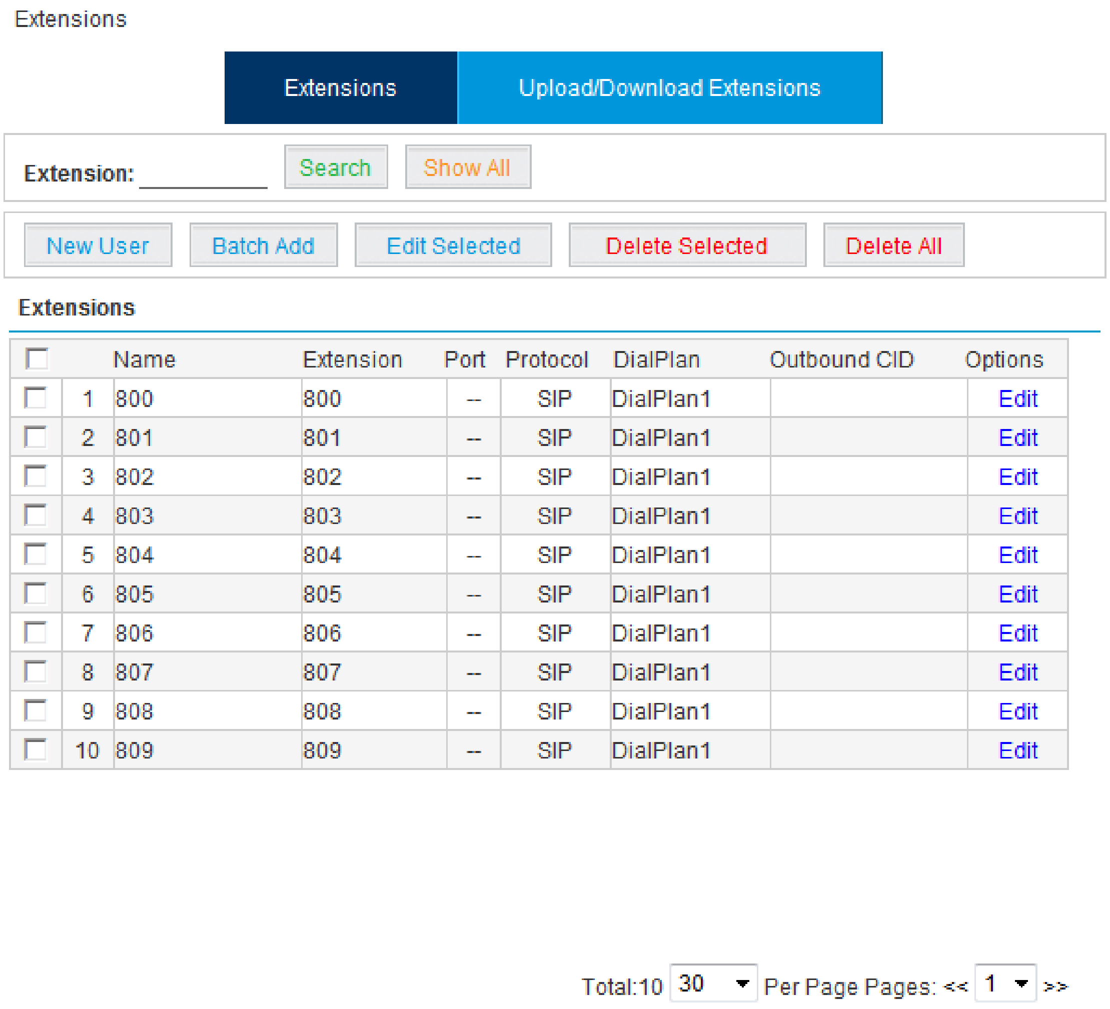
Please navigate to web menu *Advanced->Options->General.*

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



**Configure Extensions**

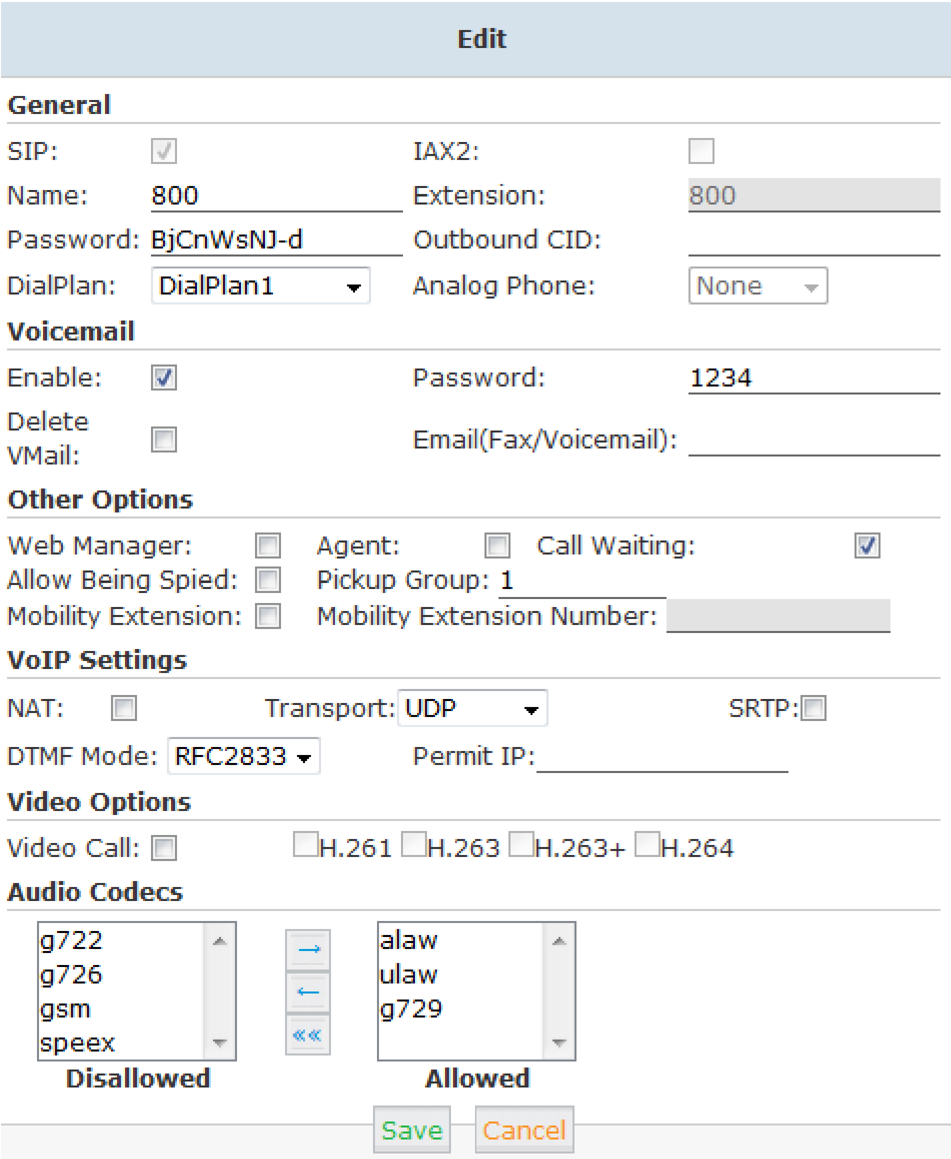
Planet IP PBX supports SIP/IAX2 and analog extension; configure extension on this page: 【Basic】----【Extensions】




| ☐ | | Name | Extension | Port | Protocol | DialPlan | Outbound CID | Options |
|---|---|---|---|---|---|---|---|---|
| ☐ | 1 | 800 | 800 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 2 | 801 | 801 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 3 | 802 | 802 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 4 | 803 | 803 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 5 | 804 | 804 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 6 | 805 | 805 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 7 | 806 | 806 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 8 | 807 | 807 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 9 | 808 | 808 | -- | SIP | DialPlan1 | | Edit |
| ☐ | 10 | 809 | 809 | -- | SIP | DialPlan1 | | Edit |

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. |

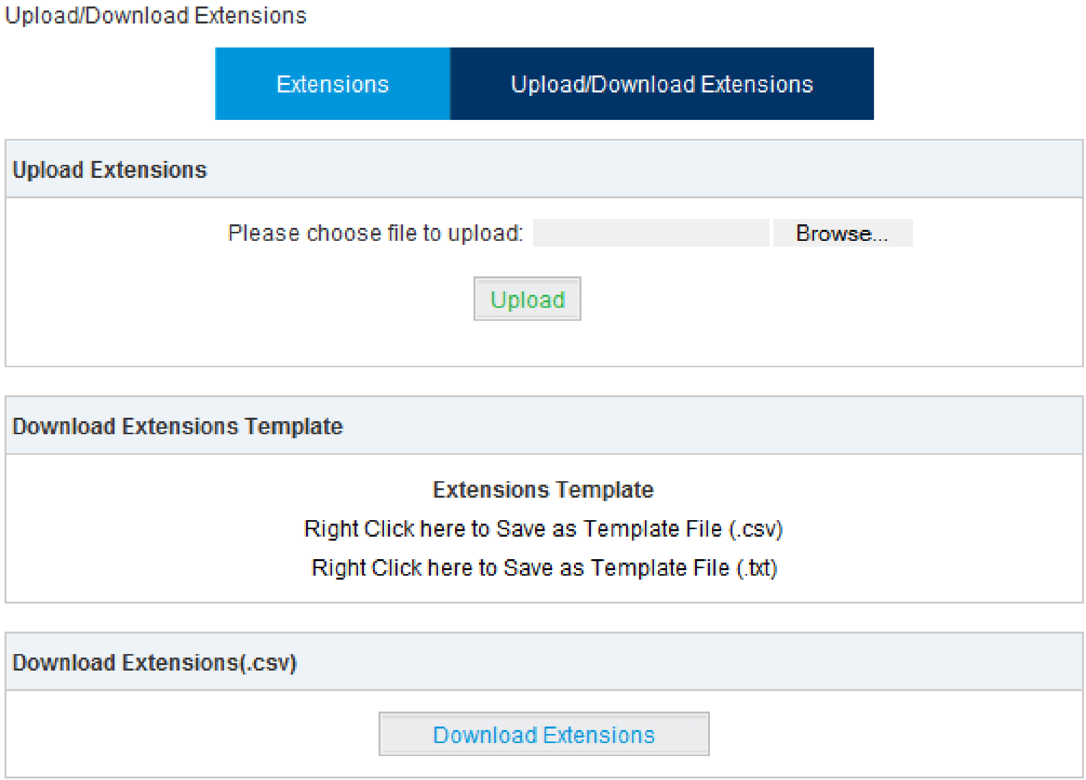
| Item | Explanation |
|---|---|
| Voicemail | Select this option to open the voicemail account |
| VM Password | Set password for Voicemail, e.g. "1234" |
| Delete VMail | Check this option to delete voicemail from system after it's sent to mail box. |
| Email (Fax/Voicemail) | Extension user's mail 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:

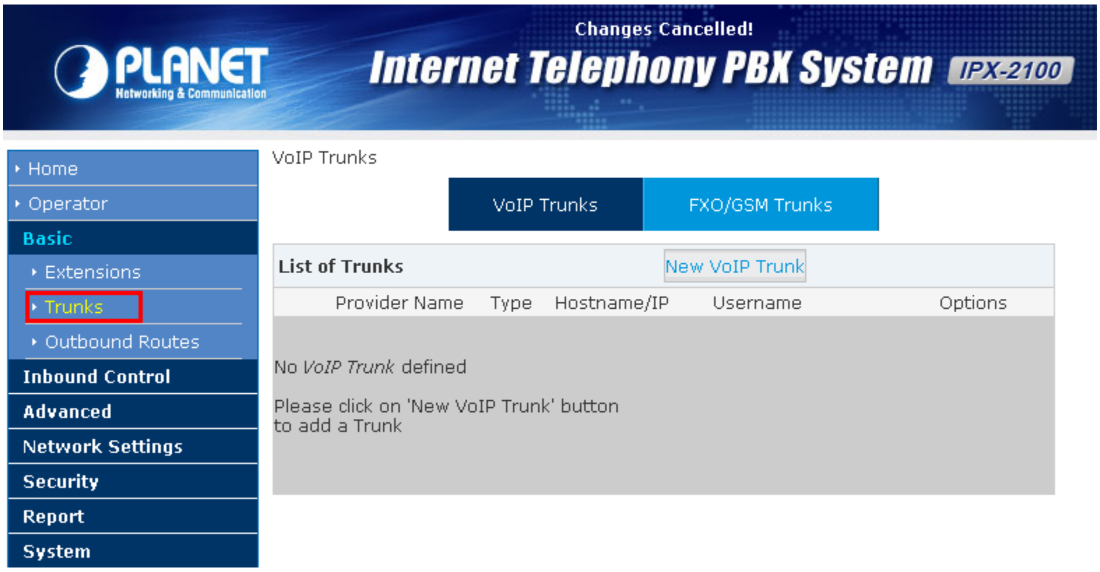



- Upload Extensions: Here you can upload .csv or .txt file to generate extensions.
- Download Extensions Template: Here you can download a template file in .csv or .txt format. Inside there are examples given, you can follow the examples to add your desired new extensions in the same format, and the 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

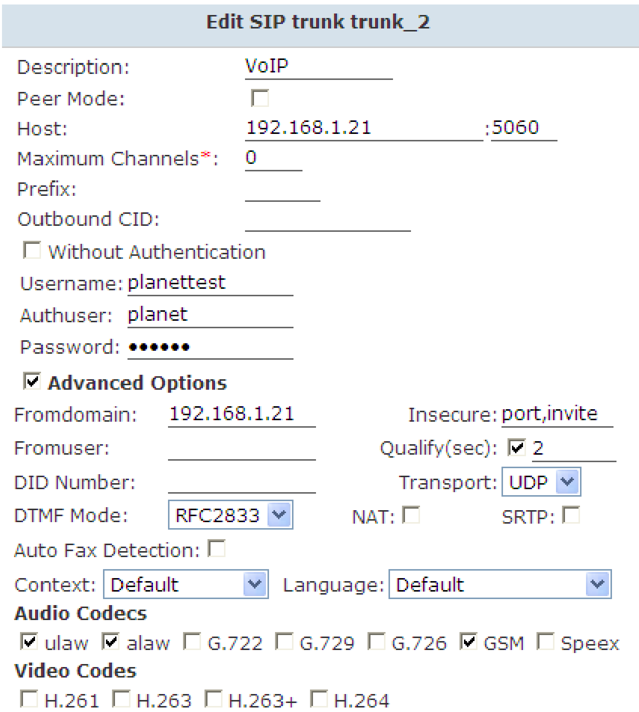
If you want to set up outbound call to connect to PSTN (Public Switch Telephone Network), GSM (Global System for Mobile Communications) or VoIP provider, please configure on this page: 【Basic】->【Trunks】




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】:

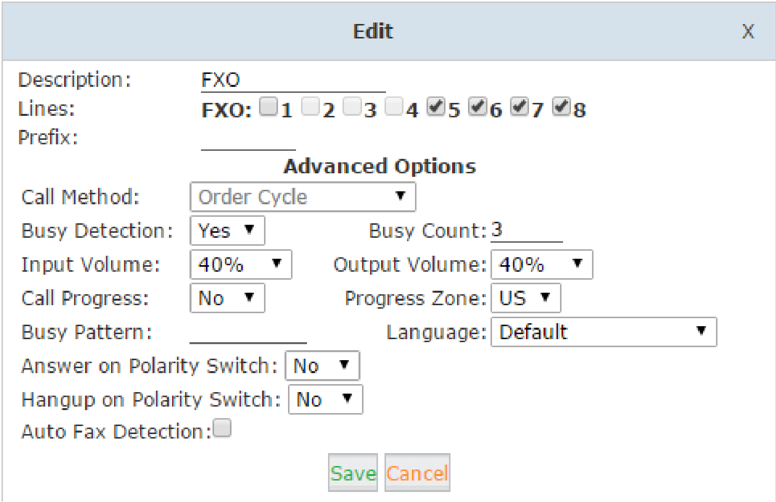
On the IPPBX front panel, red LED indicates the RJ11 interface is FXO. You should attach the telephone wire from your telecom to the FXO ports. Once connected you should be able to see the connection status on the *Operator* page in the "FXO/FXS/GSM Ports" section.

**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**

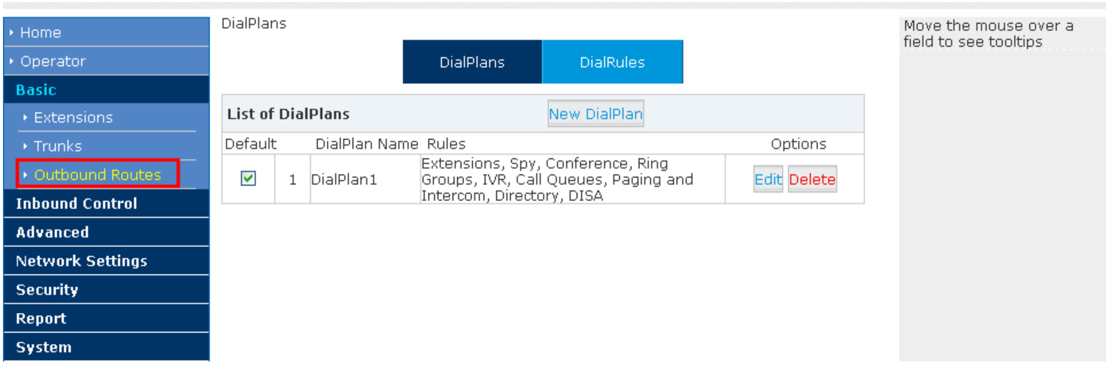
| Module | Working Frequencies |
|---|---|
| IPX-21GS | GSM/GPRS 850/900/1800/1900MHz |

## 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.

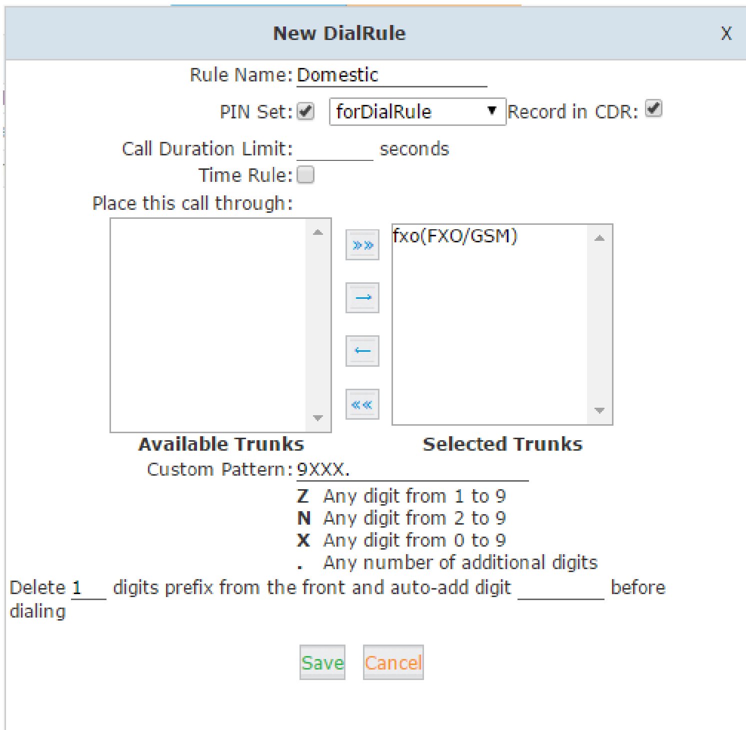
With the above mentioned possibilities, you may already have several trunks configured in the Planet IP PBX system. To be able to use different trunks for outbound phone calls, you'll have to configure several dial rules and maybe also several dial plans.

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




**Dial rules**

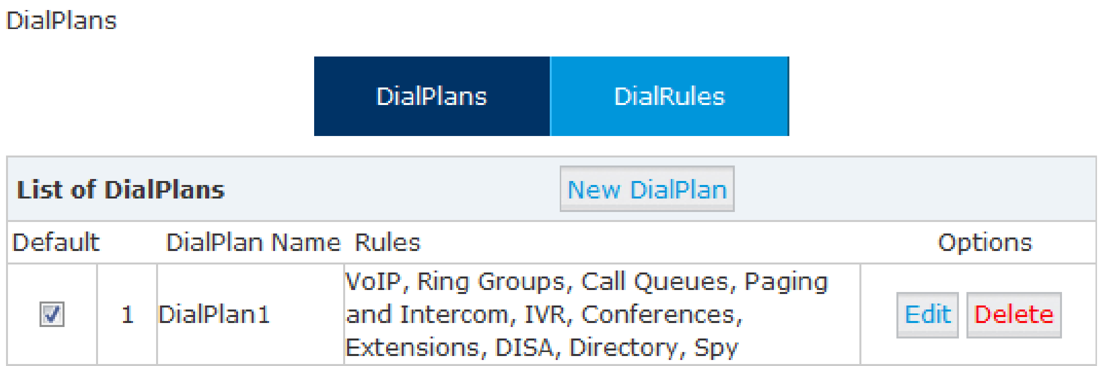
On this page, user can configure the basic match pattern of the outbound routes and create different dial plans. Please configure by clicking【Add a Dial Rule】




| 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. |

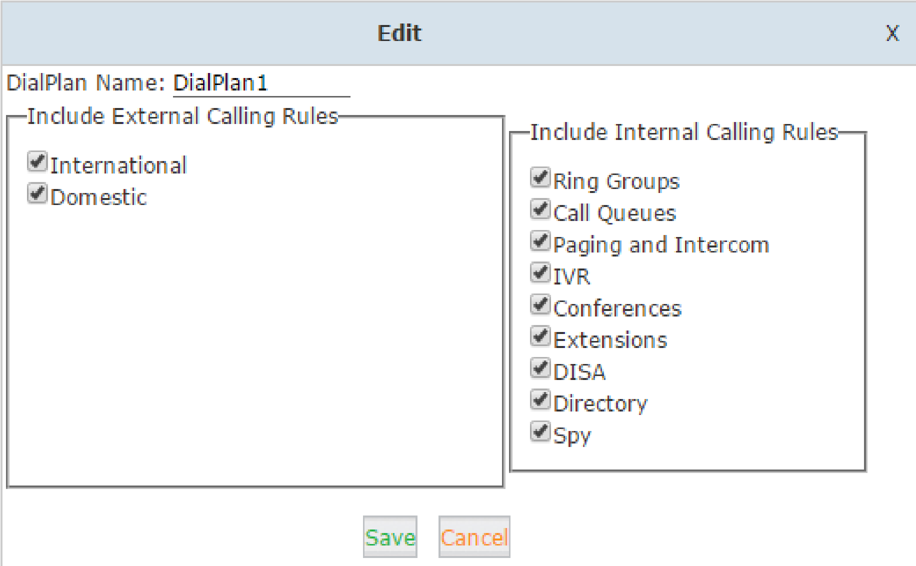
| Item | Explanation |
|---|---|
| Available Trunks | All existing trunks in the IPPBX system. |
| Selected Trunks | Trunk/Trunks can be used by this dial rule. |
| Custom Pattern | Dial patterns act like a filter for matching numbers dialed with trunks. The various patterns you can enter are similar to Asterisk's definition of them:<br>X — Refers to any digit between 0 and 9<br>N — Refers to any digit between 2 and 9<br>Z — Any digit that is not zero. (e.g. 1 to 9)<br>. — 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

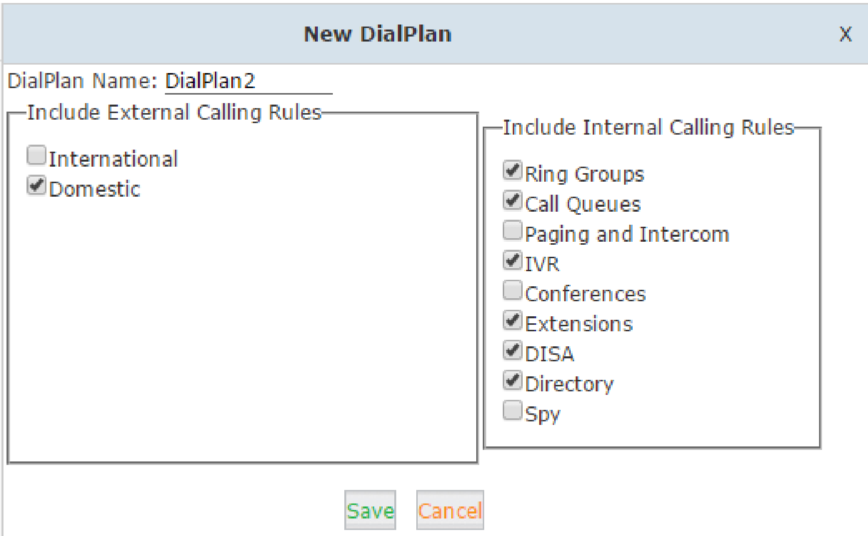



There's a default dial plan already existed in the IP PBX system. Normally you just have to click the "Edit" button on the default dial plan "DialPlan1" and tick on all dial rules to enable to the extension users to call any destinations using the trunk lines of the IP PBX system.

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.



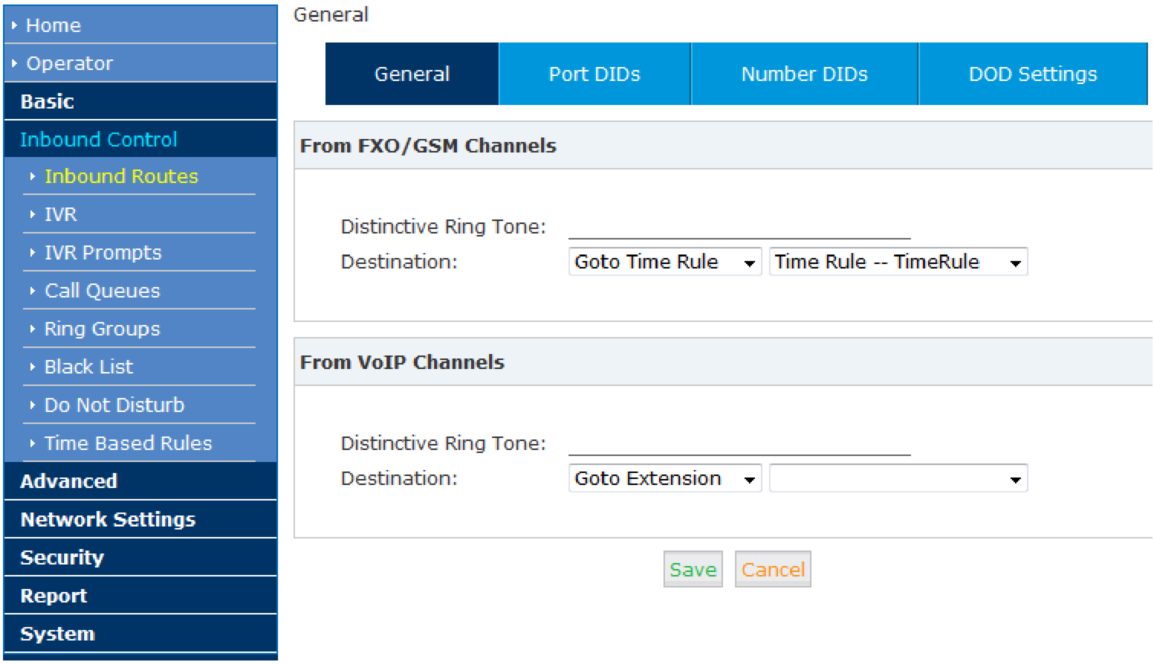

In the new dial plan you disable the rules you don't want others to use and save. After this on the extension configure page give them different dial plans; then they have different dial permissions.

# 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**

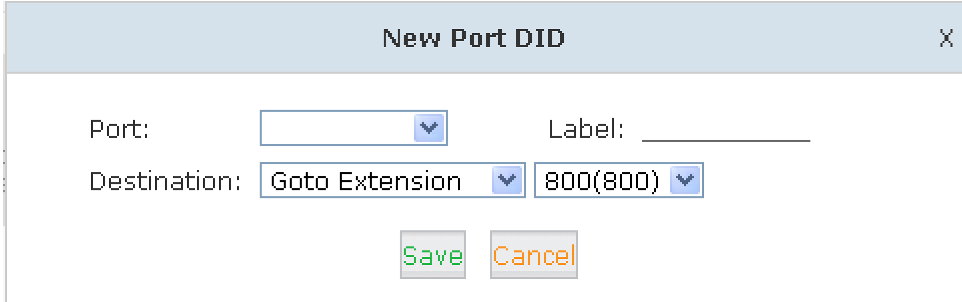
Distinctive Ring Tone: Mapping the custom ring tone file, e.g., set distinctive ring tone as "External", the phone will play this ring tone when receiving the call. Note: The phone must support such feature as well.

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】:

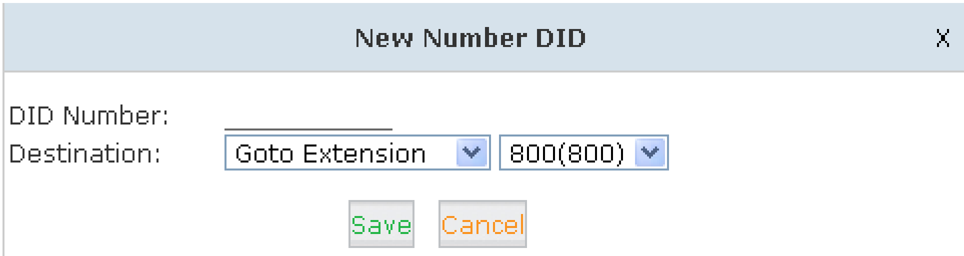



| Item | Explanation |
|---|---|
| Port | Select the port for outbound line. |
| Label | Set a label for this port. When incoming calls are from this port, the label will be displayed. |
| Destination | Incoming calls will access directly to this destination (extension user, call queue, conference, or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

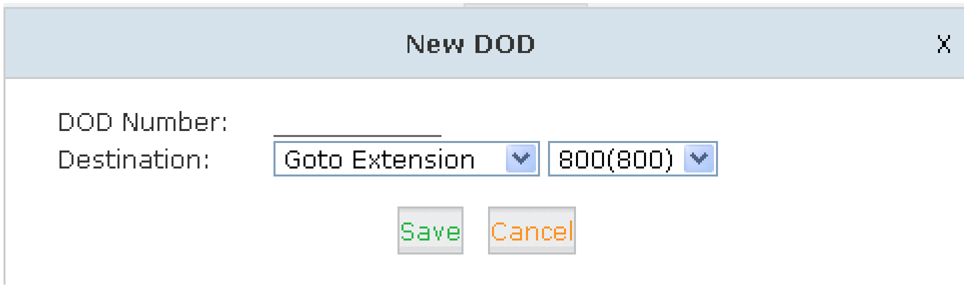
Click【Number DID】->【New Number DID】:




| 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】

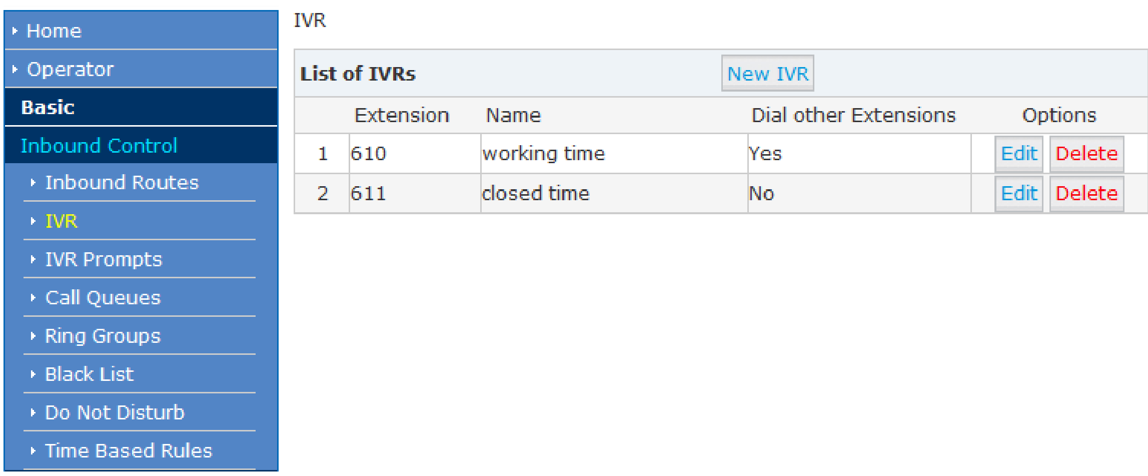



| Item | Explanation |
|---|---|
| DOD Number | Set the DOD number, and use it to match the Caller ID. If matched, the call will access to the defined destination. |
| Destination | Outbound calls will access directly to this destination (extension user, call queue, conference, or IVR). |

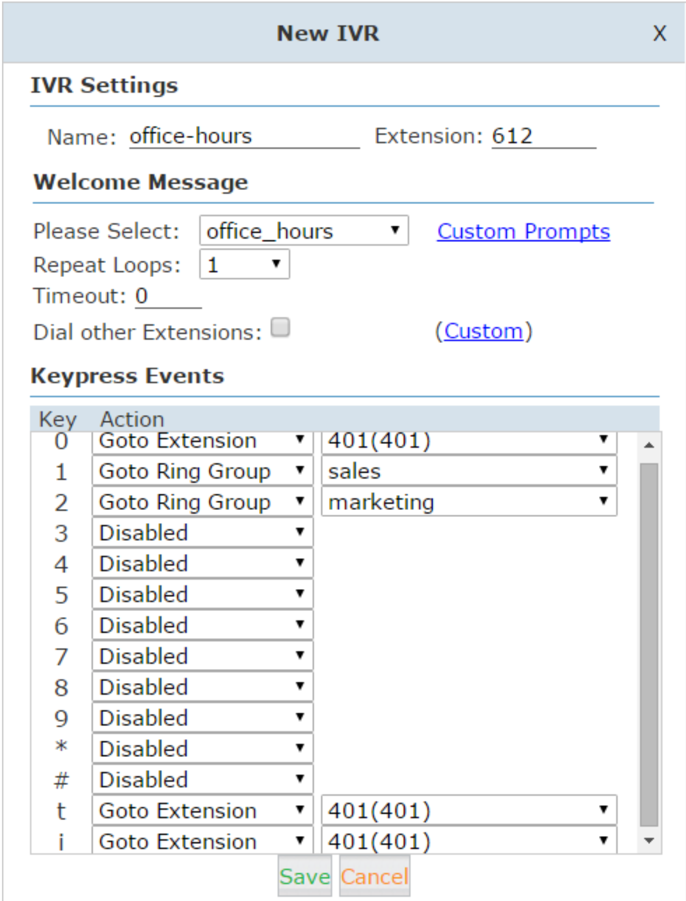
## 3.4.2 IVR

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:

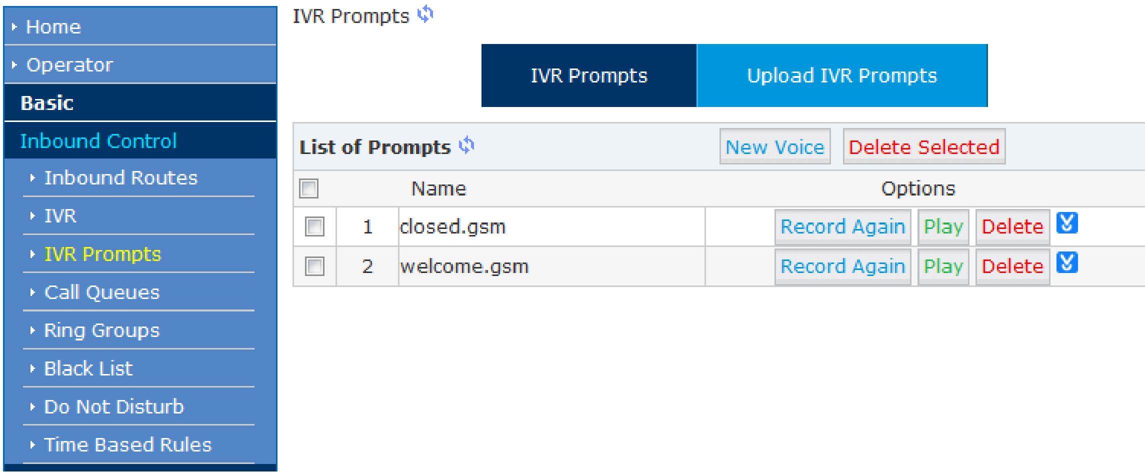



| Item | Explanation |
|---|---|
| Name | Set a name for the IVR |
| Extension | Extension number for the IVR, by calling this number can access the IVR menu. |
| Please Select | Select a voice prompt for this IVR menu. |
| Custom Prompts | Click this button to navigate to *Inbound Control->IVR Prompts* 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 |

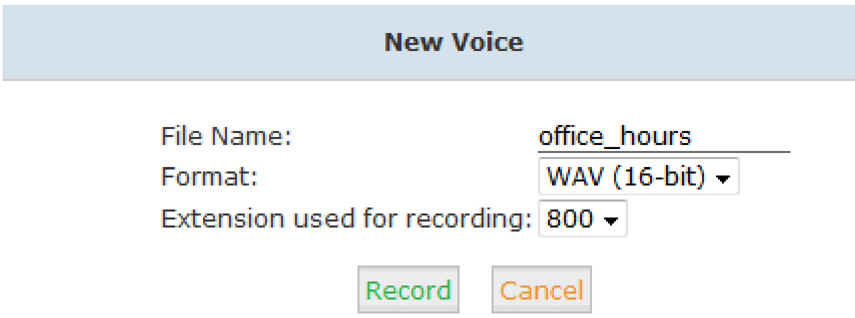
| Item | Explanation |
|---|---|
| | timeout, after all IVR loops played completely without pressing any key the incoming call will be handled by “t” parameter. |

### 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

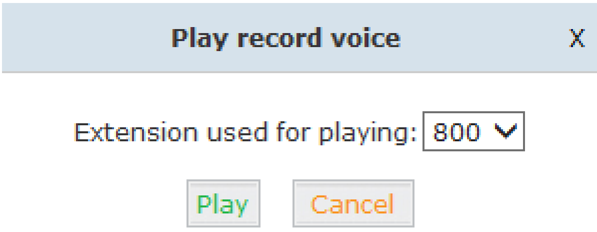



Click【IVR Prompts】----【New Voice】to create new IVR prompt:



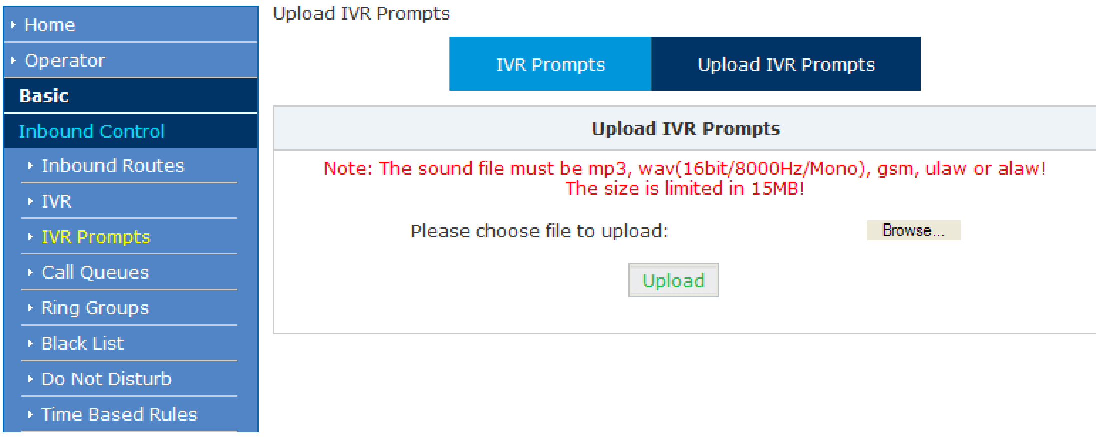

| Item | Explanation |
|---|---|
| File Name | Define a name for this voice file. |
| Format | Select the voice format, GSM/WAV (16bit) supported only. |
| Extension used for recording: | Select the extension which is used for recording the IVR prompt. Click【Record】, this extension will ring, and then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

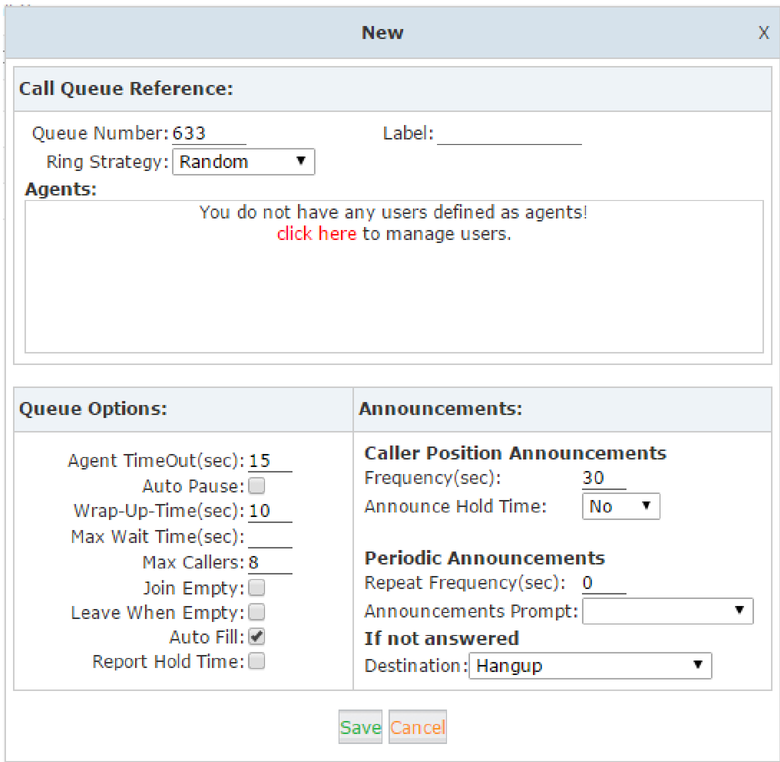



| | |
|---|---|
| Note | Uploading customized audio file must be in the mp3, wav, gsm, ulaw, alaw format, and size must be less than 15MB. |

## 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.

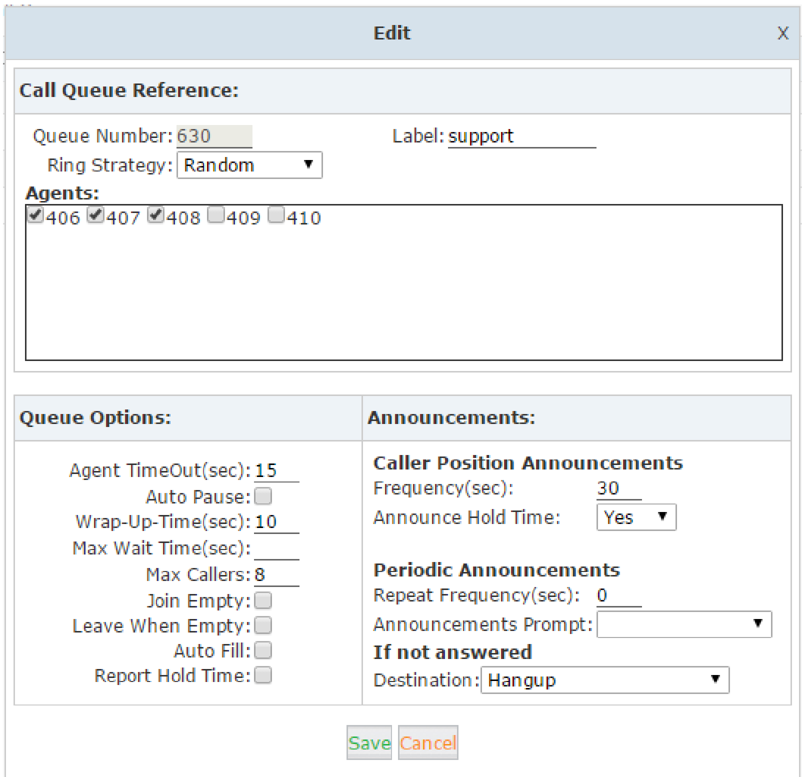
There are 3 existing call queues. All you have to do is click the “Edit” button to configure them. If you want more call queues, you can click “New Call Queue” to add more queues.




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 |

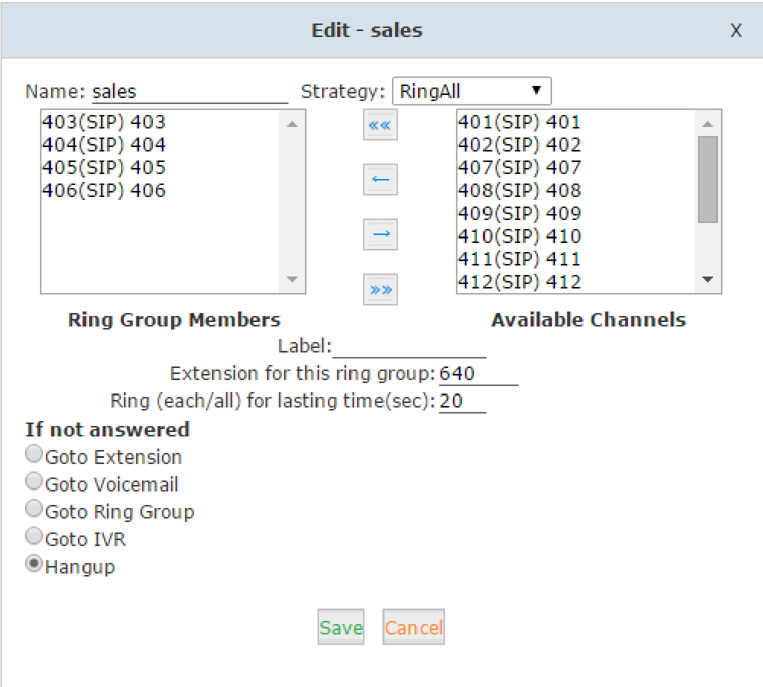
| Item | Explanation |
|---|---|
| | remembers which extension was called last and begins the round robin with the next agent. |
| Agent | Check each agent that is to be a member of this specific Call Center Queue. |
| Agent TimeOut (sec) | Specify the number of seconds to ring an agent's extension before 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】:



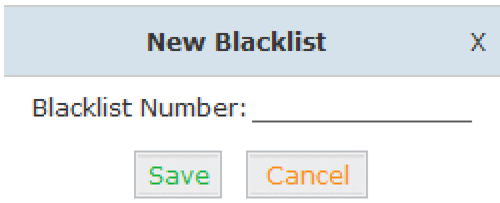

| Item | Explanation |
|---|---|
| Name | Define a name for the Ring Group. |
| Strategy | Define how to ring the group members; select “RingAll” will ring all the member extensions at the same time, select “Ring In Order” will ring the member extensions one by one. |
| Ring Group Members | The extensions 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. |

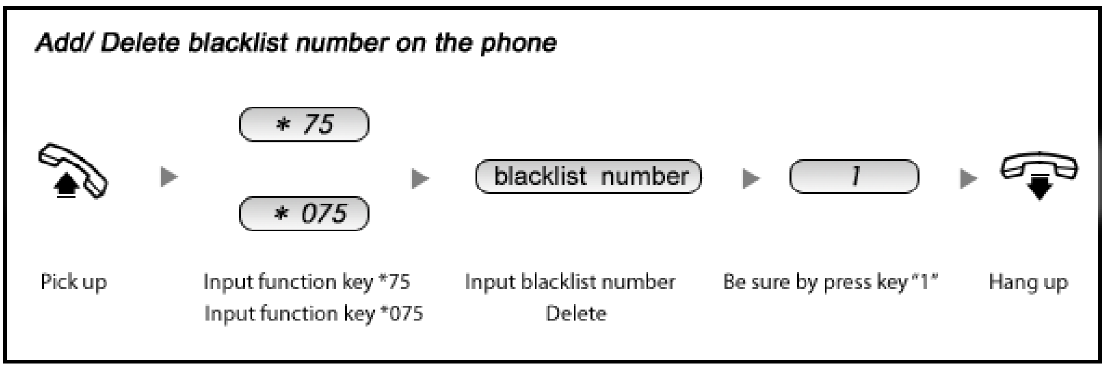
## 3.4.6 Black List

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】



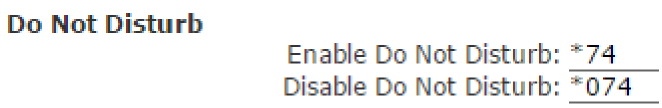

Input caller's number in the blank, then this caller's number will be blocked when the call comes again. Meanwhile, extension user can add or delete the blacklisted number by function key on the phone. Please operate according to the following diagram:



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

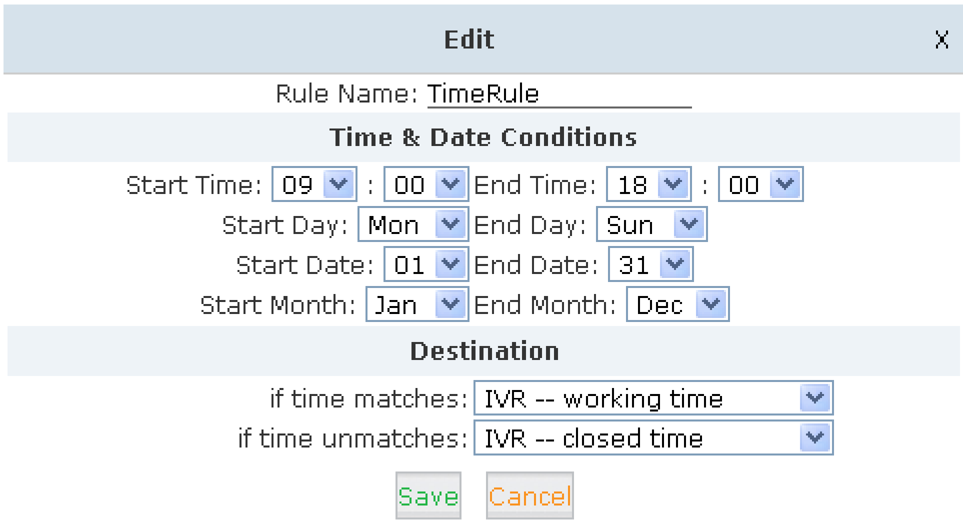



With Do Not Disturb (DND) feature enabled, an extension can make phone calls but others cannot call this extension. An extension user of the IP PBX system dials *74 from their phone, system will play a beep sound to indicate DND has been activated. To disable DND, just dial *074, another beep sound will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

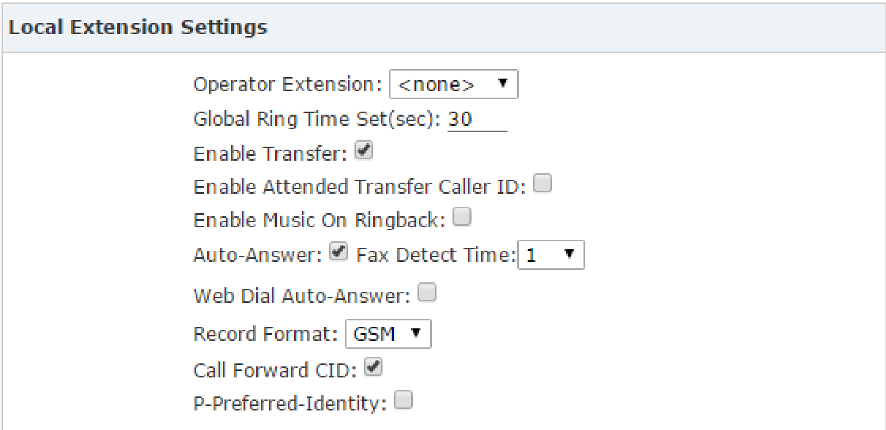
| Item | Explanation |
|---|---|
| | example, inbound call can be directed to operator in working time. |

# 3.5 Advanced

## 3.5.1 Options


### 3.5.1.1 General

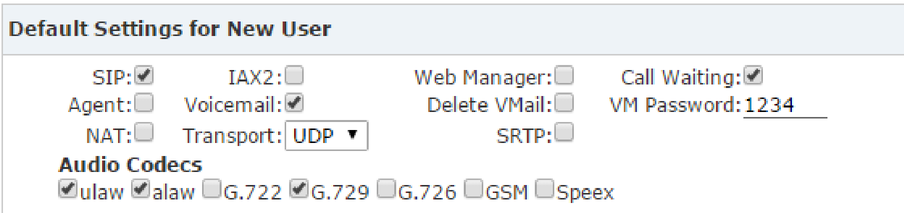
Here on this page you can configure some global options for all the user extensions. In the "Local Extension Settings" section you have the options shown below that can be configured.




| 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. |

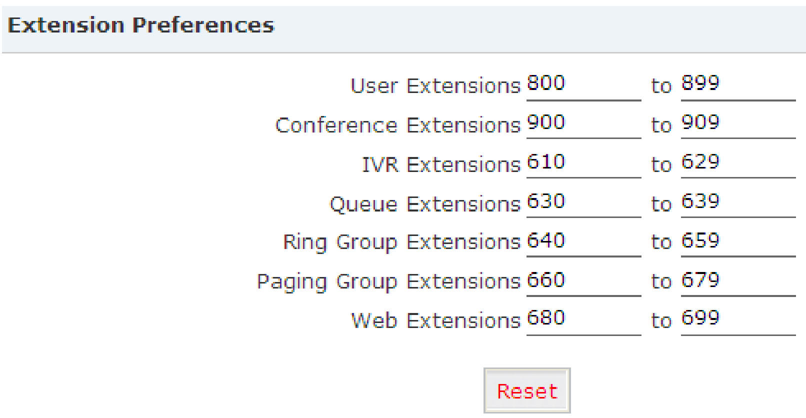
| Item | Explanation |
|---|---|
| Enable Music On Ringback | If enabled this option, callers will hear music instead of ringback tone while calling other extensions. |
| Auto-Answer | Auto answer enables the IPPBX to automatically answer the inbound calls from analog ports. |
| Fax Detect Time | If 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**

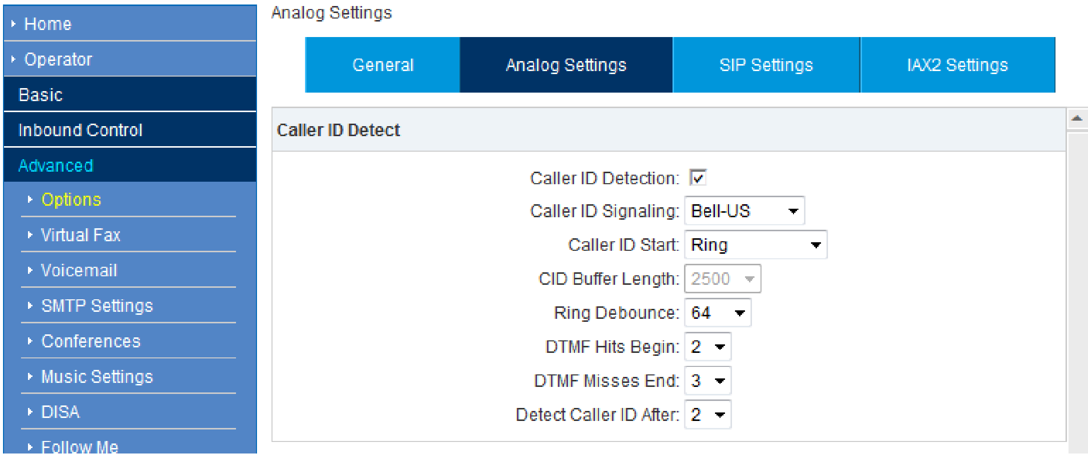



The user extension number and system extension number ranges are defined here to avoid confusion of the numbers in the IP PBX system. You can modify these number ranges according to your real applications.

### 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.

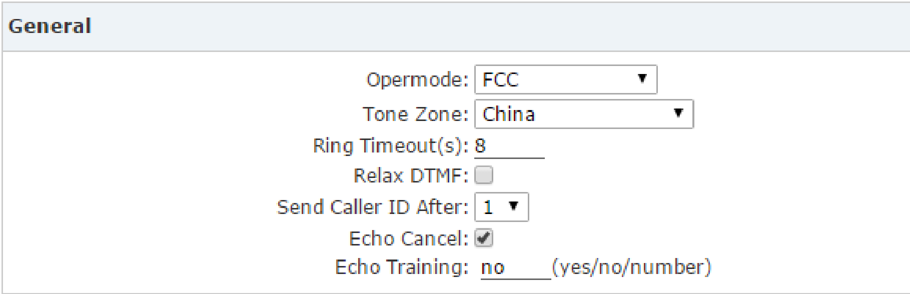
**Caller ID Detect**




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.<br>Bell-US—Also known as BellcoreFSK. Used in the Canada, China, Hong Kong and US.<br>DTMF—Dual Tone Multi-Frequency. Used in Denmark, Finland and Sweden.<br>V23—Mostly used in UK.<br>V23-Japan—Mostly used in Japan. |
| Caller ID Start | When the caller ID starts.<br>Ring—Caller ID starts when a ring received.<br>Polarity—Caller ID starts when polarity reversal starts.<br>Polarity (India)—Can be used in India.<br>Before Ring—Caller ID starts before a ring received |
| CID Buffer Length | The buffer length can be used to store caller ID info. |

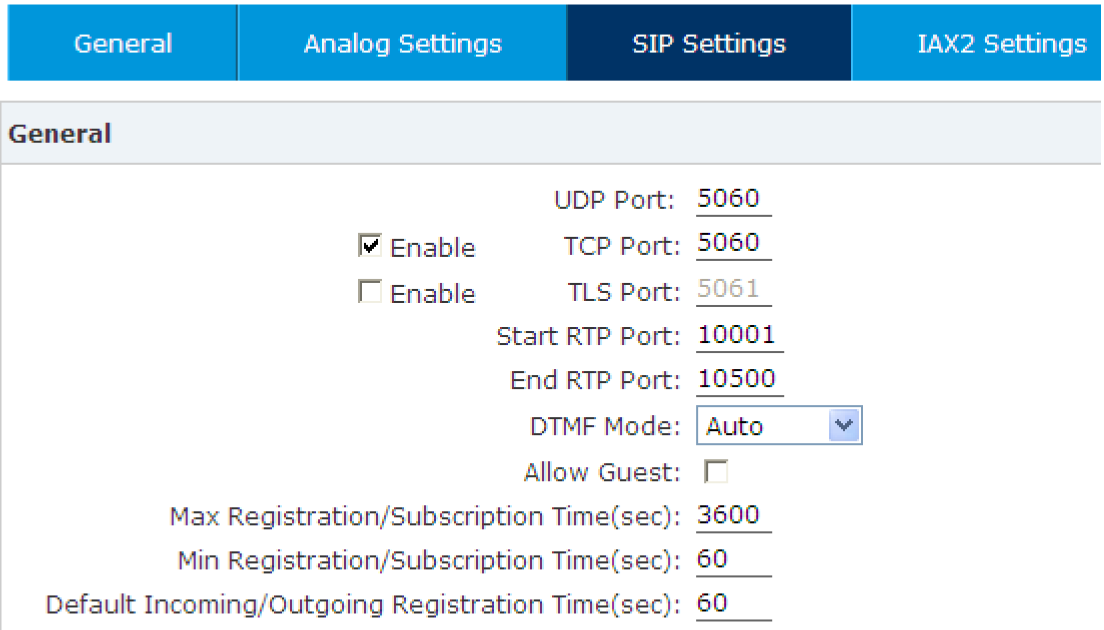
**General**



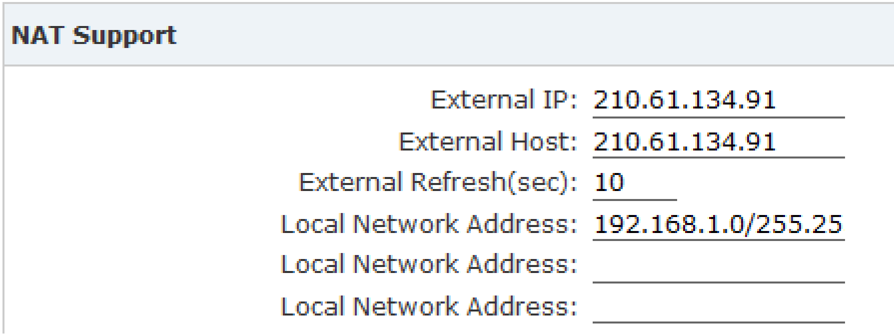

| Item | Explanation |
|---|---|
| Opermode | Set the Opermode for FXO Ports |
| ToneZone | Select the tone zone of your country. |
| Ring Timeout(s) | FXO (FXS signaled) devices must have a timeout to determine if there was a hangup before the line was answered. This value can be tweaked to shorten 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

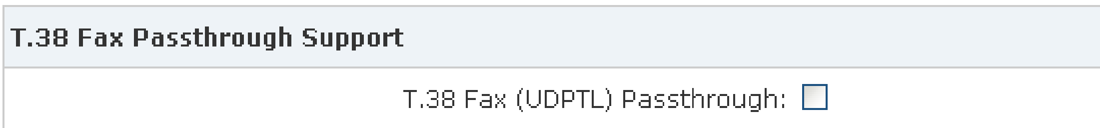
【Global SIP Settings】is appropriate for professionals. If anything needs to be modified, please contact our tech-support people.



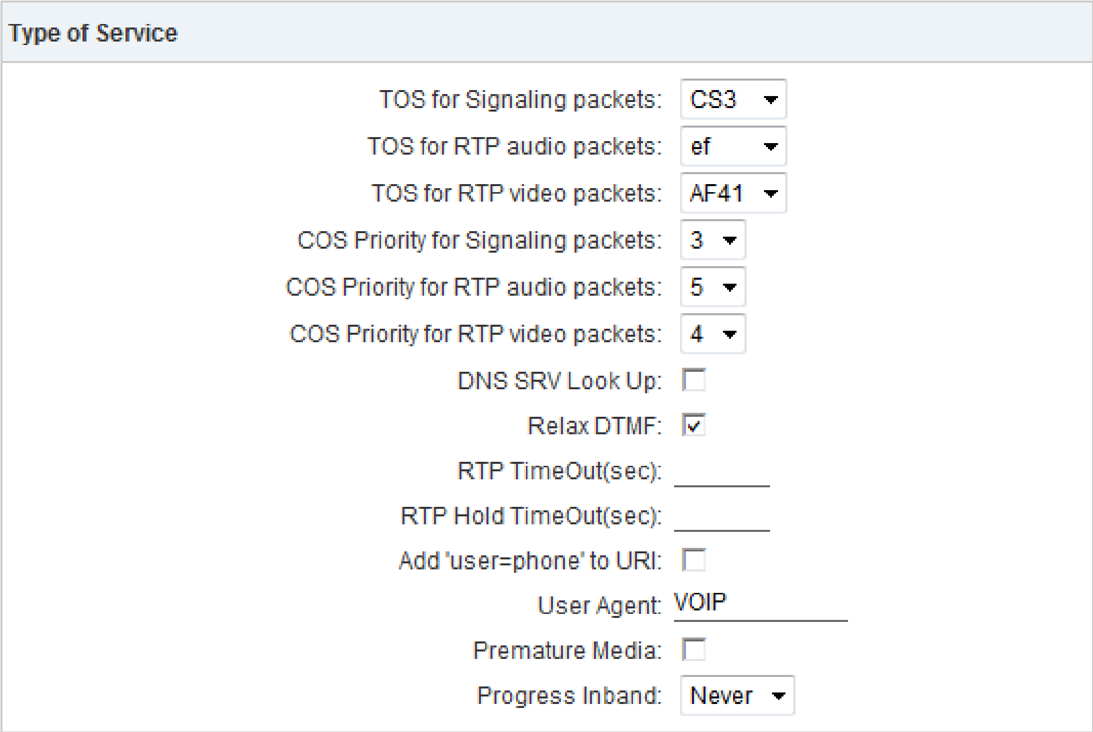

| Item | Explanation |
|---|---|
| UDP Port to bind to | SIP standard port is 5060 |
| TCP Port | Default TCP port is 5060 |
| TLS Port | Default TLS port is 5061 |
| Start RTP Port | RTP port range |
| End RTP Port | RTP port range |
| DTMF Mode | Set default DTMF mode for sending DTMF, support auto, 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 |

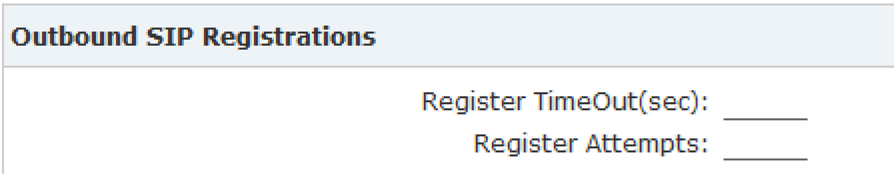



| Item | Explanation |
|---|---|
| T.38 fax (UDPTL) Passthrough | Enables T.38 fax (UDPTL) passthrough on SIP to SIP calls |




| 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 |

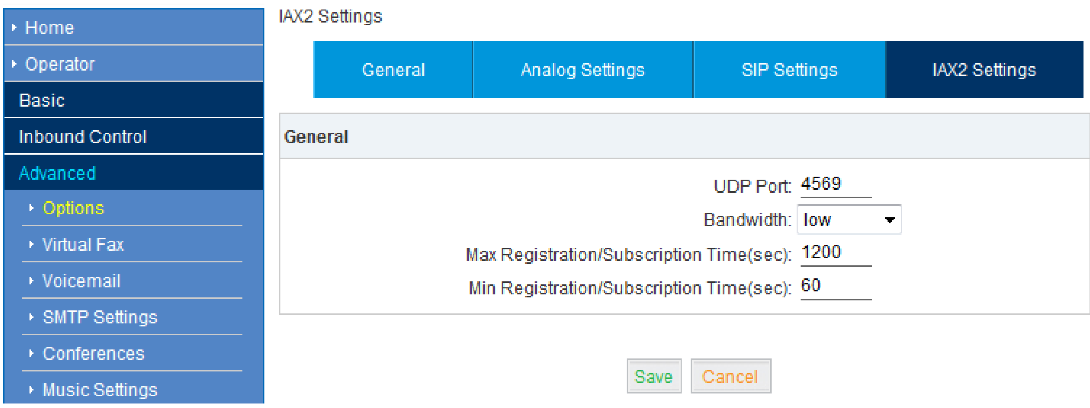
| Item | Explanation |
|---|---|
| | ";user=phone" on URI. |
| UserAgent | Allows you to change the user agent string. The default user agent string also contains the Asterisk version. If you don't want to expose this, change the user agent string here. |




| 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. |
|---|---|

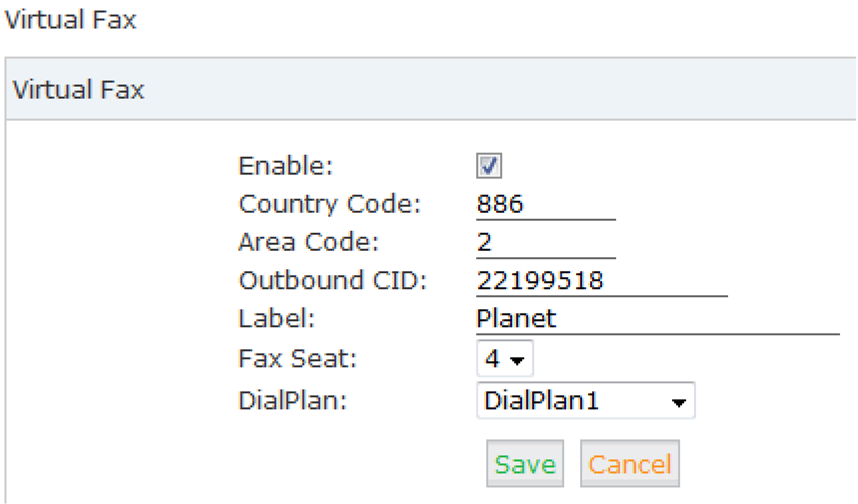
### 3.5.1.4 IAX2 Settings




| Item | Explanation |
|---|---|
| UDP Port | IAX2 signaling and media port, default is 4569. |
| Bandwidth | Specify bandwidth of low, medium, or high to control which codecs are used in general. |

| Item | Explanation |
| --- | --- |
| 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

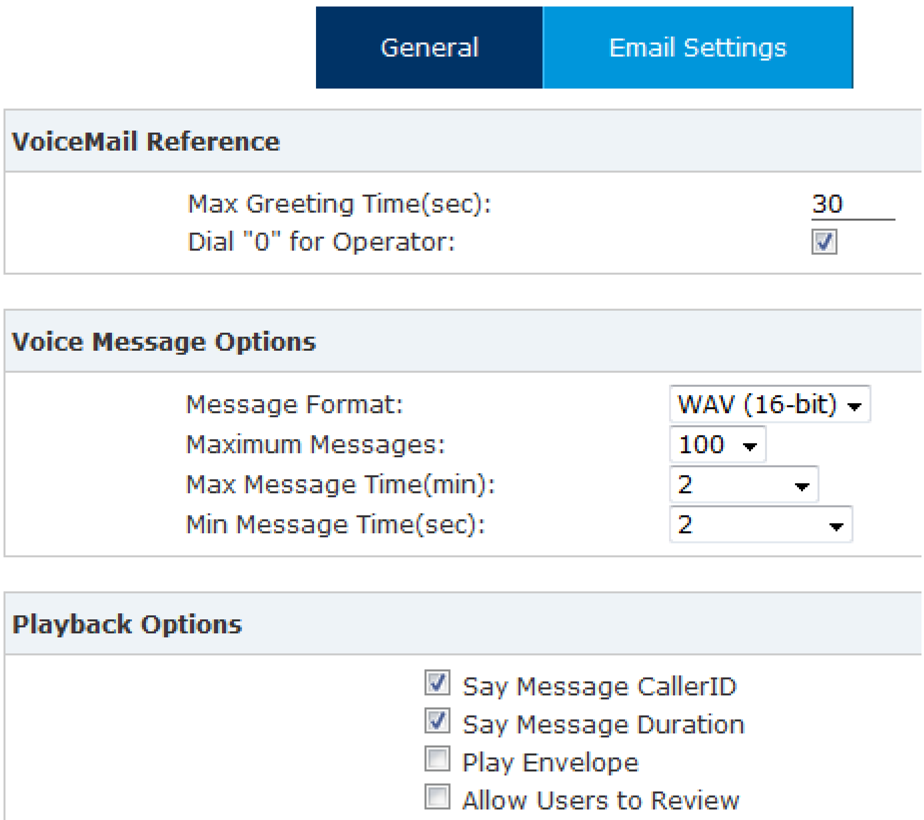



| Item | Explanation |
| --- | --- |
| Enable | Enable the following settings for outbound fax. |
| Country Code | Enter your country code here. (Optional). |
| Area Code | Enter your Area Code here. (Optional) |
| Outbound CID | Only works if the outbound fax goes out through VoIP trunks. The other 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:

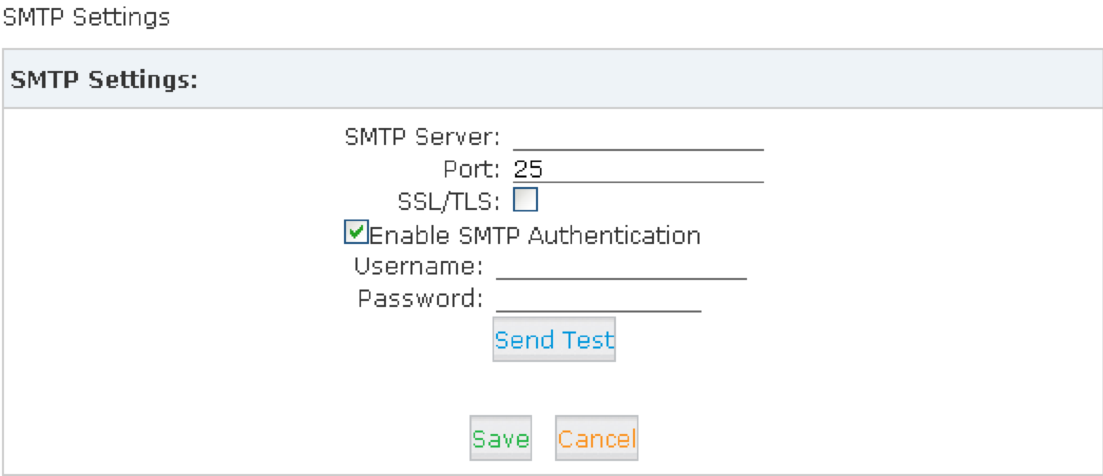
General




| 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. |

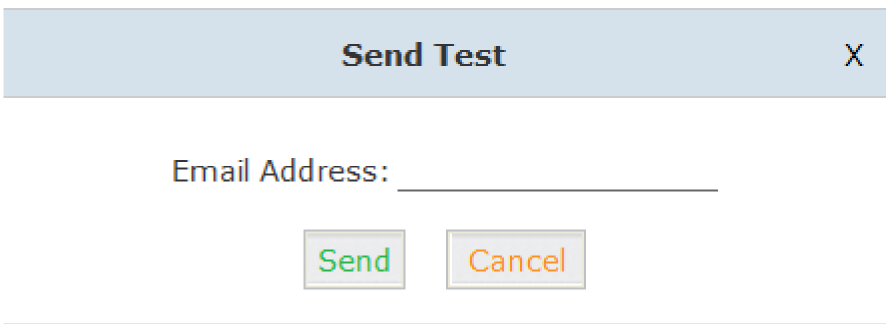
| Item | Explanation |
|---|---|
| Allow Users to Review | Check this option to allow users to review the voice message. |

### 3.5.4 SMTP Setting




| 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. |

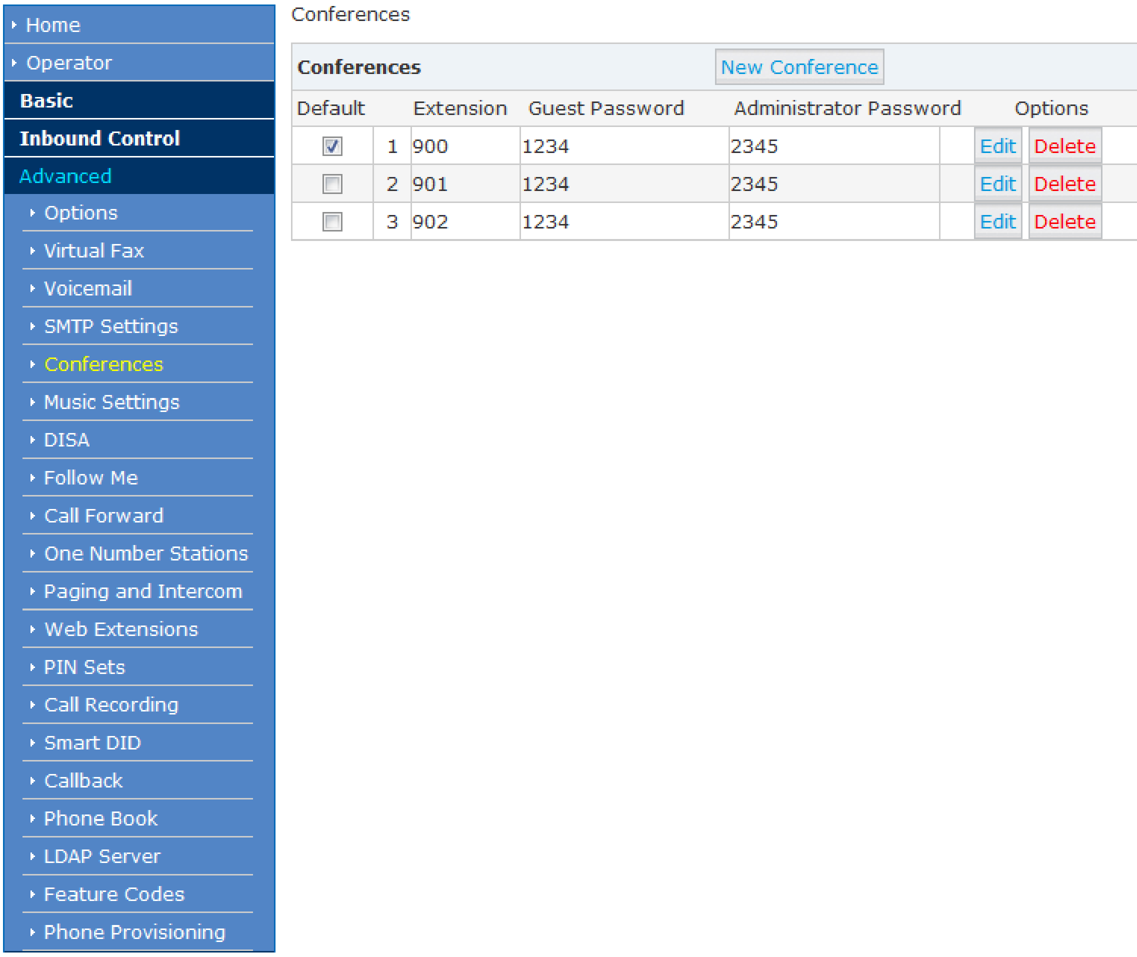
Click【Send Test】after configuration, the following diagram will be displayed to ask you to input the Email for receiving.

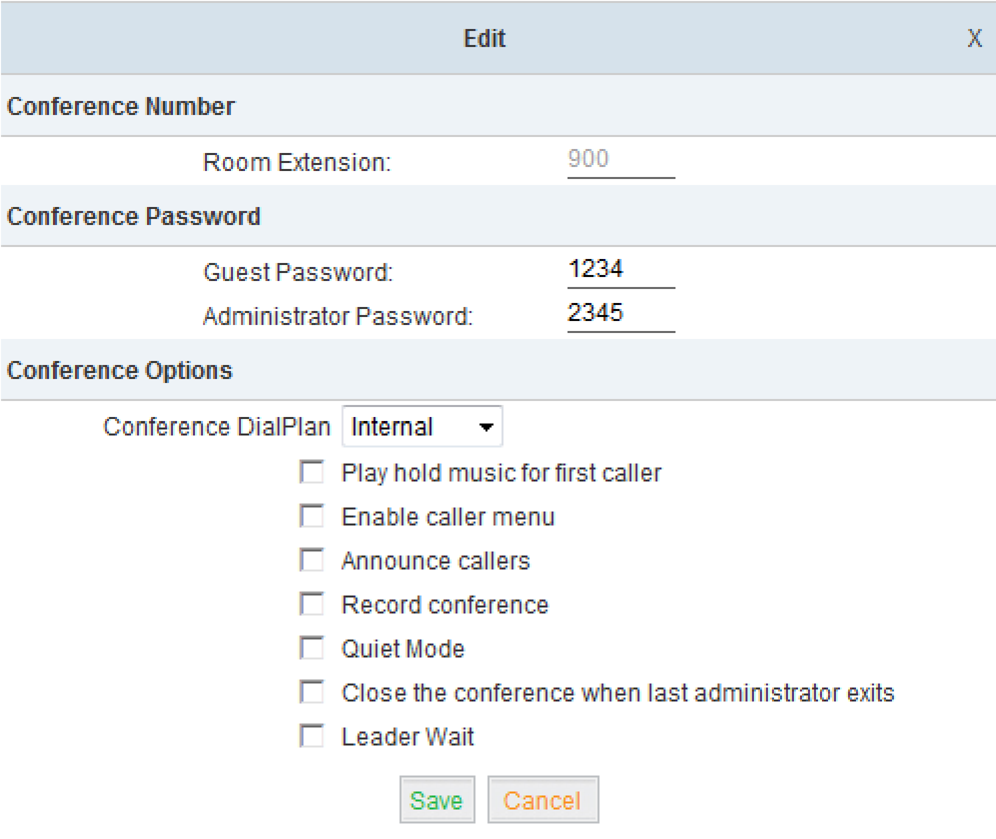


Input the Email and click【Send】to send the test email. Login your Email to check; configuration is successful if you receive the test email; otherwise, it fails. Please check your email settings.

## 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】:



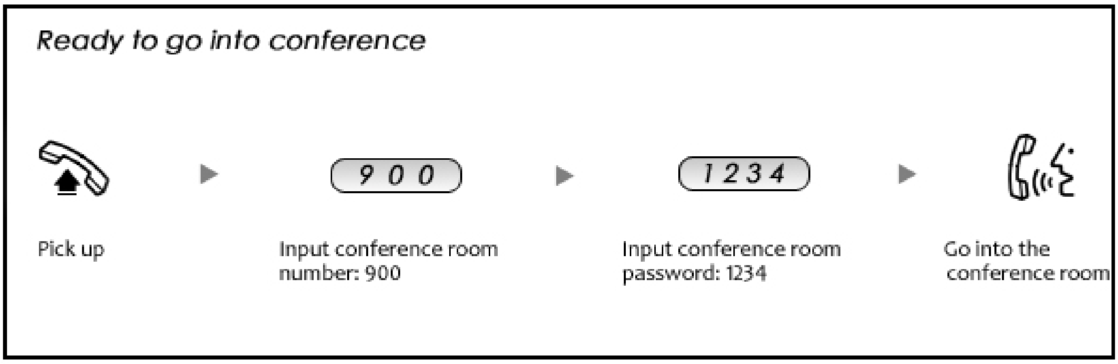



| Item | Explanation |
| --- | --- |
| Room Extension | By calling this extension number to enter the conference room |
| Guest Password | If the callers use this password to enter the conference then they are ordinary participants |
| Administrator Password | If the callers use this password to enter 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:

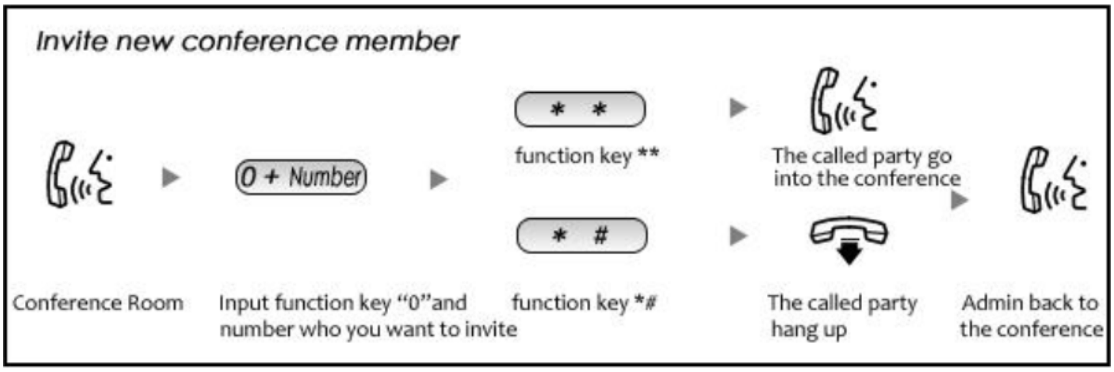
**Go to conference:**




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:

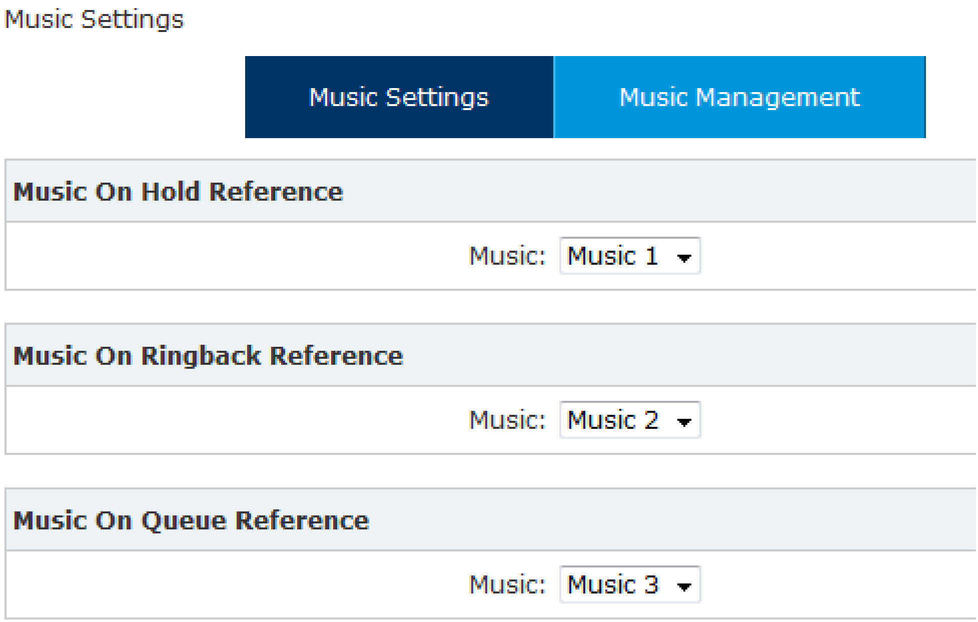


## 3.5.6 Music Settings

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

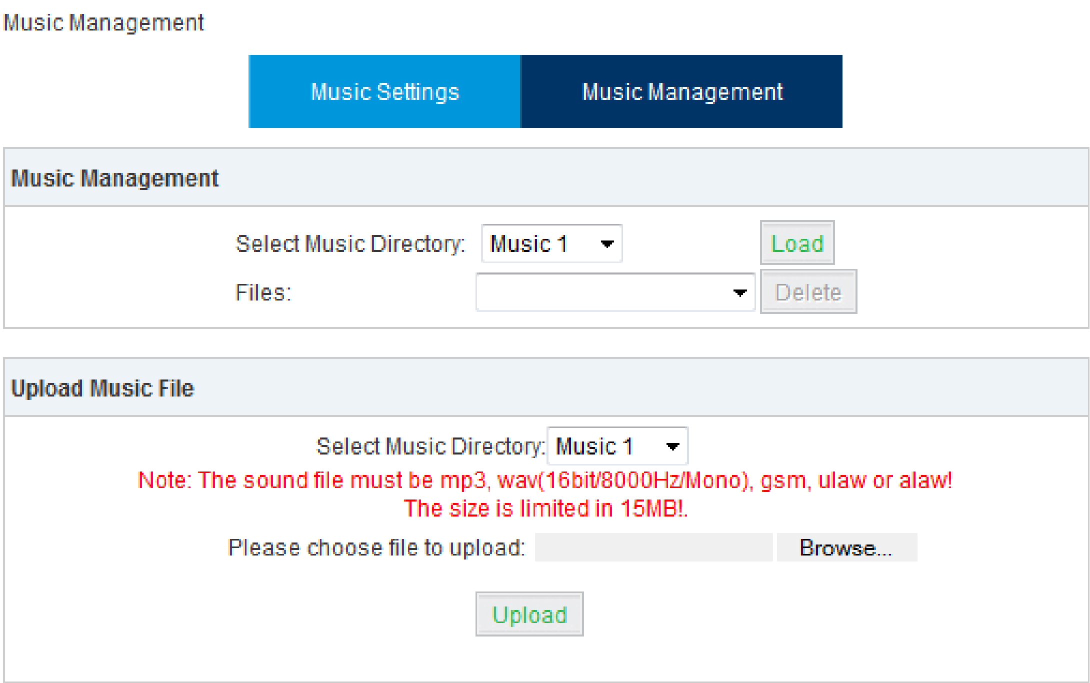
Click【Music Settings】 to display the dialog shown below:

**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. |

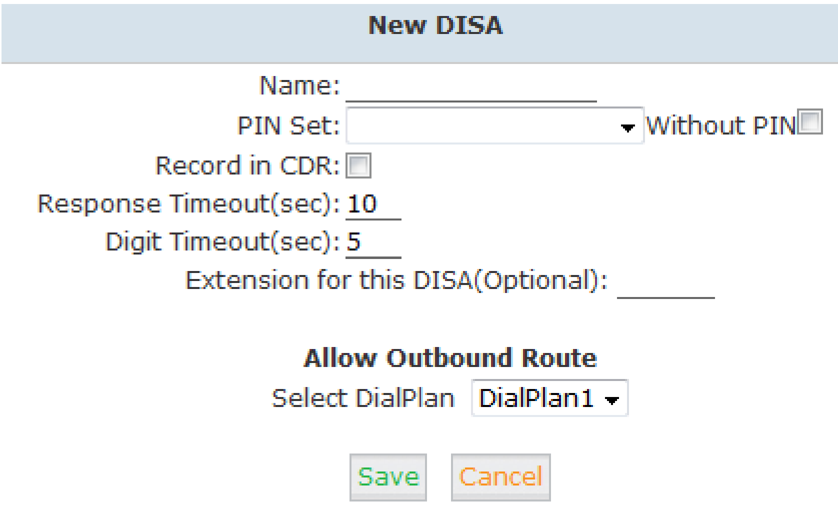
Note

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:



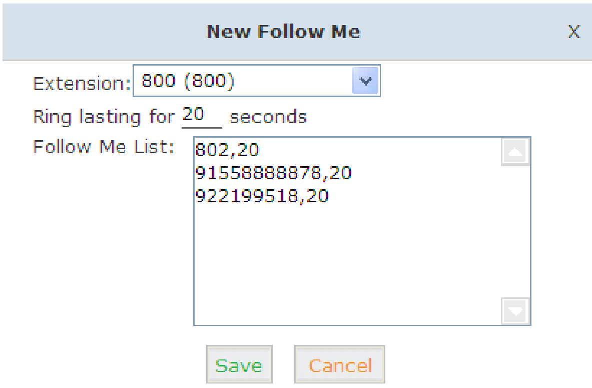

| Item | Explanation |
|---|---|
| Name | Define a name for DISA. |
| PIN Set | A set of PIN codes to authorize the callers using the system features and facilities. |
| Without PIN | If enabled, the callers will not be required to enter any PIN code to be able to use the system features can facilities (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.

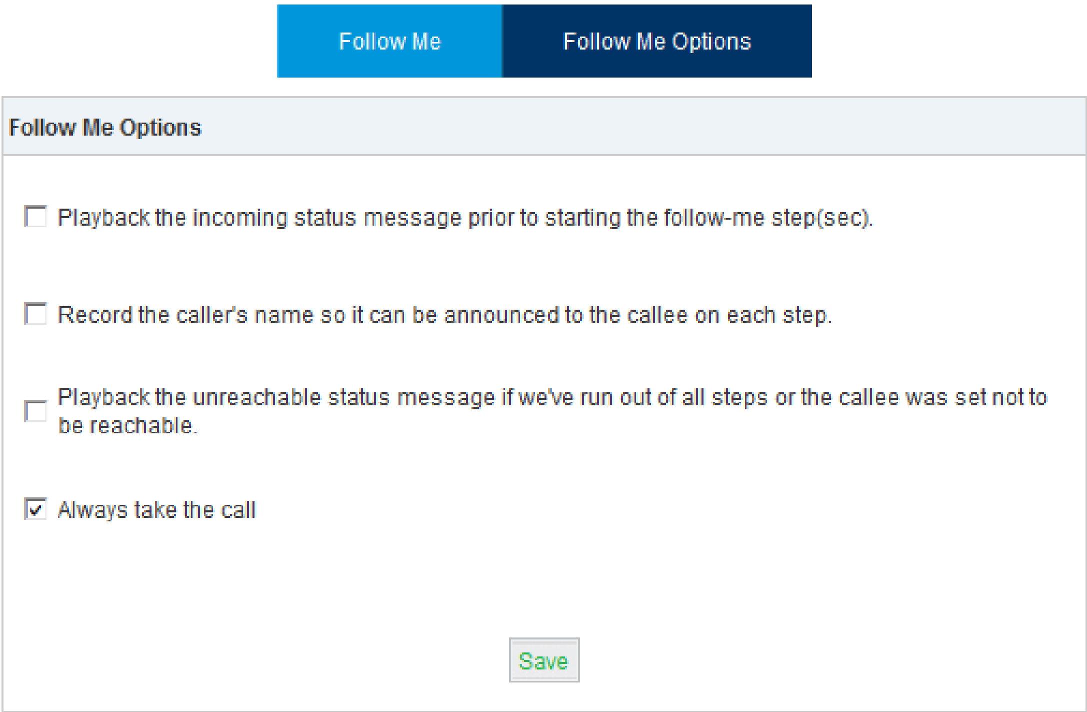
Navigate to web menu *Advanced->Follow Me*. Click on "New Follow Me" to configure follow me for an extension.




| 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. |

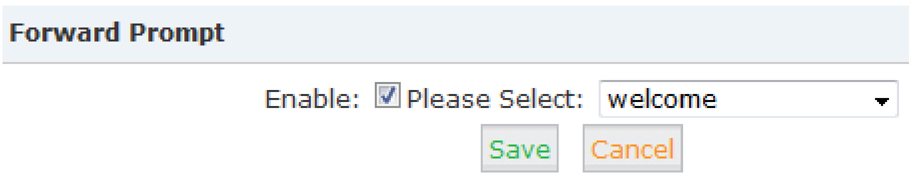
Follow Me Options



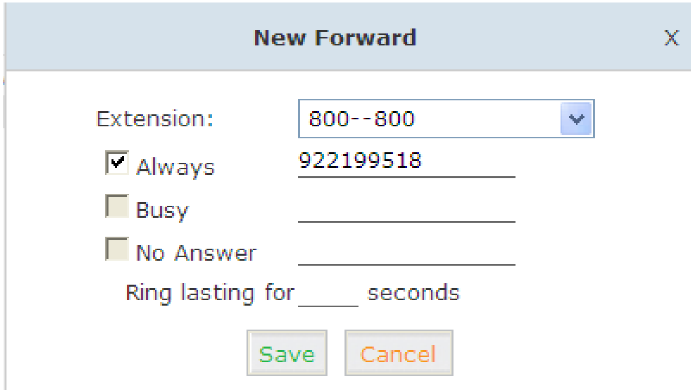

## 3.5.9 Call Forward

### 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.



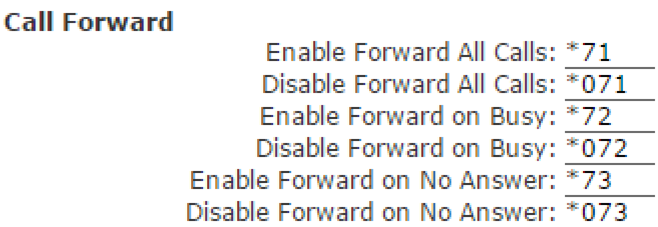

| Item | Explanation |
|---|---|
| Always | Unconditionally forward the incoming calls. |
| Busy | Forward the incoming calls only if the extension is busy. |
| No Answer | Forward the incoming call only if the extension didn’t answer. |
| Ring lasting for ____ seconds | Only is call forward on “No Answer” 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.<br>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.

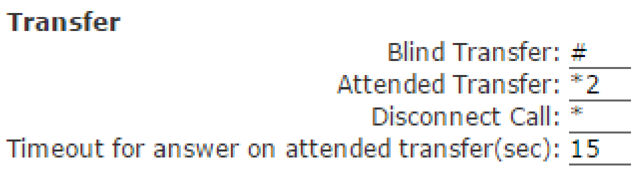
Deactivate call forward no answer: Dial *073.

### 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:

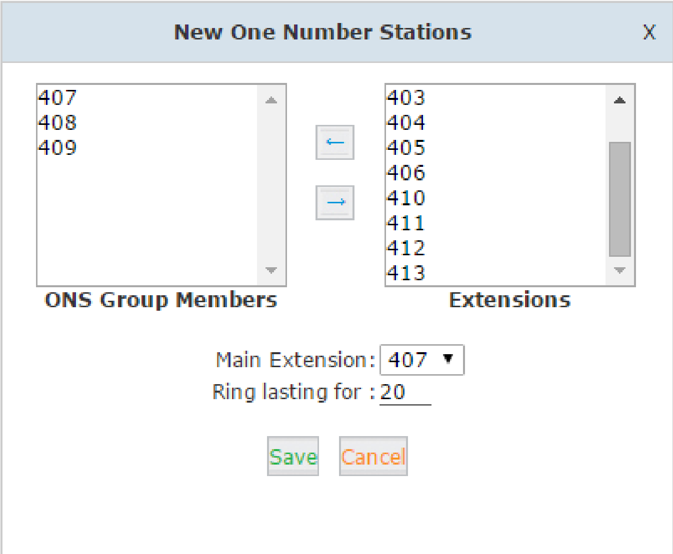



| Item | Explanation |
|---|---|
| Blind Transfer | In a live call, extension user can press # key and the IP PBX system prompts "Transfer", then you enter the number to be transferred to. This call will be transferred instantly and the user can hang up. If the transferred number didn't answer this call 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.

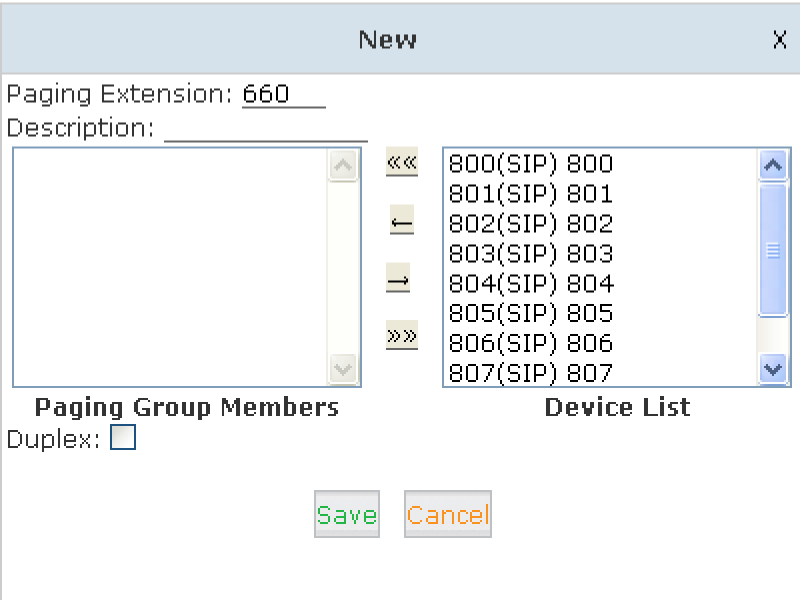
In this case, no matter what extension -- 407, 408 or 409, if they call other extensions, others only see it is extension 407 calling. Others call 407, all these 3 extensions will ring.
As you can see on this page there's a feature code Switch Station available.


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.<br>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.

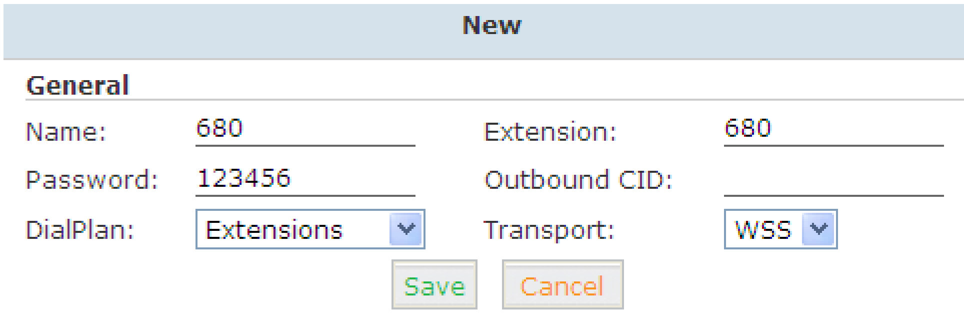
Click on the “New User” button to add a new web extension.

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:**

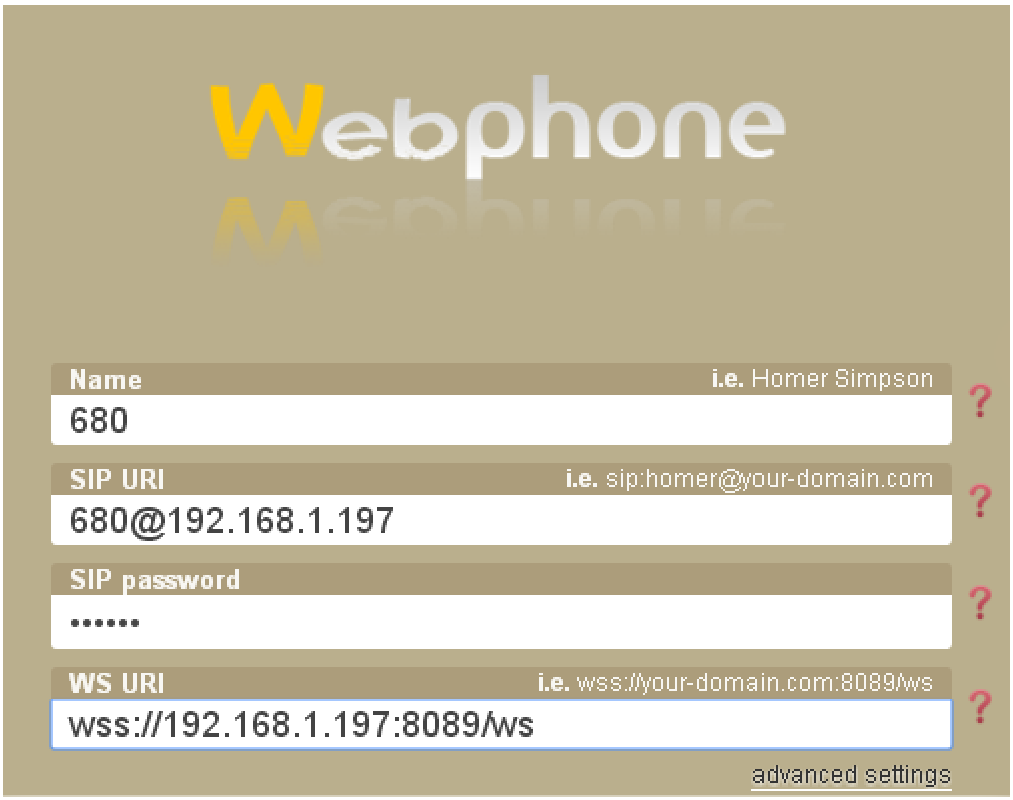
**Upgrade Web extension patch**

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:

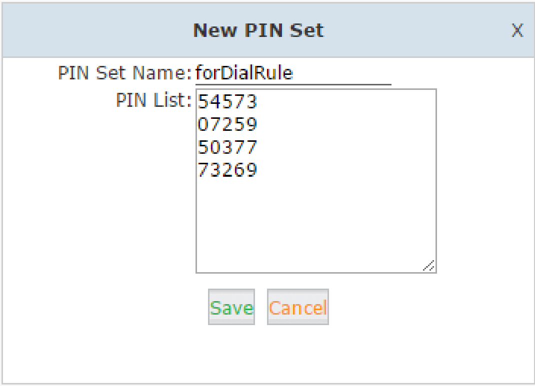



Next, press Enter and the web extension will be registered and is ready for phone calls just like any other standard extension. WebRTC can even be adapted to the enterprise website which can help an enterprise serve their customers with direct voice communication via their website.

## 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.

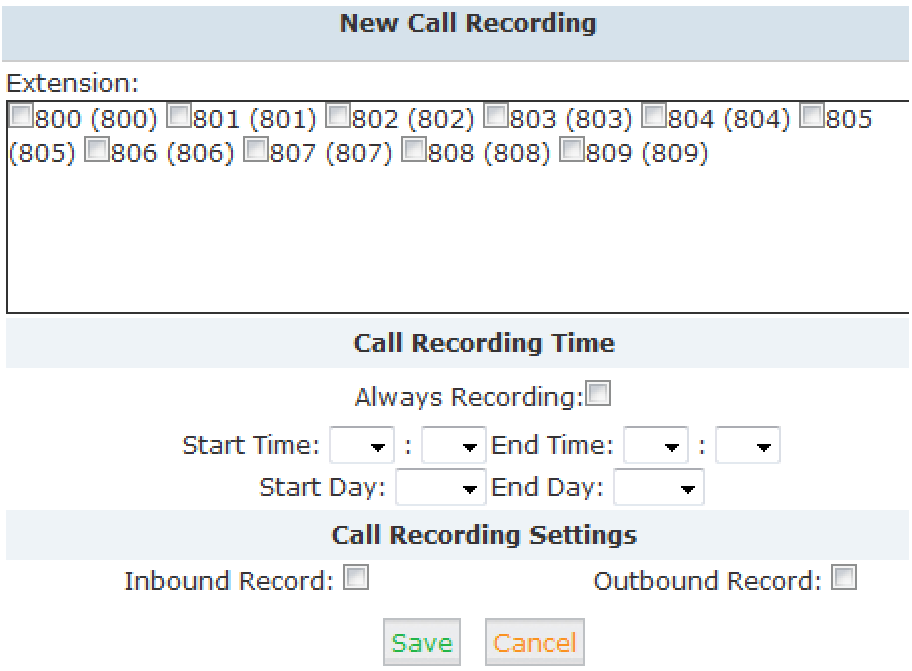



Each line is a PIN code. Press Enter to write down the next PIN code without any symbols.

### 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:

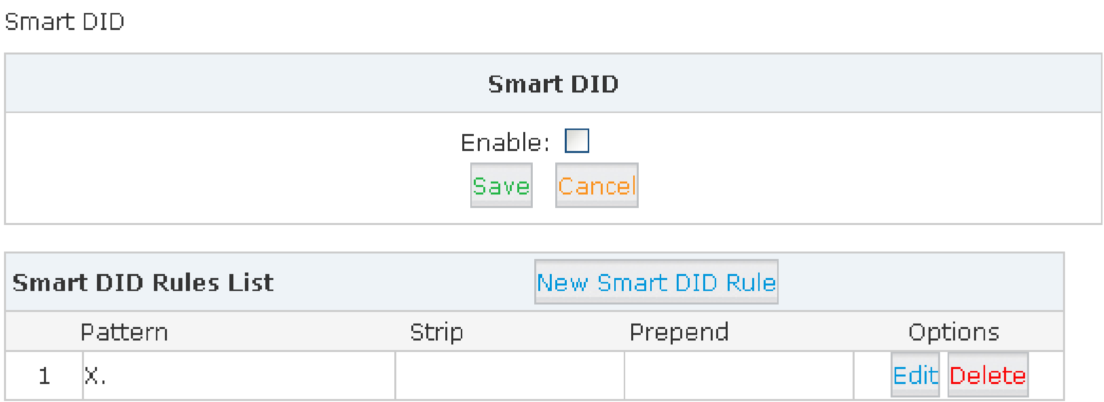
| Item | Explanation |
|---|---|
| Extension | Select the extensions which you want all their calls to be recorded. |
| Always Recording | If enabled all calls of the above selected extension will be recorded no matter when the calls have been made and received. |
| Start Time, End Time, Start Day, End 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

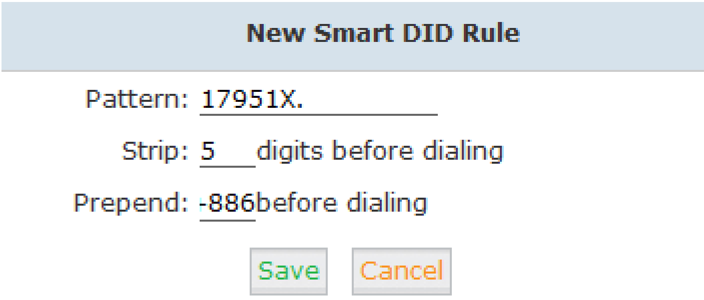
The IPPBX system has the ability to route an inbound call directly to an extension if previously the extension called this number without answering. It is convenient for the called party to make a call back and finds the extension user directly without going through the IVR menu or any other improper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

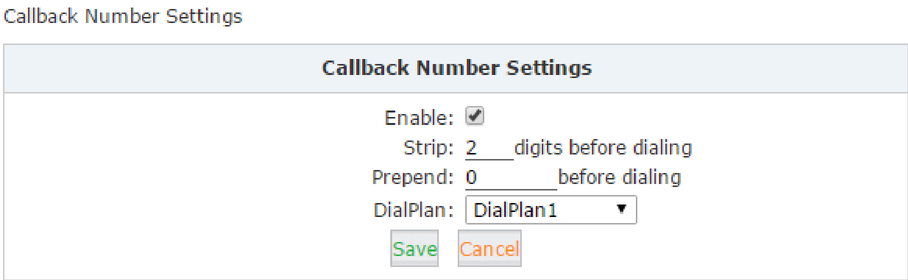
Note

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

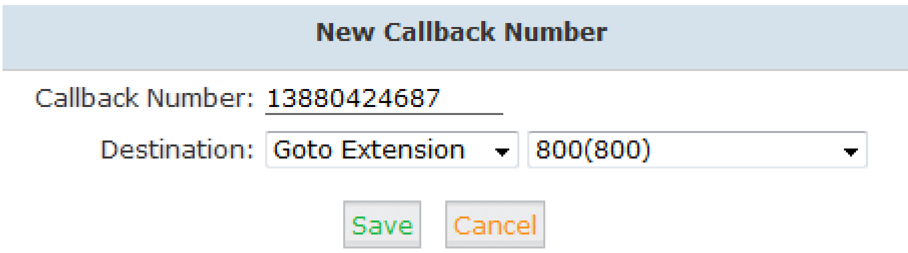
Call Back is a basic service on an IPPBX system for saving on international calls and reducing company phone costs. Ideal for SMB and Corporate business, this PBX feature is designed for users who are making calls from any international destination back to their home country. Please configure it as shown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

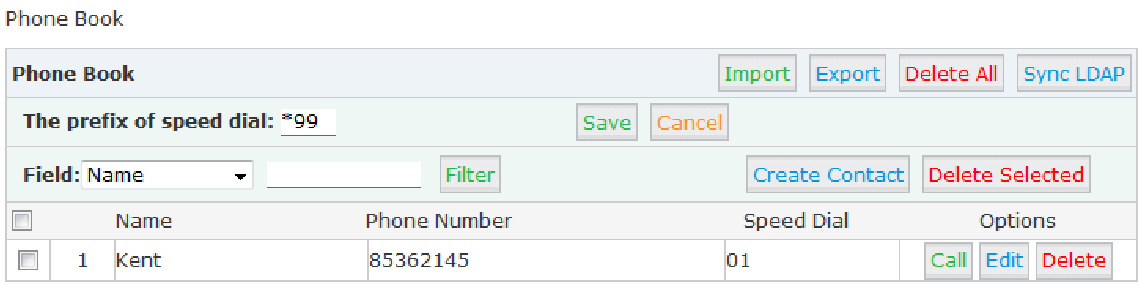
| Item | Explanation |
|---|---|
| Callback Number | The number which calling in to the IPPBX system will be handled by Callback. |
| Destination | An extension or another call destination which will be used to call the callback number. |

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

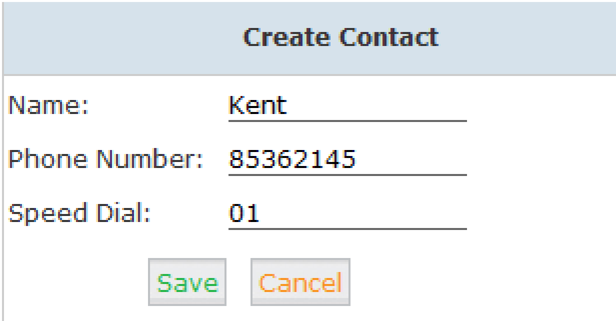
When incoming call matches the number in the phone book, the name of the matched number will be displayed. Please configure it as shown below:

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:

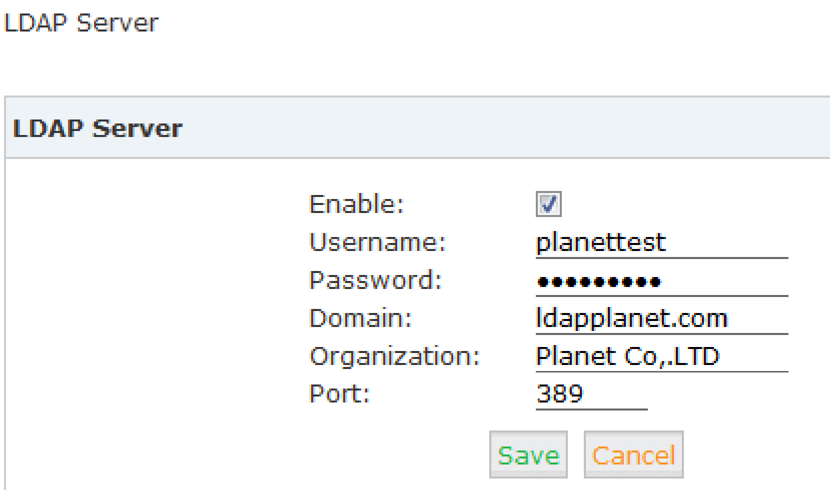



| Item | Explanation |
| --- | --- |
| Name | Input contact's name. (Letter or figure only). |
| Phone Number | Input Phone Number of contact. |
| Speed Dial | Speed dial number which can be used to call this contact from the extensions.<br>After the contacts have been created they will be listed here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

## 3.5.18 LDAP Server

### 3.5.18.1 LDAP Server Settings

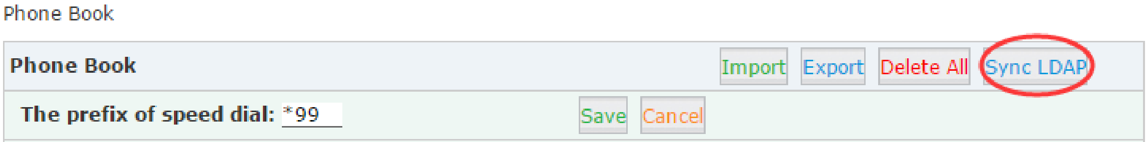
LDAP (Lightweight Directory Access Protocol) is an open, vendor-neutral, industry standard application protocol for accessing and maintaining distributed directory information services over an IP network. LDAP server has been embedded to IP PBX which is mainly used to centralize manage the phonebook.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



| 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. |

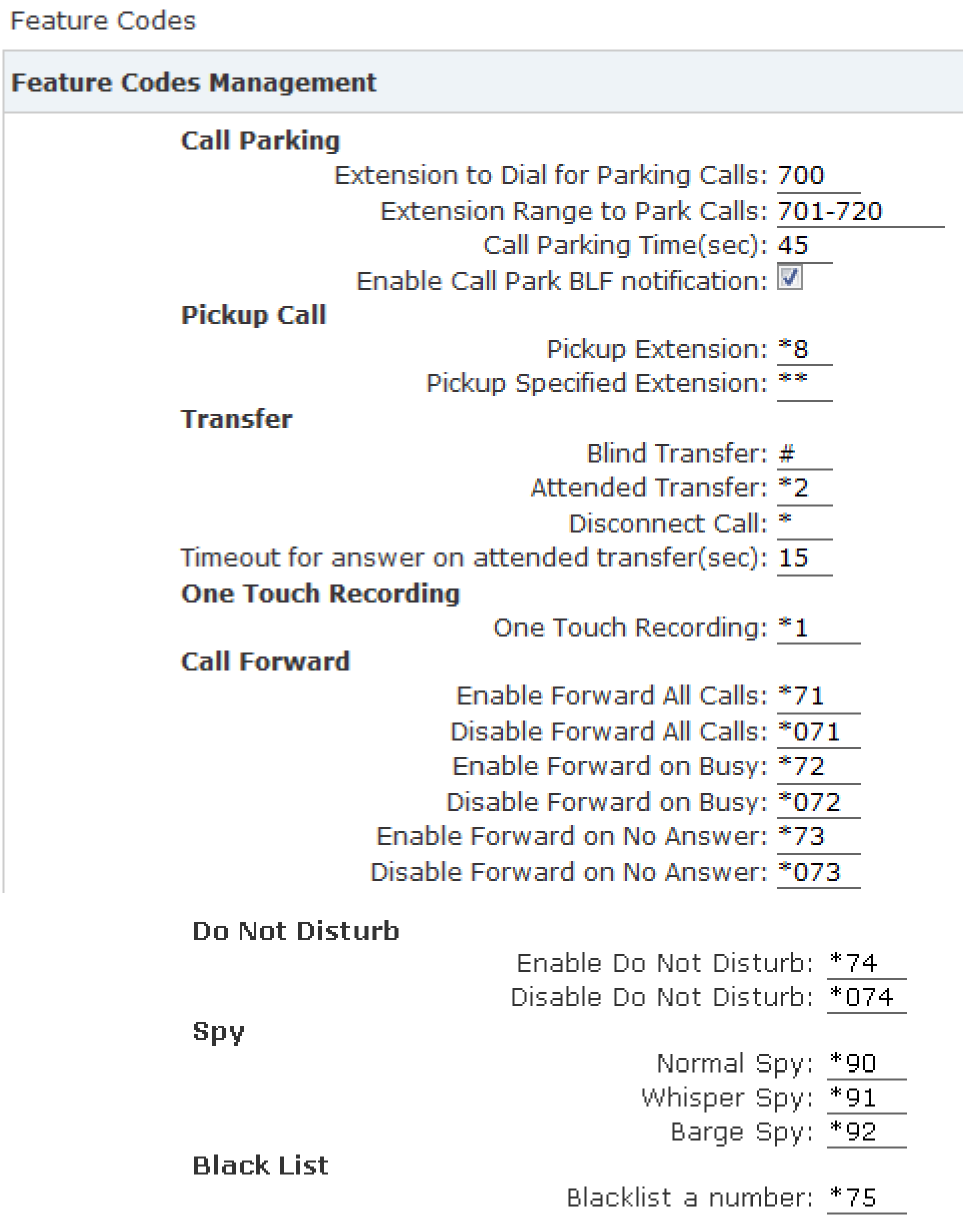
### 3.5.18.2 Synchronize Contacts with LDAP Server

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 |

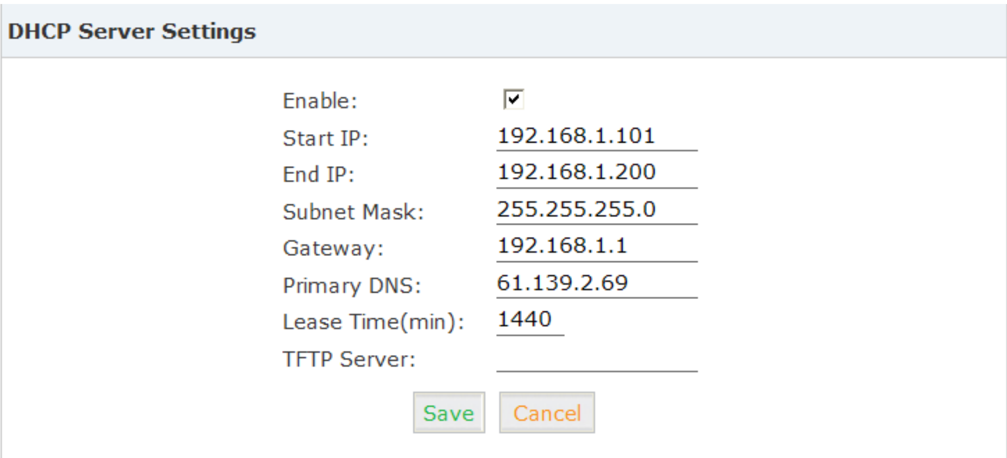
## 3.5.20 Phone Provision

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.

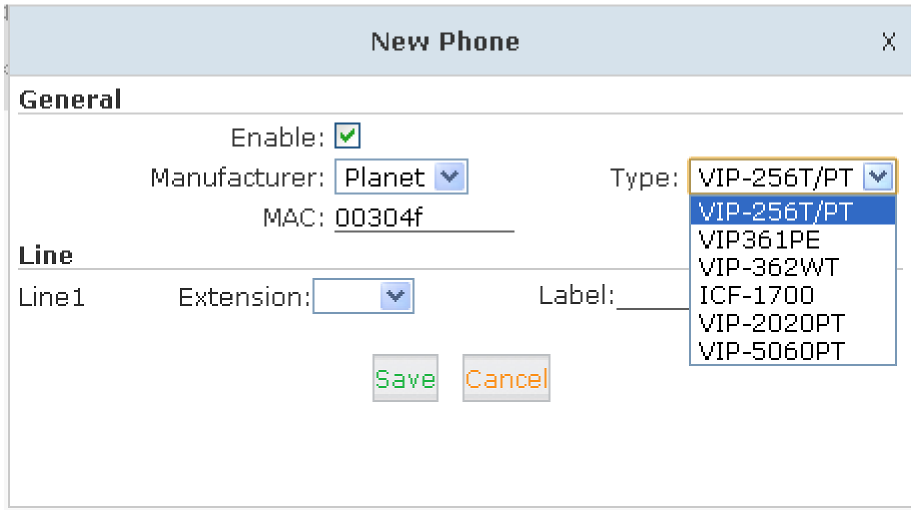
There are two operation methods to fulfill this function. Please see details shown below:

**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.

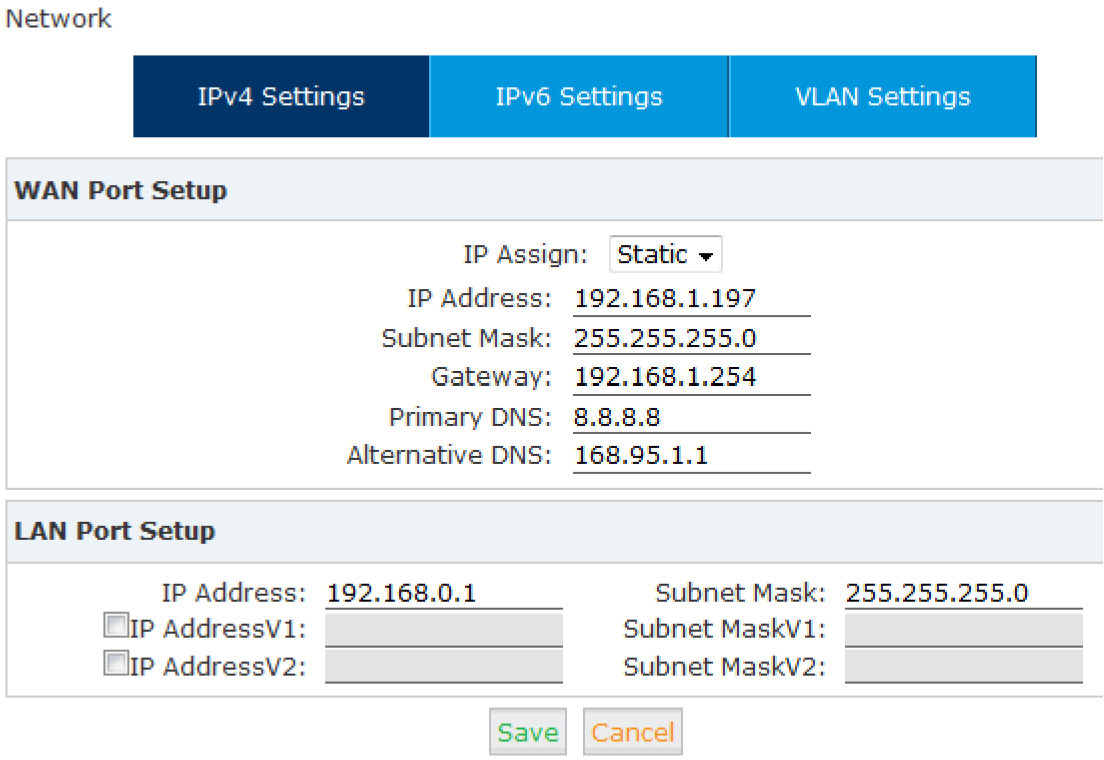
# Chapter 4. Network Settings

## 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

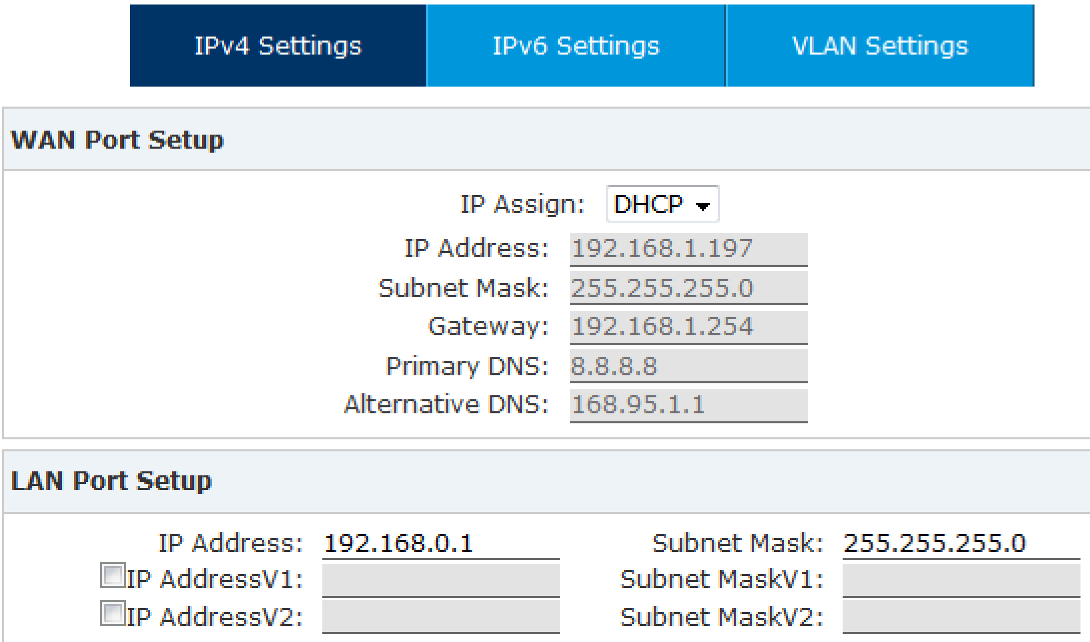
| Item | Explanation |
|---|---|
| IP Assign | Static/ DHCP/PPOE supported. |
| LAN Interface | Define the LAN interface. |

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

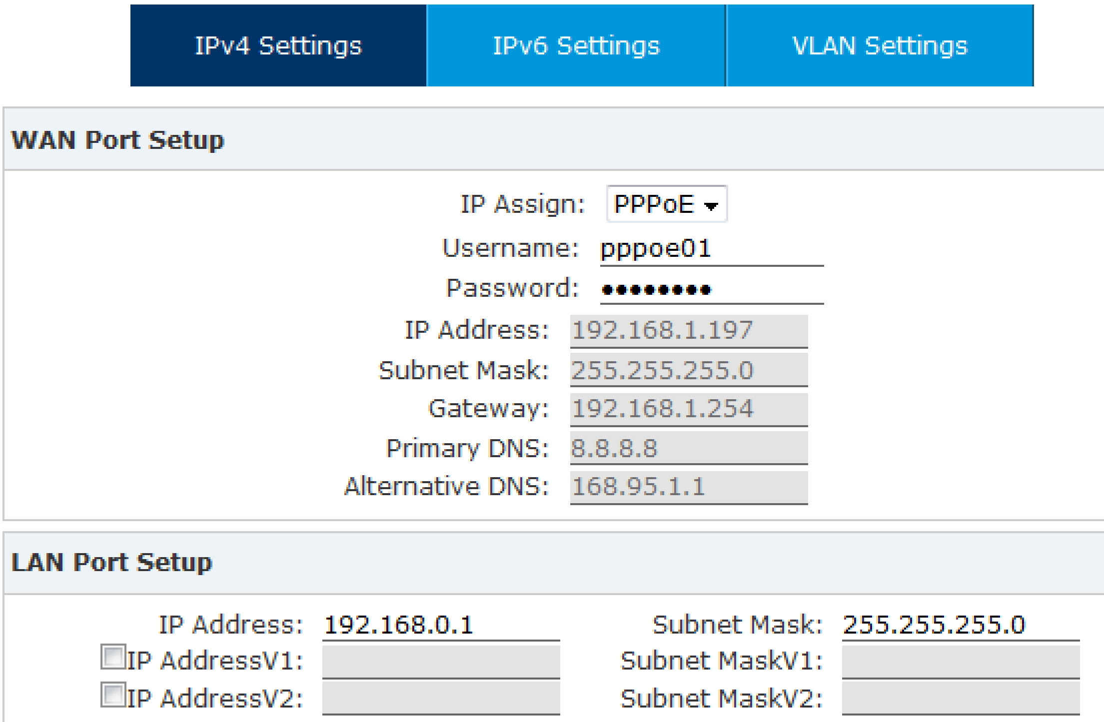
If your Internet connection automatically provides you with a usable IP address, you can select “DHCP” on WAN interface.




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.

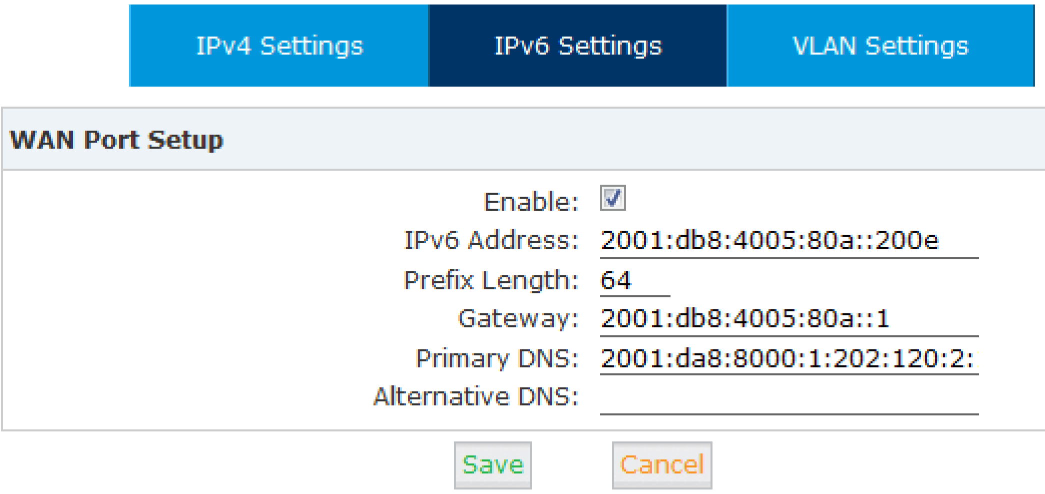
## 4.1.2 IPv6 Settings

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】



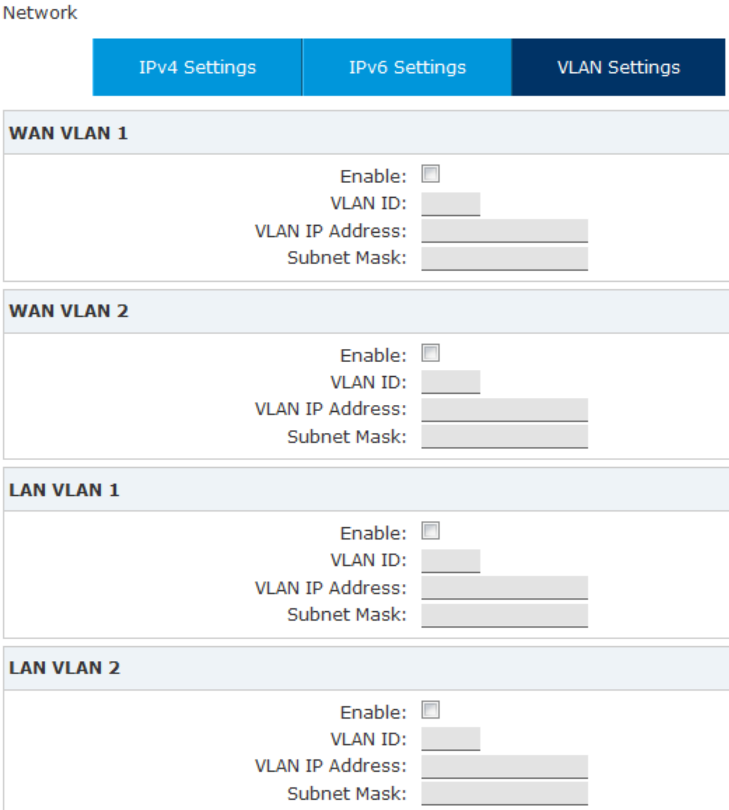

IPv6 Reference:

| 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:

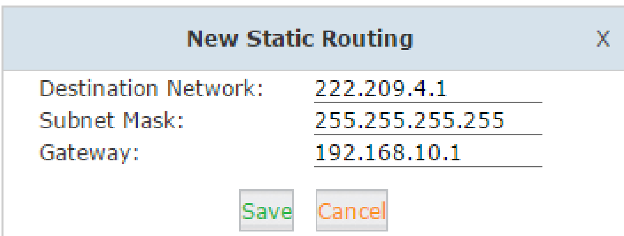
| Item | Explanation |
|---|---|
| Enable | Enable VLAN to define the VLAN address and VLAN ID. |

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

# 4.2 Static Routing

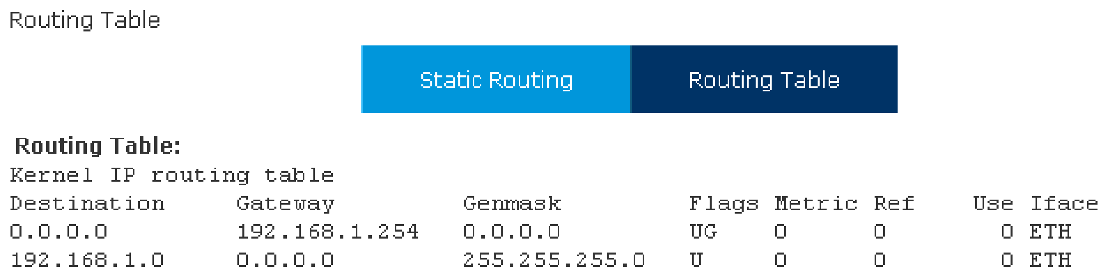
Static Routing is a form of routing that occurs when a router uses a manually-configured routing entry, rather than information from a dynamic routing protocol to forward traffic.

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



| Item | Explanation |
|---|---|
| Destination | Set the IP address of destination host or network address.<br>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. |

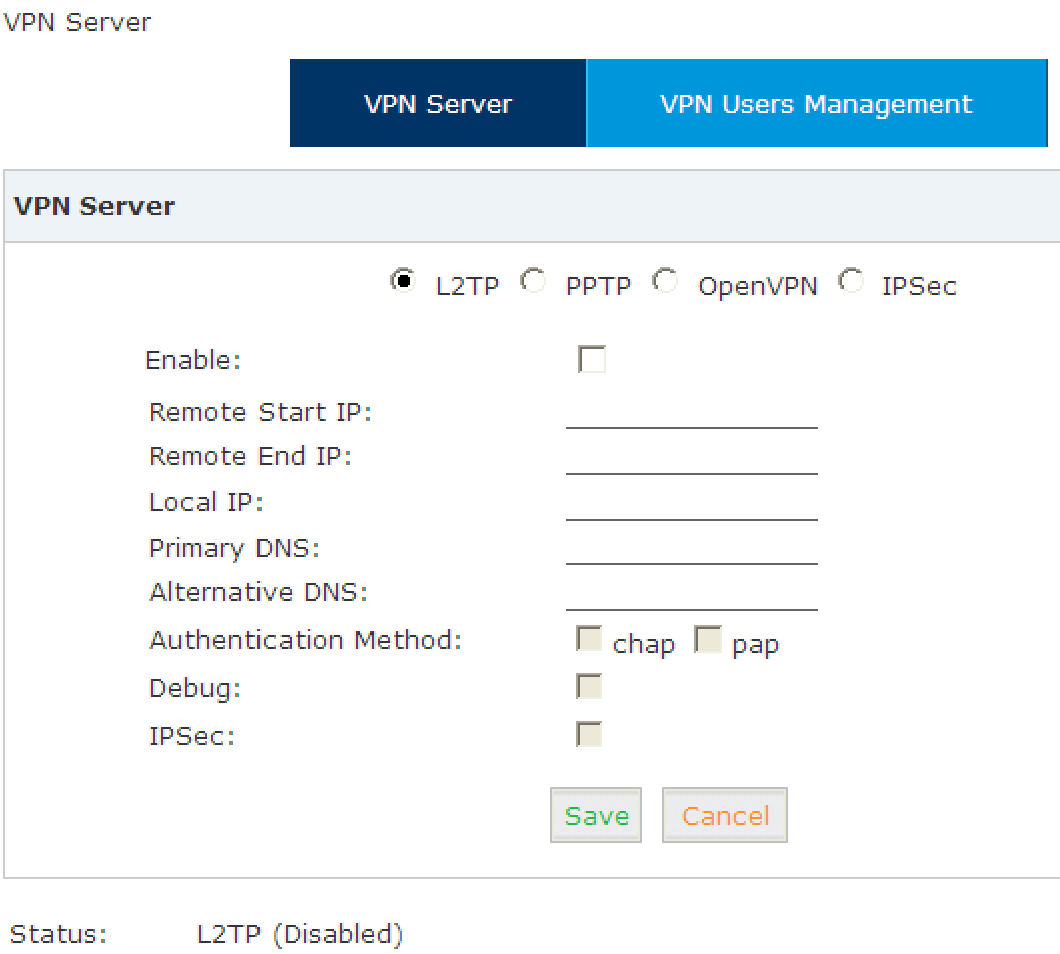
Click【Network Settings】->【Static Routing】->【Routing Table】, and the current routing information will be displayed below:




## 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

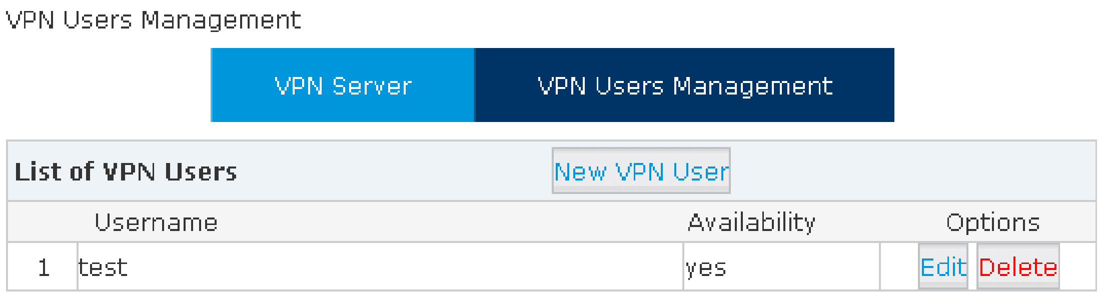

Reference:

| 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.<br>pap: Password Authenticate Protocol PAP works like a standard login procedure; it uses static user name and password to authenticate the remote system.<br>chap: Challenge Handshake Authentication Protocol<br>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】:



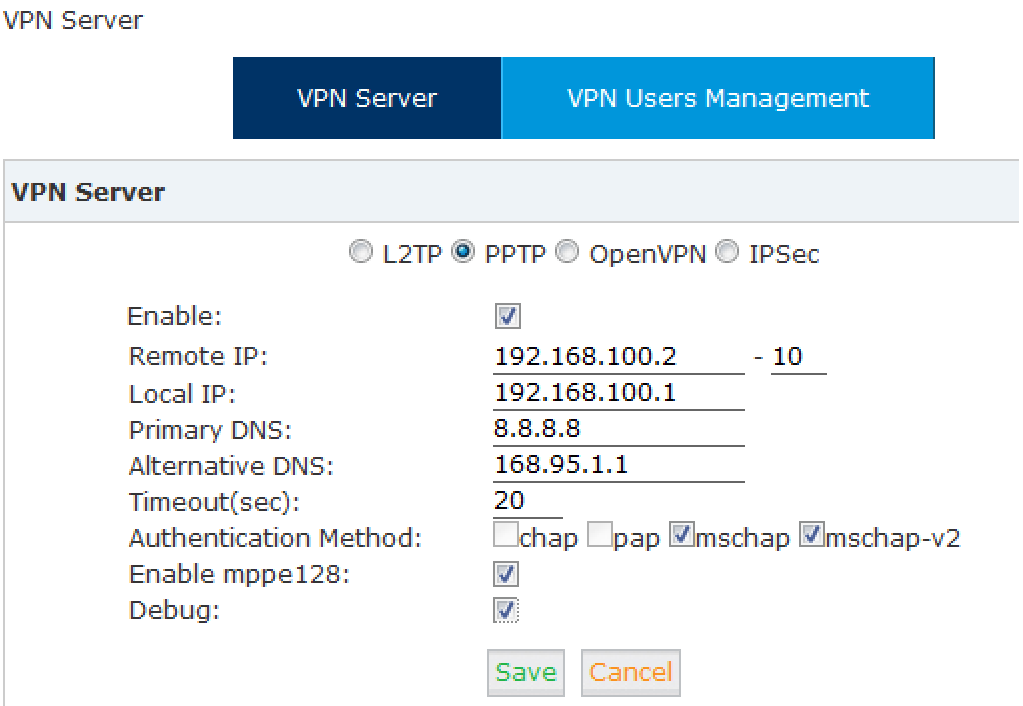

This page is used for management of VPN user name and password.

## 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.<br>● **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.<br>● **pap**: Password Authenticate Protocol PAP works like a standard login procedure; it uses static user name and password to authenticate the remote system.<br>● **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.

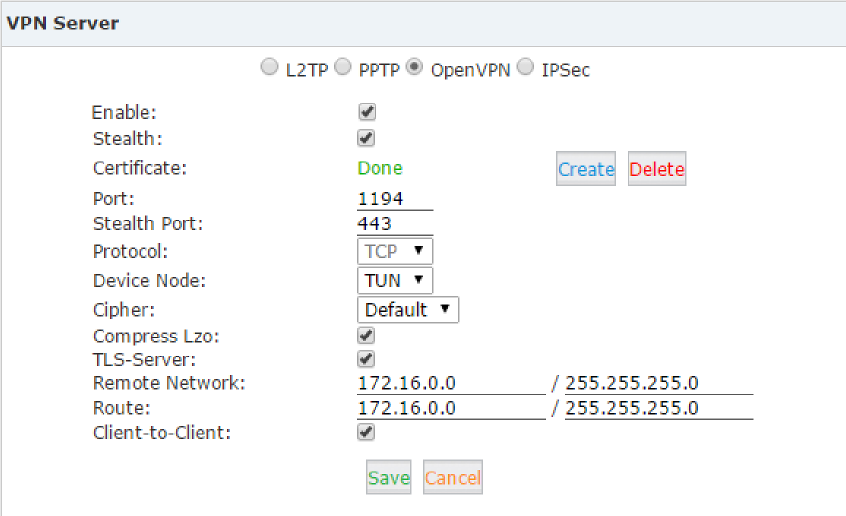
| Note | If the IPPBX system is behind NAT, you need to open ports 1723 on the router/firewall. |
|---|---|

## 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. |

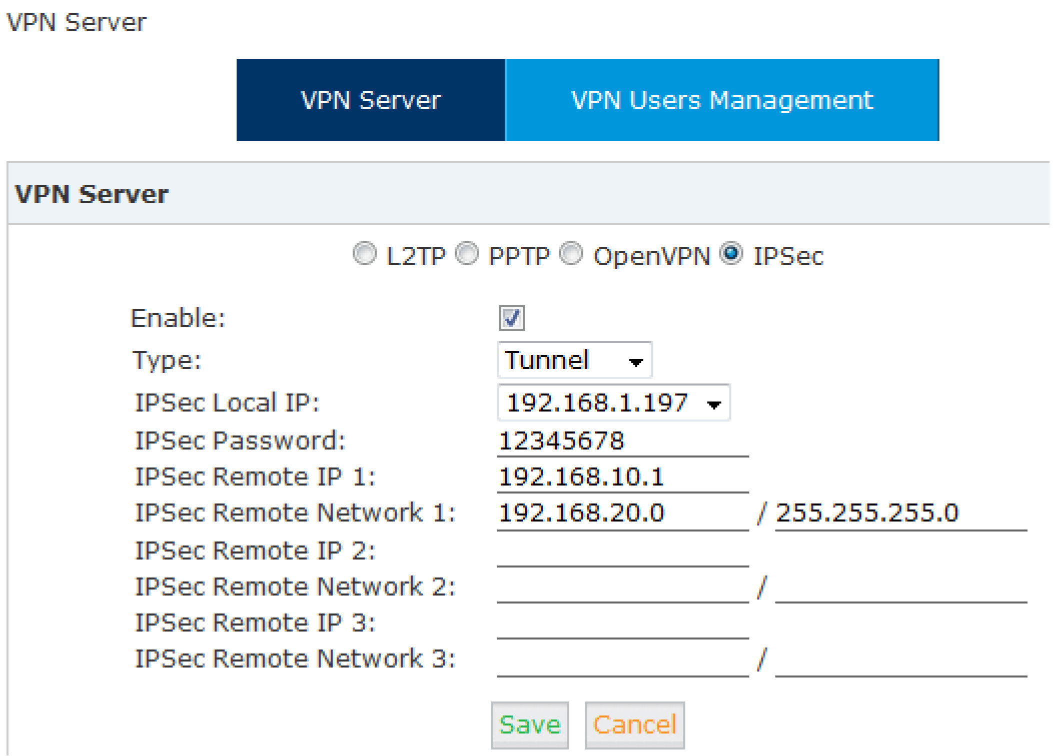
## 4.3.4 IPSec VPN

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. |

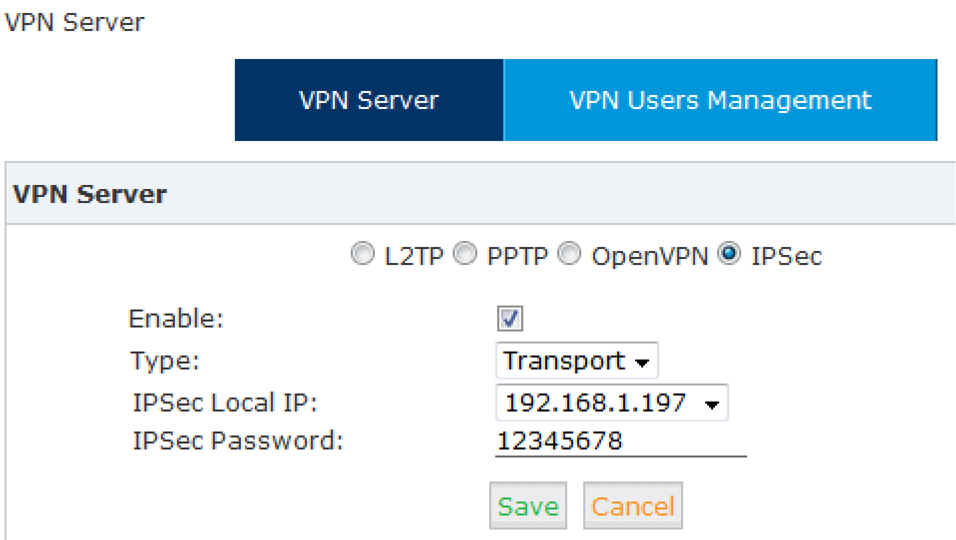
**Note**

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.



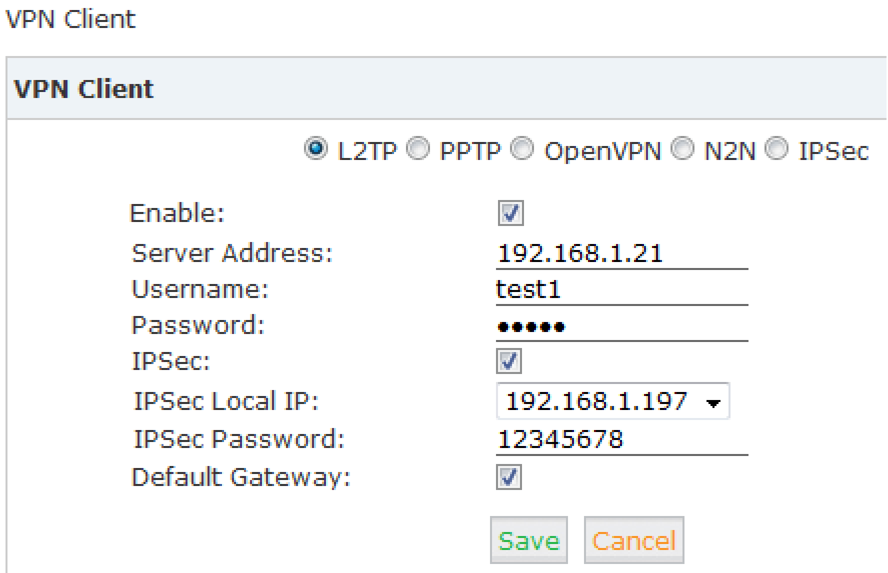

| Item | Explanation |
|---|---|
| Enable | Tick the checkbox to enable IPSec VPN server. |
| Type | Select Transport mode. |
| IPSec Local IP | IPPBX WAN IP.(Same as configuring Tunnel mode) |
| IPSec Password | Define a password for authentication of the IPSec client. |

# 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

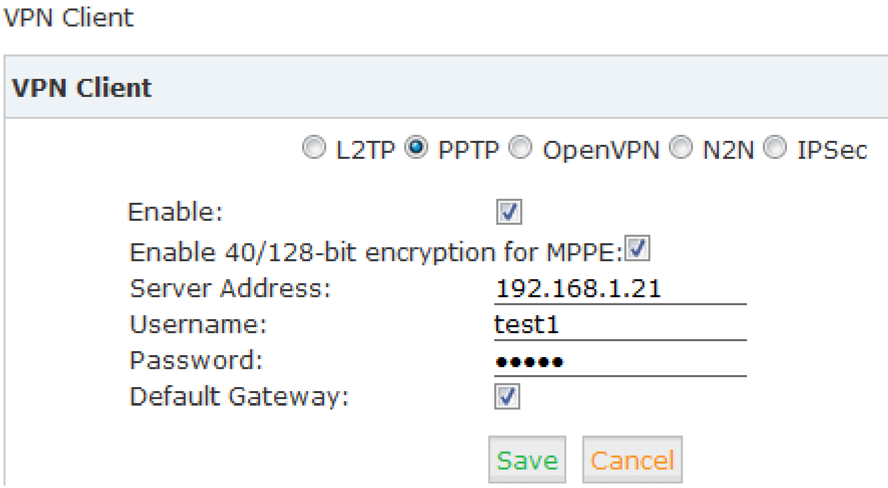



Reference:

| 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.

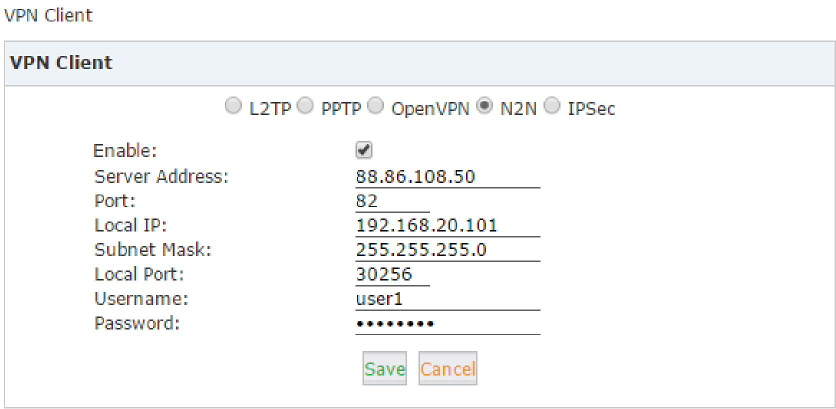



| Item | Explanation |
|---|---|
| Enable | Tick to enable PPTP VPN client. |
| Enable 40/148-bit encryption for MPPE | Tick to enable 40-bit key (standard) or 128-bit key (strong) MPPE encryption schemes. |
| Server Address | PPTP VPN server public IP. |
| Username | PPTP VPN user name given by the 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.

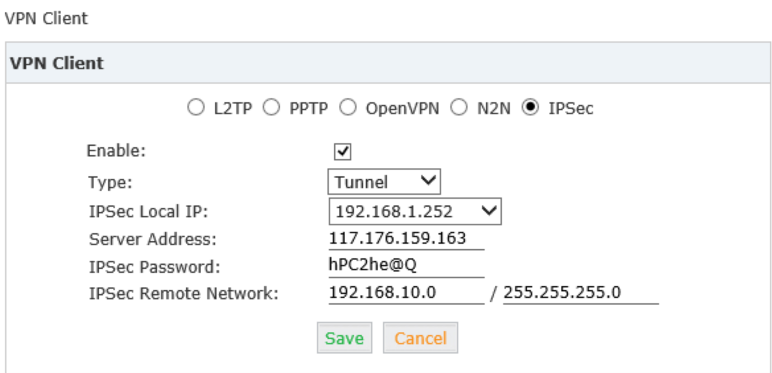



| Item | Explanation |
|---|---|
| Enable | Tick this checkbox to enable N2N VPN client |
| Server Address | N2N server(supernode) IP address. |
| Port | N2N service port number. 82 by default. |
| Local IP | VPN local IP. |
| Subnet Mask | Netmask of the VPN network. |
| Local Port | N2N local service 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. |

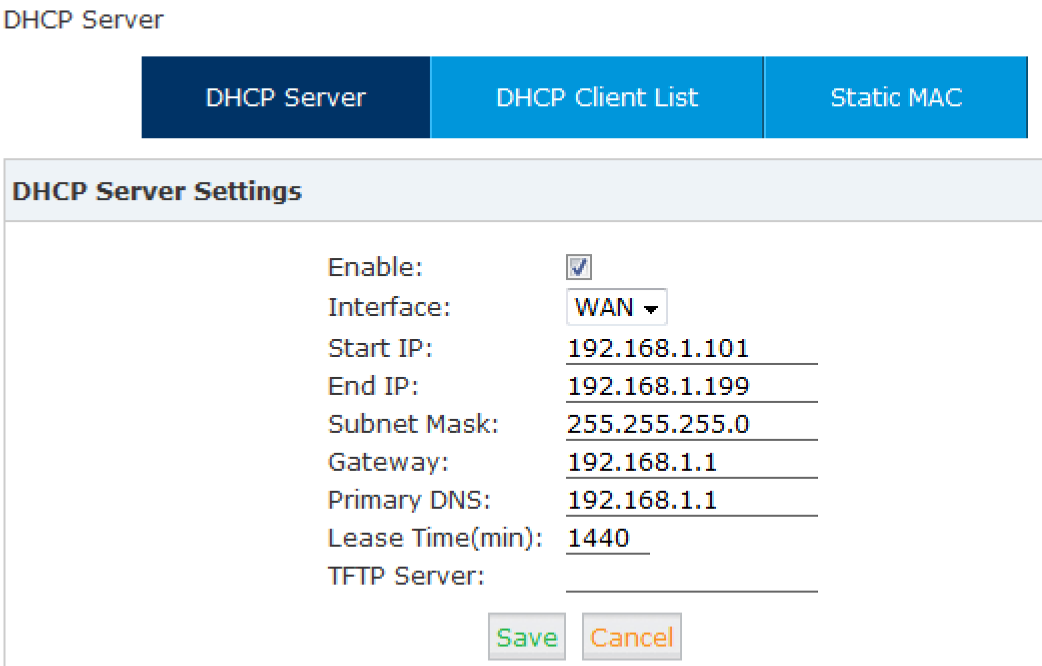
# 4.5 DHCP server

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



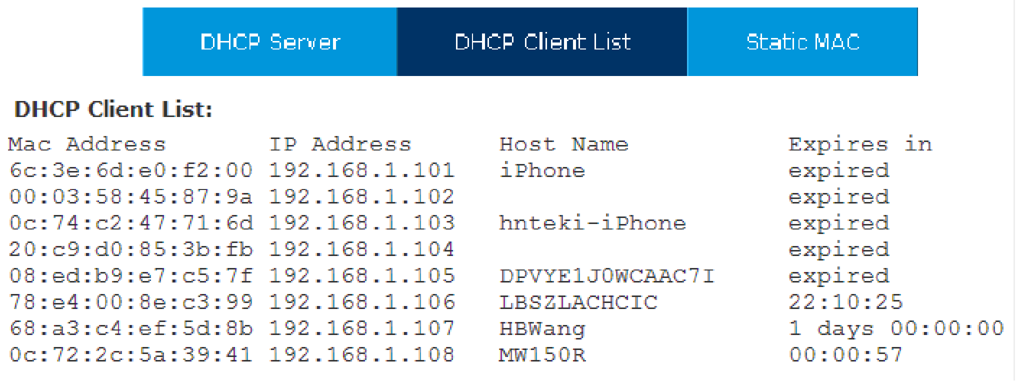

| Item | Explanation |
|---|---|
| Enable | Enable DHCP service. |

| 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 |

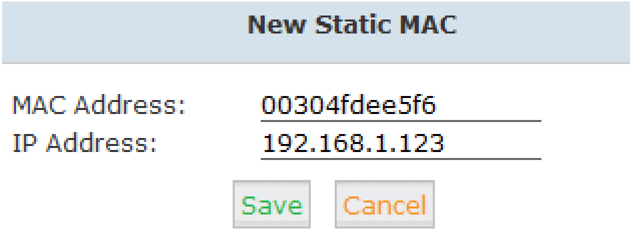
This page is used to display DHCP Client address and related information.

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.

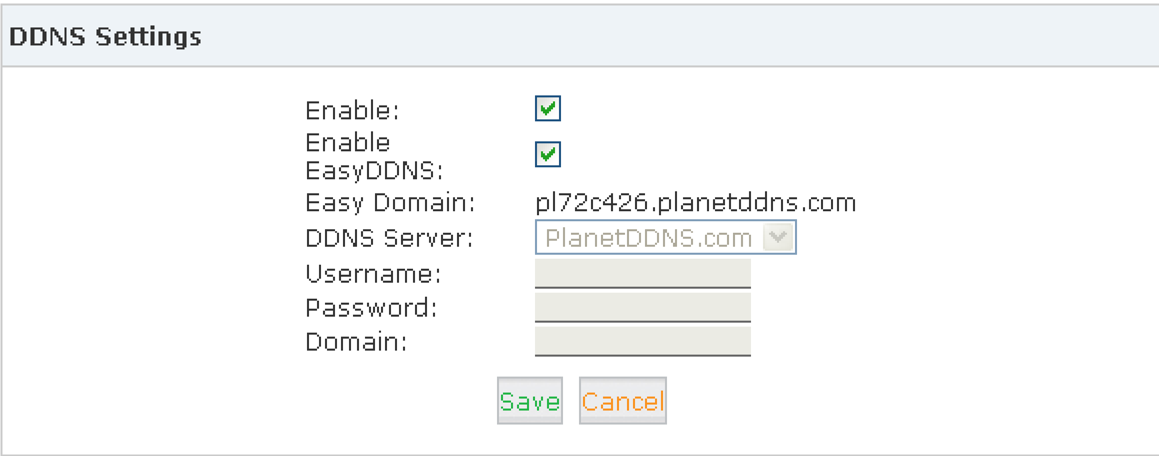



## 4.6 DDNS Settings

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.

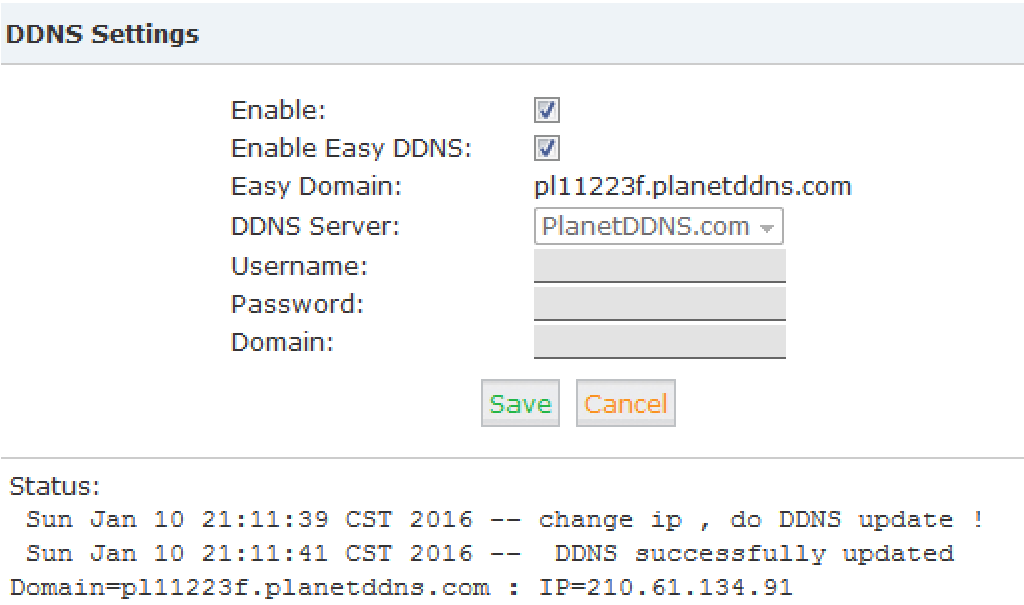
After setting DDNS, IP PBX phone services can be accessed from remote site via the domain name which DDNS provider gave. Also remote management is possible even without a static public IP.

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

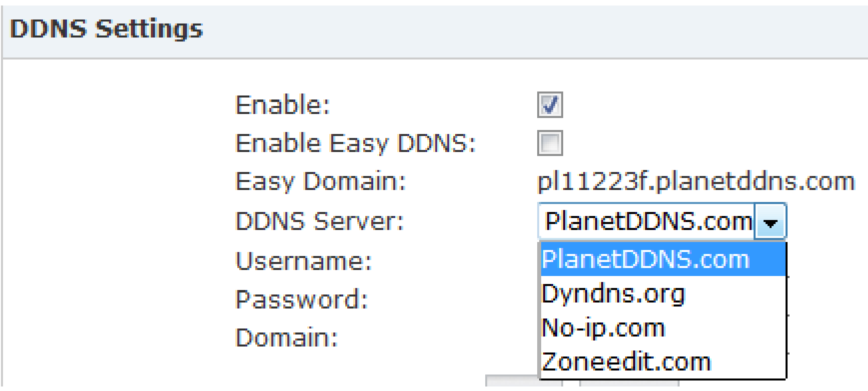



| Item | Explanation |
|---|---|
| Enable | Tick to enable DDNS service |
| DDNS Server | Select the DDNS service provider which you subscribed the DDNS service. |
| Username | Username you subscribed to the service provider. |
| Password | Password you used to sign up to the service provider. |

| Item | Explanation |
|---|---|
| Domain | Your domain name. |



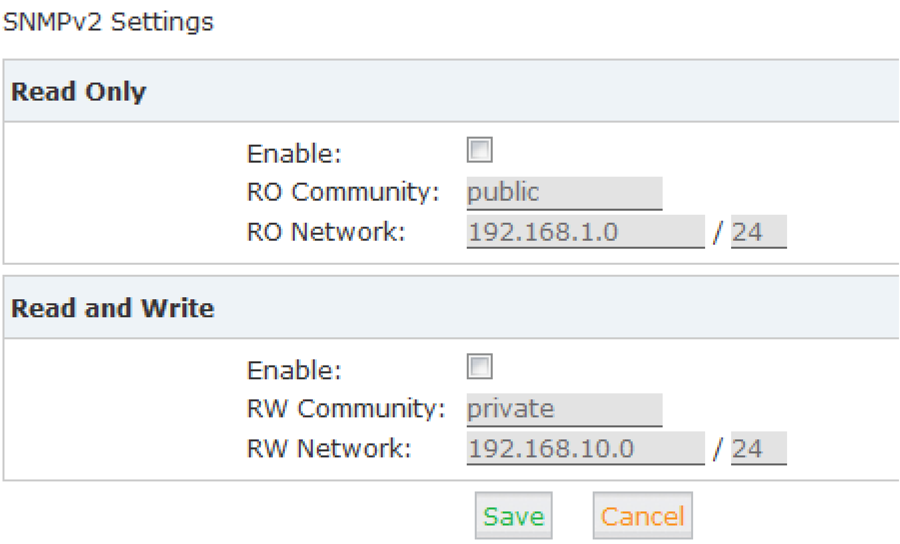

IP PBX supports DDNS provided by Planet DDNS, Dyndns.org, No-ip.com and zoneedit.com.



## 4.7 SNMPv2 Settings

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

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

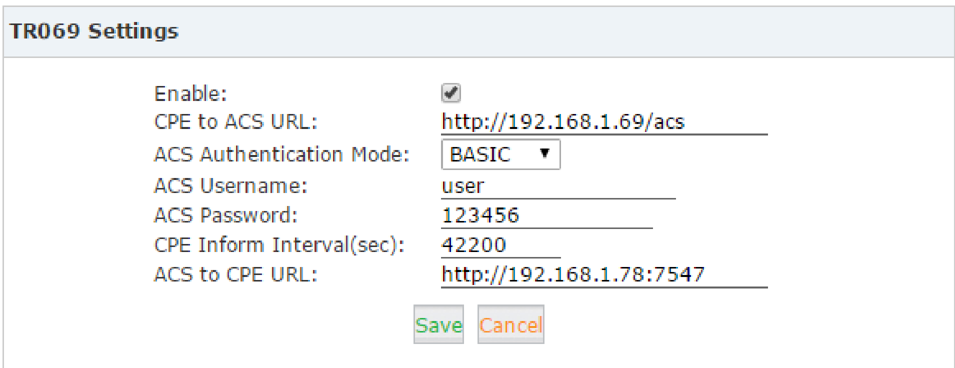



Reference

| 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). |

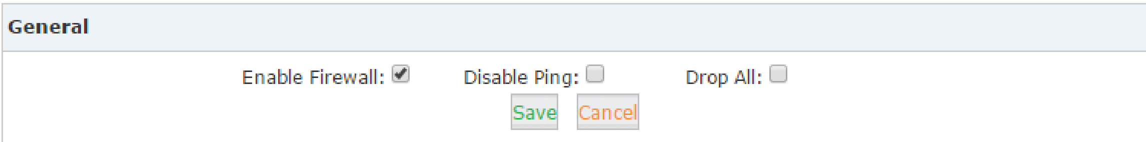
# Chapter 5. Security

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

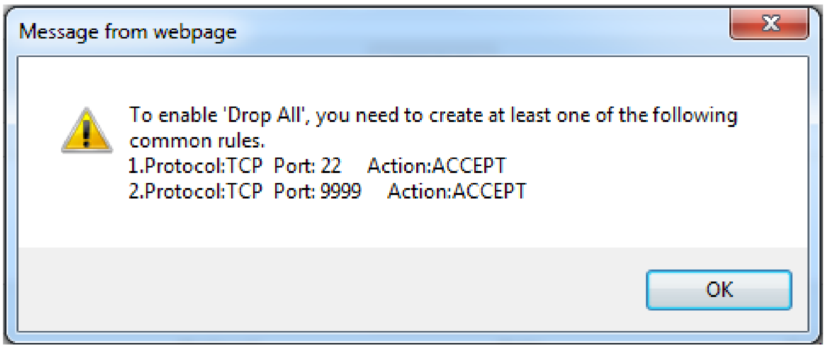
## 5.1 Firewall

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. |

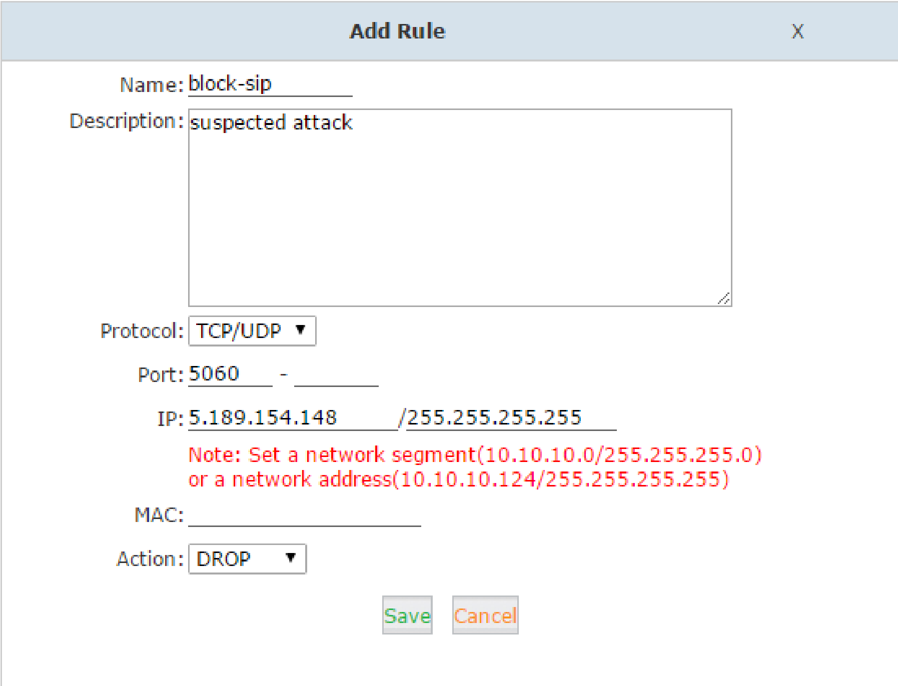



The rule/rules can be created in the “Common Rules” section.

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. |

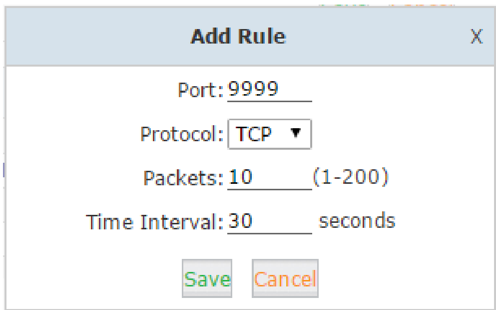
**Auto Defense**

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.

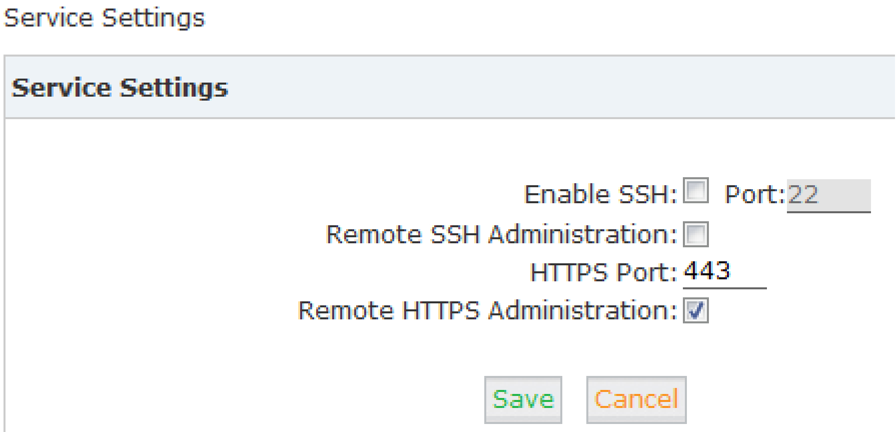
| Rejected IP | | |
|---|---|---|
| Type | IP | Options |
| VOIP | 212.83.154.178 | Delete |
| VOIP | 173.249.158.227 | Delete |
| VOIP | 5.189.154.148 | Delete |

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. |

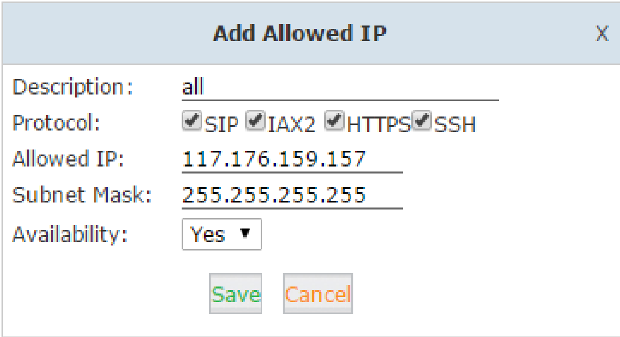
| Note | If you want remote access to SSH and web GUI of the IPPBX system, you can forward the corresponding ports on your router. Before doing this please make sure you have set strong password words for root user and web admin user. |
|---|---|

## 5.3 Fail2Ban

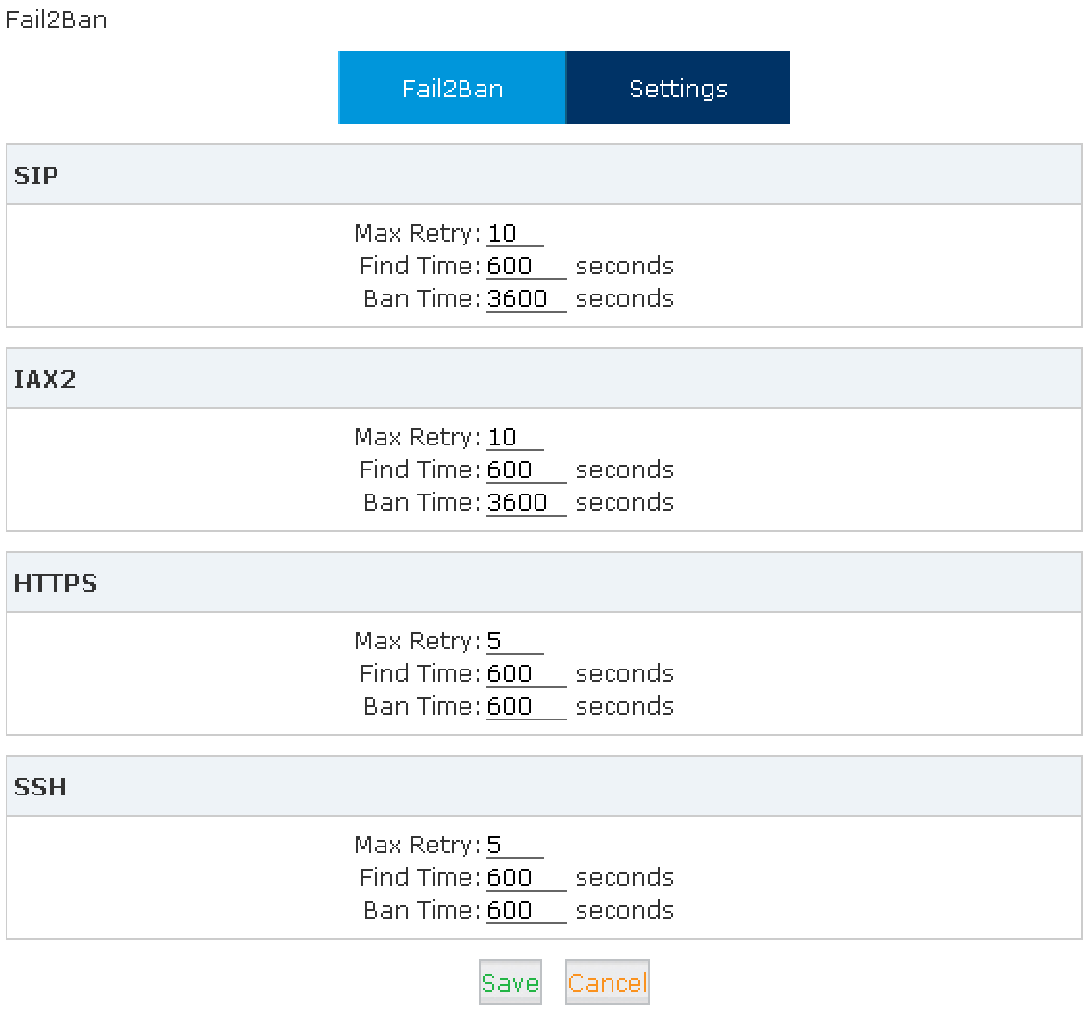
Planet IPPBX system uses Fail2Ban to perform intrusion detection; iptables is used for blocking any attack attempts. Fail2Ban is an intrusion prevention framework written in the Python programming language. It works by reading Asterisk logs and some other logs in the IPPBX system, and uses iptables profiles 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.

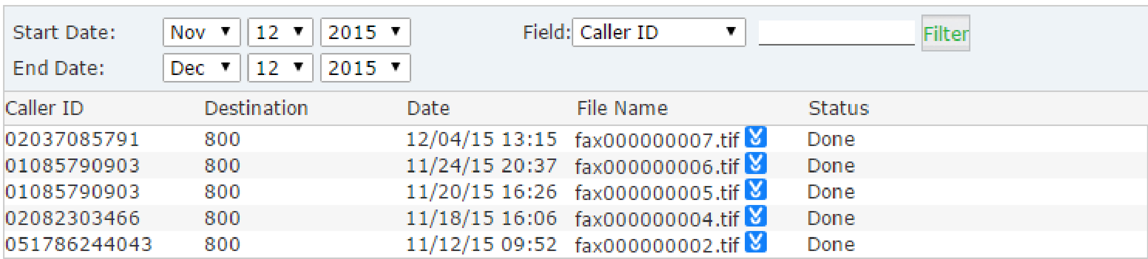
**Status and Description**

- 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.

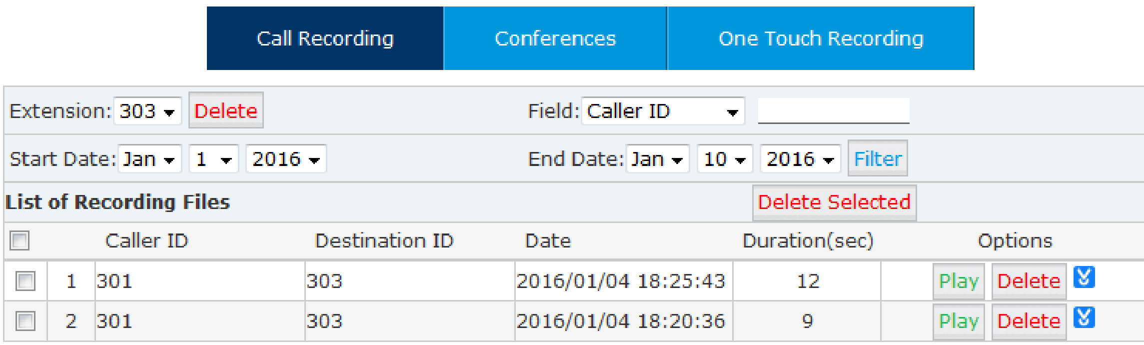
The faxes can be downloaded to your PC hard drive by clicking the button.

## 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

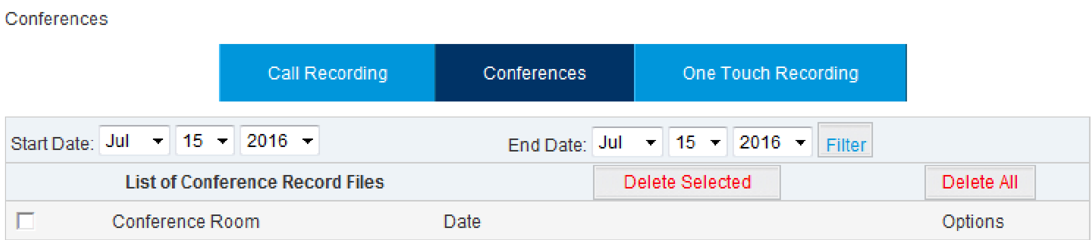



| Item | Explanation |
|---|---|
| Extension | Select an extension number to search the recordings of this extension. |
| Delete | Delete all recordings of the selected extension number. |

| 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

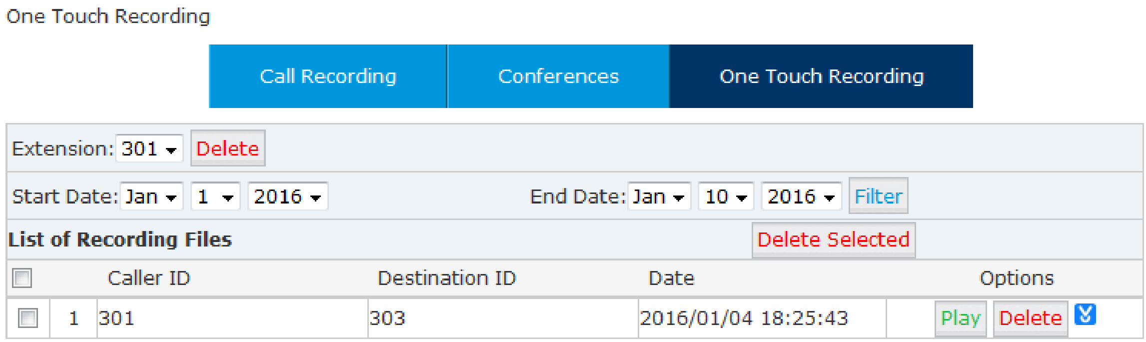
All recorded conferences can be found here on the Report->Record List->Conference page.




| 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

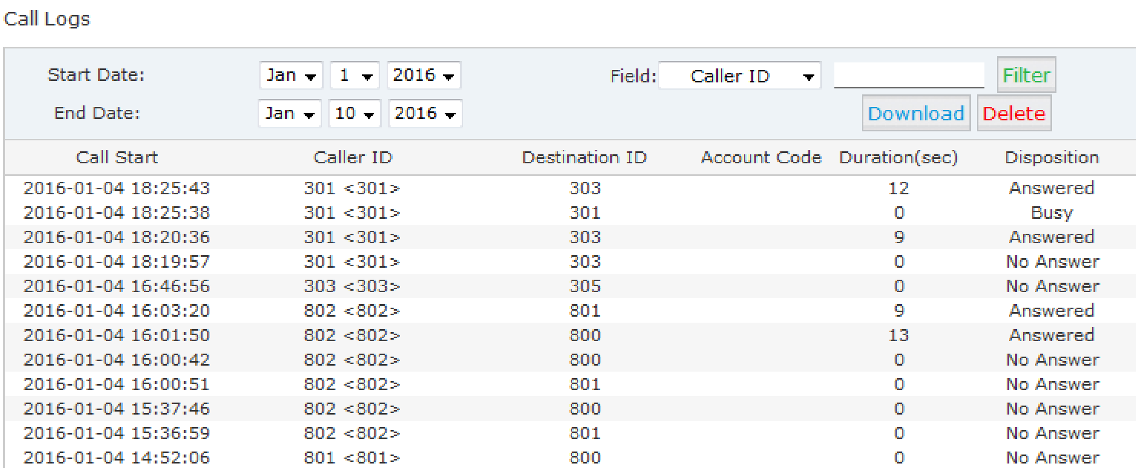
By clicking the “Play” button on a call recording file, you’ll see a dialog like below:


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. |

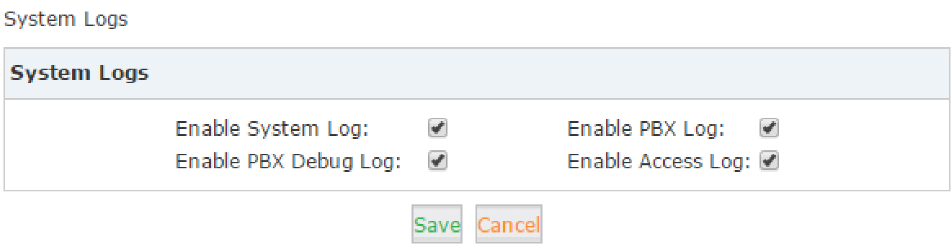
## 6.5 System logs

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

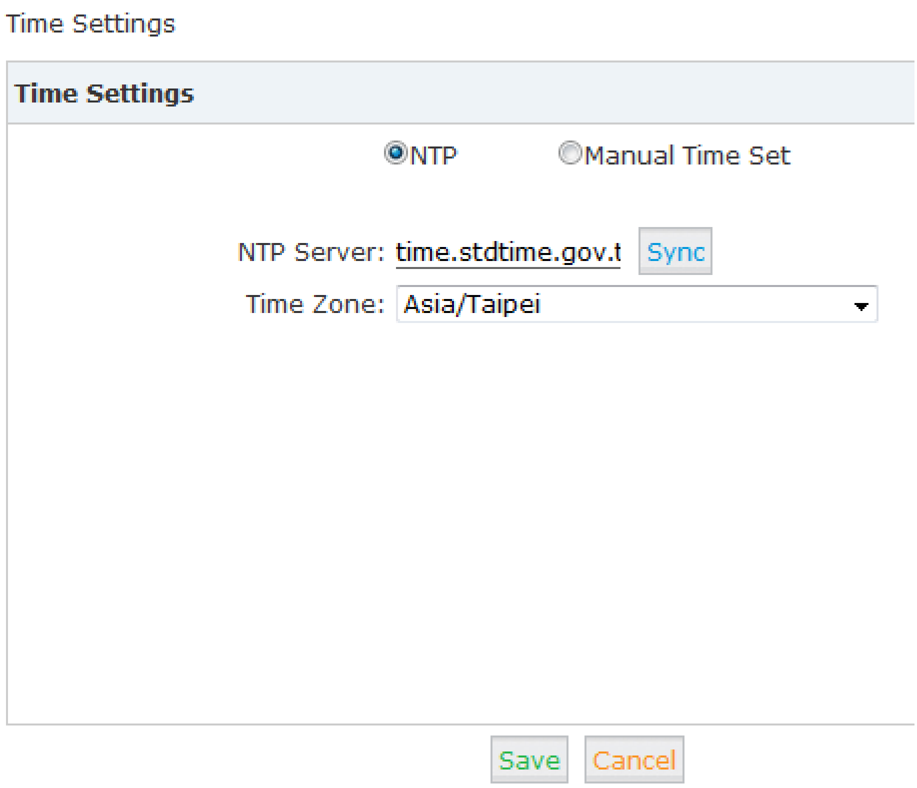
## 7.1 Time Settings

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.

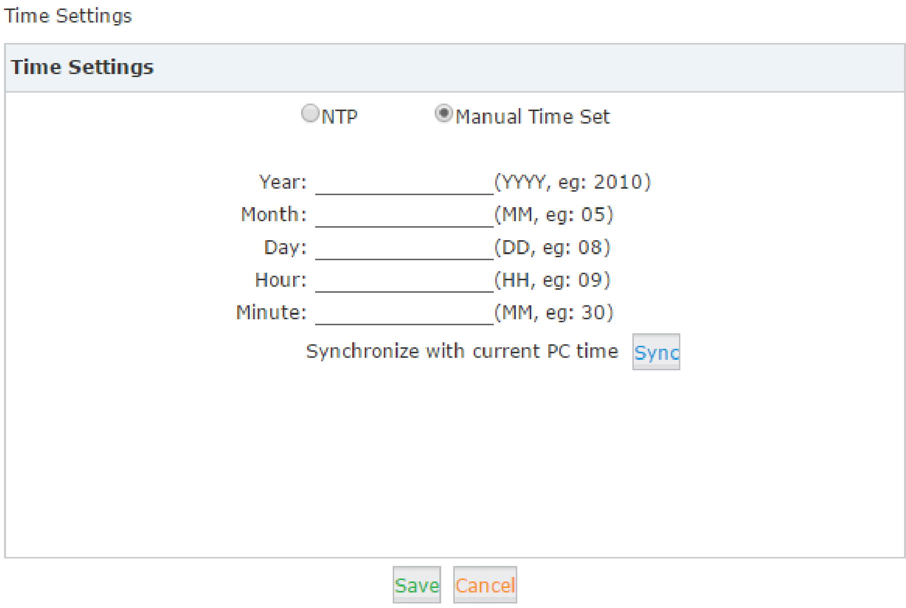



Once done, click the "Sync" button to enable IPPBX system to try to synchronize the current time from the Internet. It might take a while depending on the network conditions. After the process is done, you'll get notice "Sync Failed!" or "Sync Success!". If failed please check if the

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

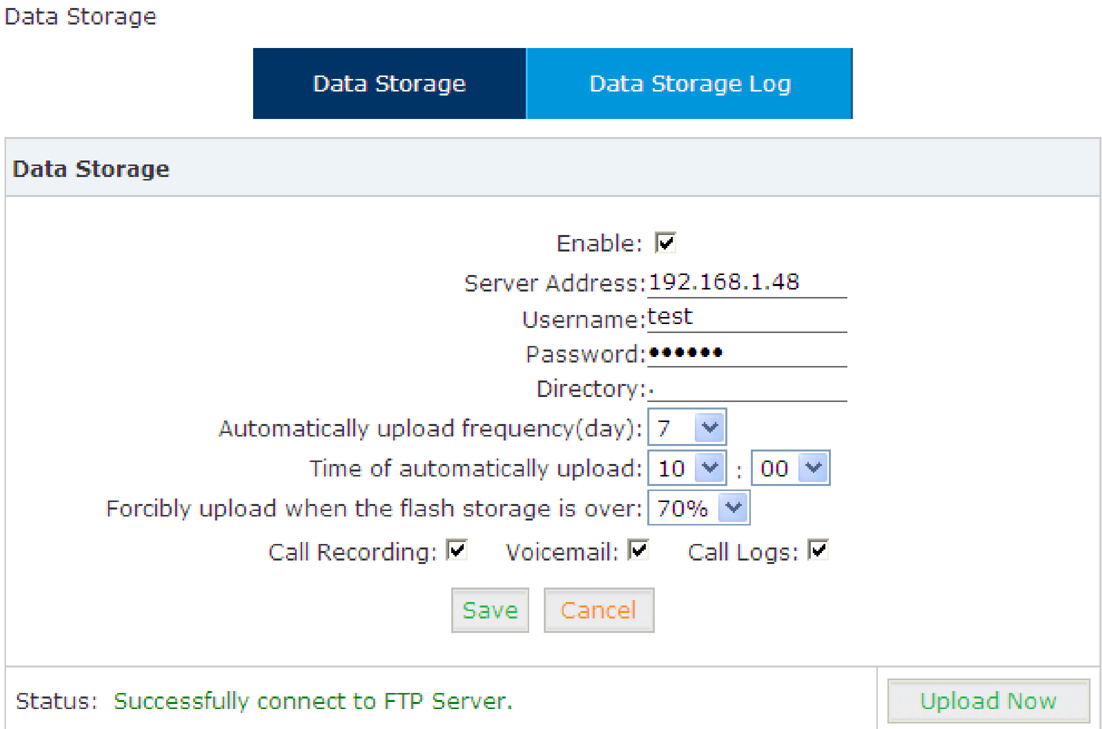
Planet IPX-2200 and IPX-2500 IPPBX systems need proper module settings to load correct drivers and configure files to drive the E1 and BRI telephony modules. Default module settings are with module types FXS/FXO/GSM on both telephony module slots. So if you don't have E1 and BRI modules installed, you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

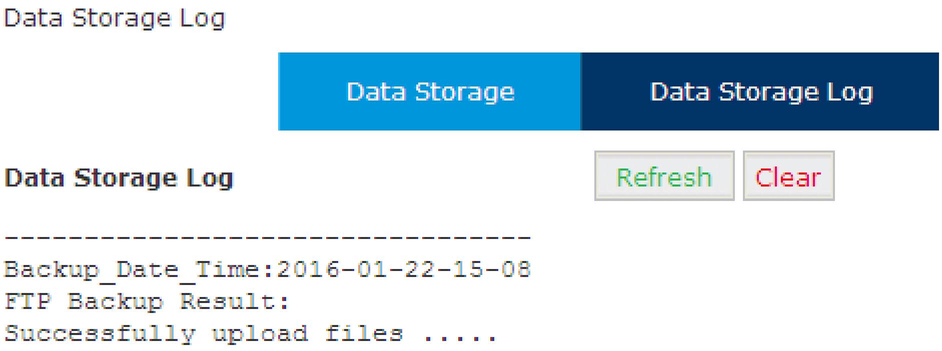
### 7.3.1 Data Storage

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.

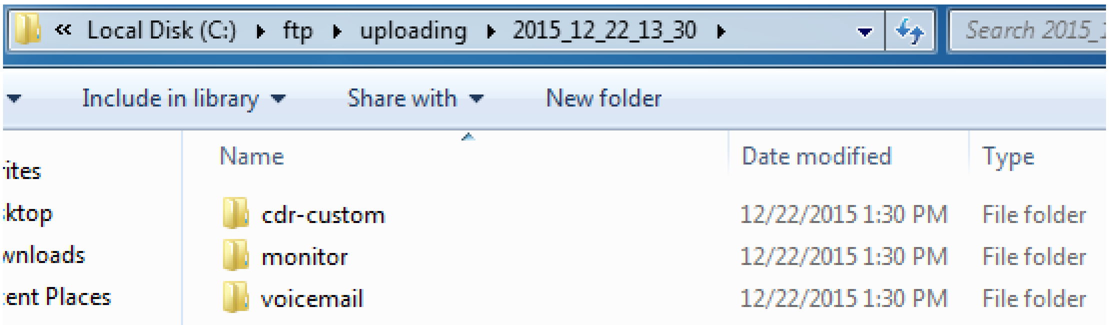



After these settings click “Save” to see the status “Successfully connect to FTP Server.” You can click “Upload Now” to instantly upload a data.

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.

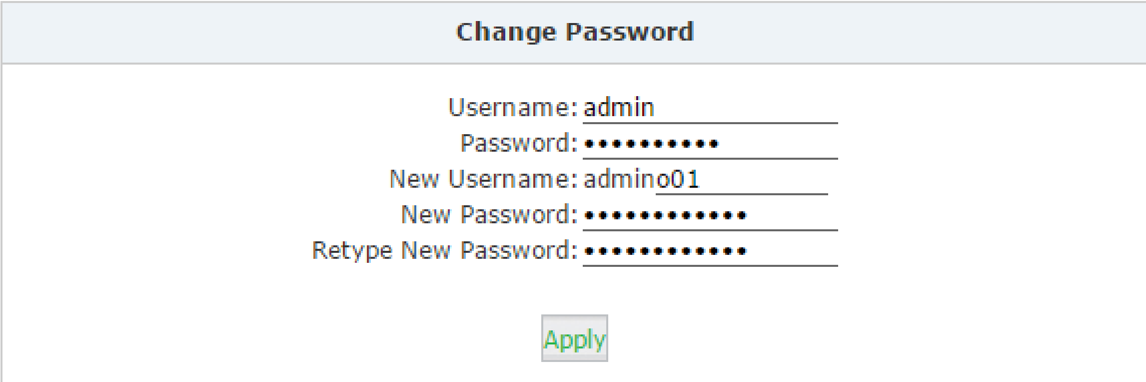



| Note | After each upload except the call logs (Master.csv inside cdr-custom folder), other files will be removed from the IP PBX system, including the call recordings (files inside monitor folder) and voice messages (files inside voicemail folder). So after each upload, you will get only the newly-generated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



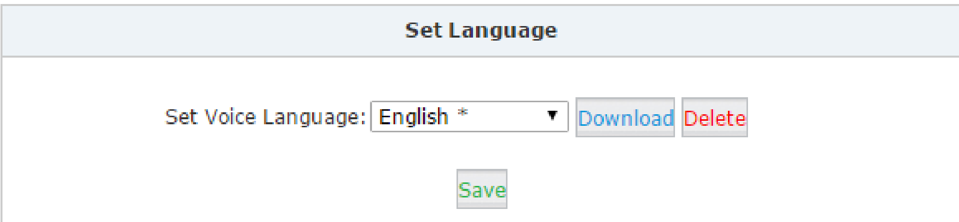

Once completed, click “Apply” to automatically get you logged out and redirected to the login page; now you are able to login with the new username and password.

## 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.

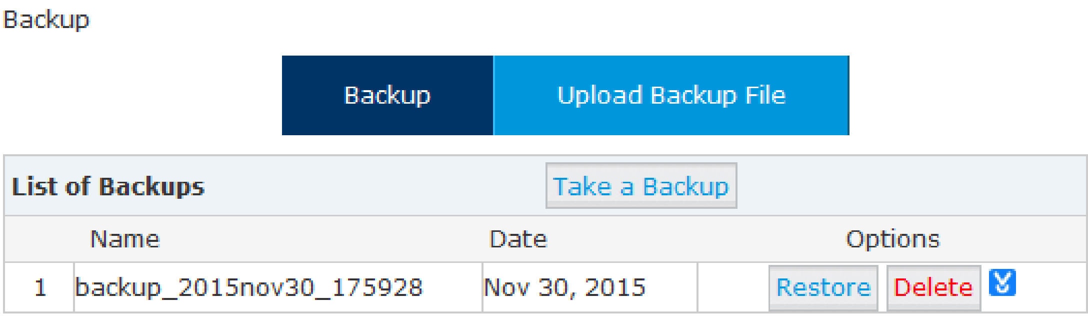
# 7.5 Backup

## 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.

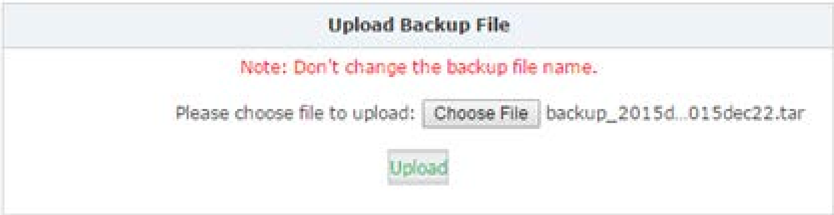



Once done, you get the backup file listed on this page. And the file is stored in the file system. Any time, by clicking the “Restore” button you can restore the configurations. By clicking the “Delete” button you can delete this backup. And you can also download the backup to your computer hard disk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

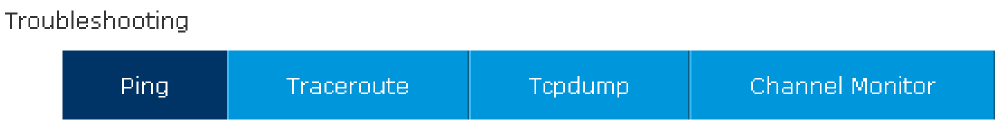
Troubleshooting includes two tools for you to check the network reachability, ping and traceroute. With these tools you'll get an outside view of your network response time and network topology, which allows you to track down possible errors more easily

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 |
|---|---|---|---|

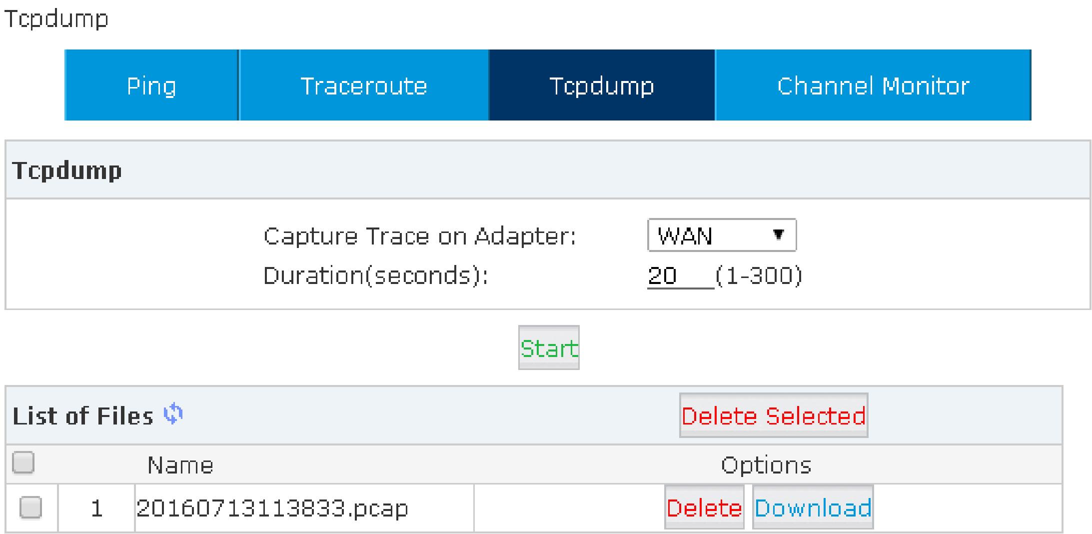
```
Traceroute 8.8.8.8          Run Stop

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.

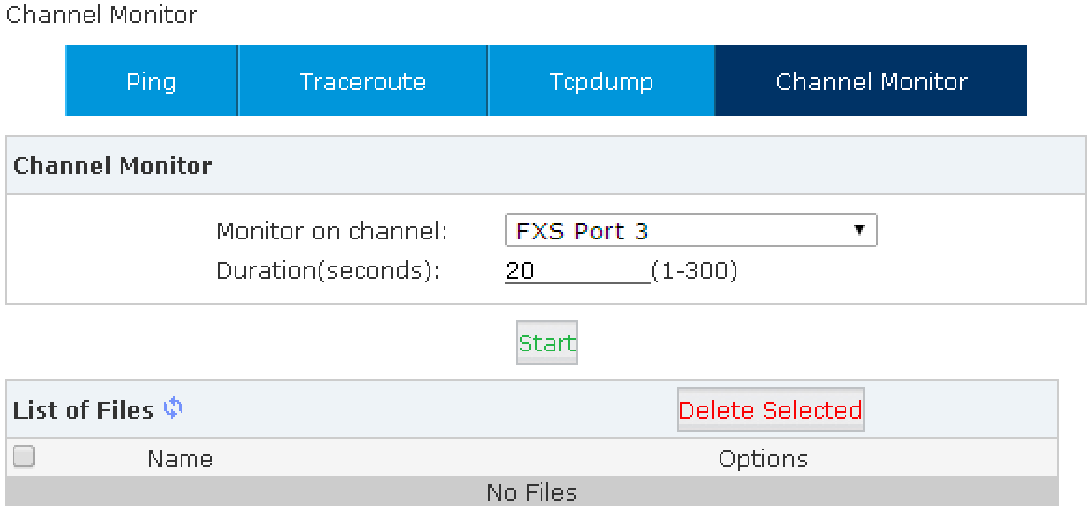



Select an interface and specify the duration of this capture and then click on “Start”. The process will begin and now you can make a call to recur the problem. Once the time is up, the captured packets will be displayed in the “List of Files” section. You can download it to analyze the SIP packets for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

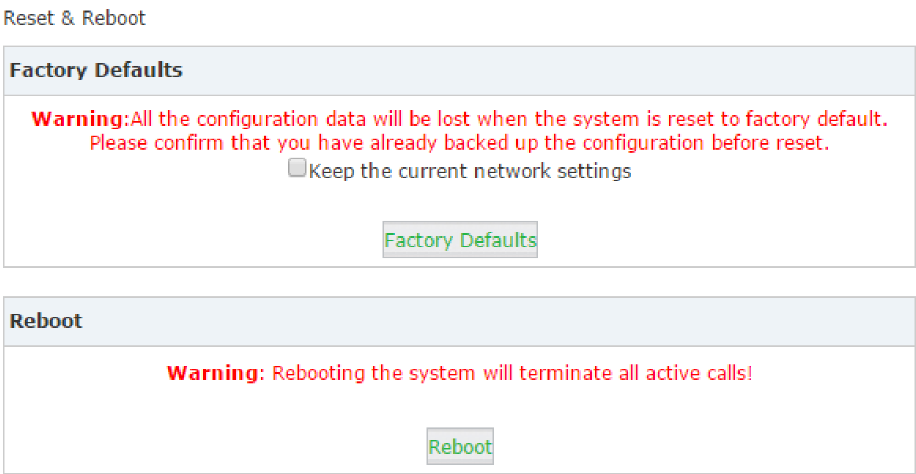
Usually Channel Monitor can be used to capture the caller ID signaling of an FXO channel. If you are experiencing caller ID problem, you can perform channel monitor on the FXO port and then analyze the captured packets.




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**.

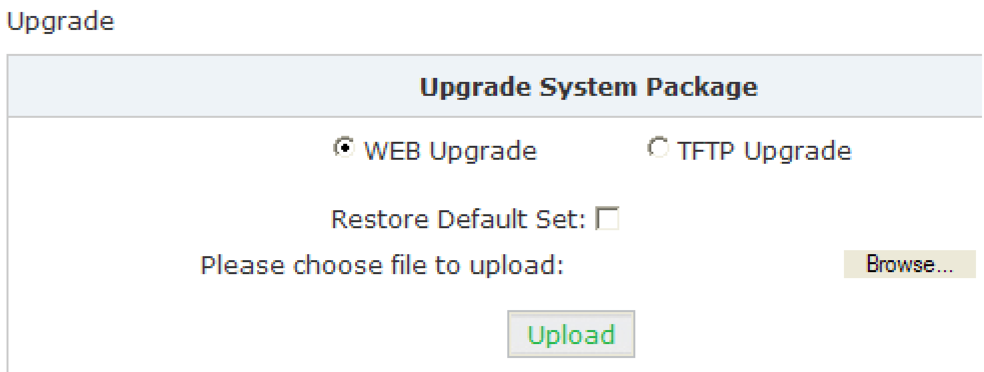
### 7.7.2 Reboot

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



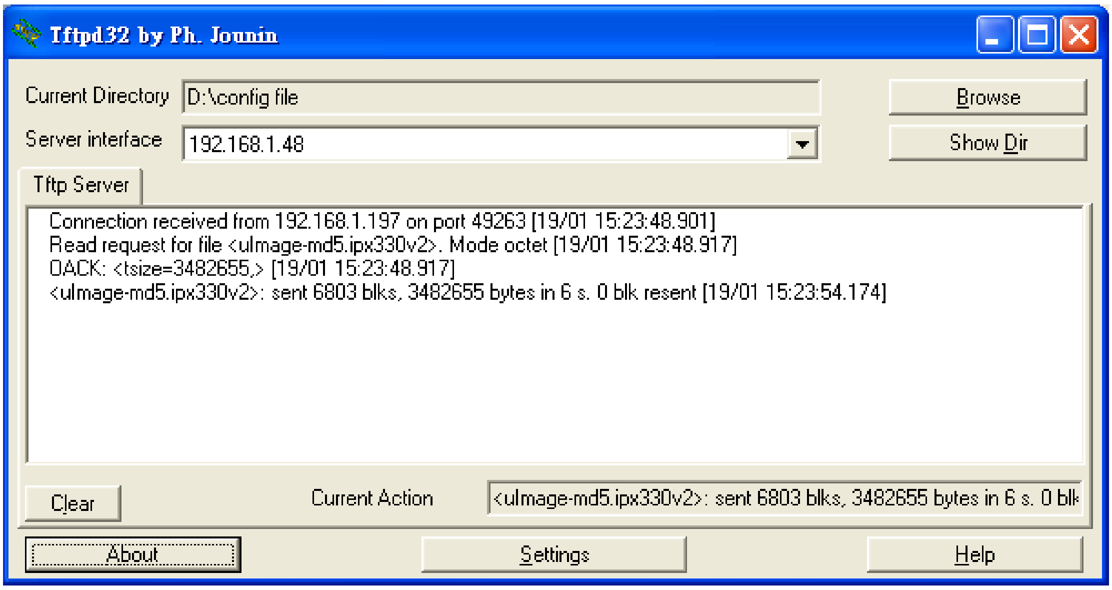

Check the "Web Upgrade" radio button and click the "Browse" button to locate the new firmware in your PC hard drive. Click "Upload" and it will ask you to confirm if restarting the IP PBX system to complete the upgrading process. You can click "Yes" to continue upgrading.

| 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 |
|---|---|

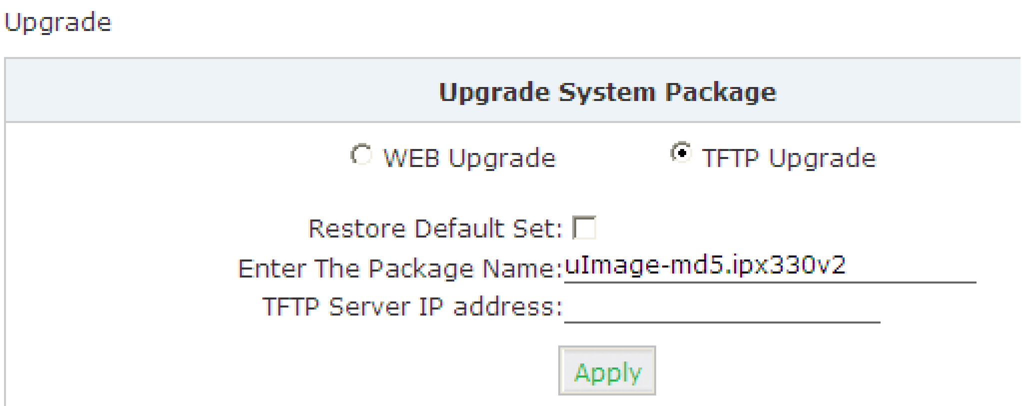
| | |
|---|---|
| | reset the IP PBX system. If you want to reset, it will reset all system configurations including the network profiles. |

## 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.



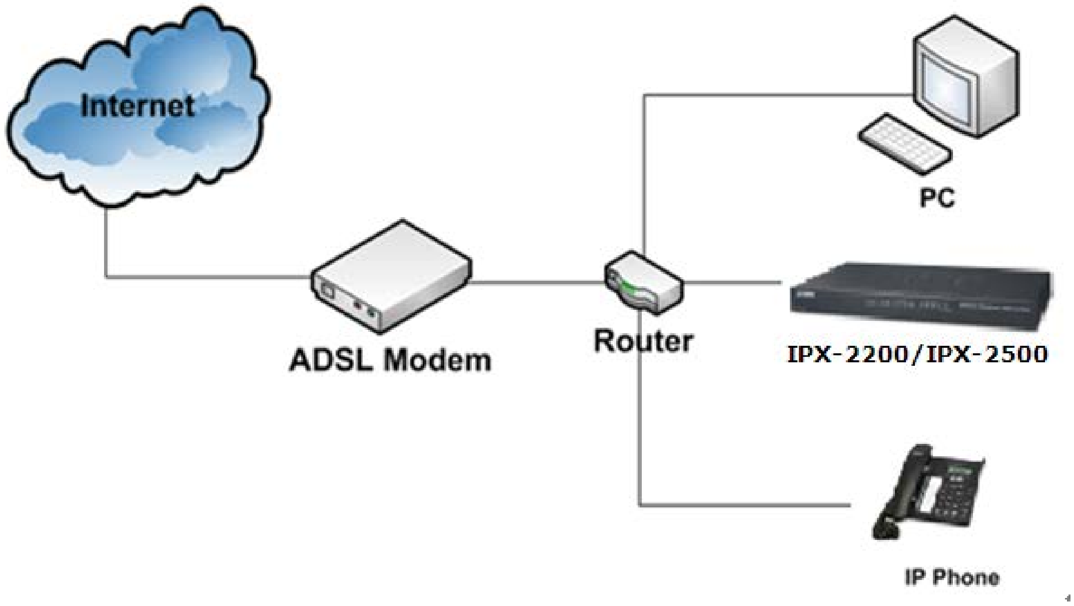

Please double-check the file name and TFTP server IP address and then click "Apply" to enable to upgrade the firmware just like web upgrade.

# Chapter 8. Operating Instructions

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

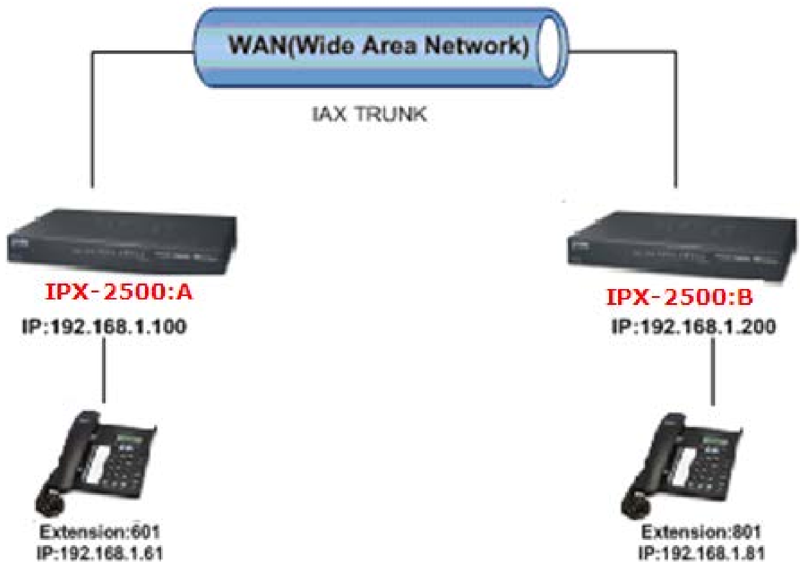
## 8.1 How to connect the IP PBX to the Internet

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:

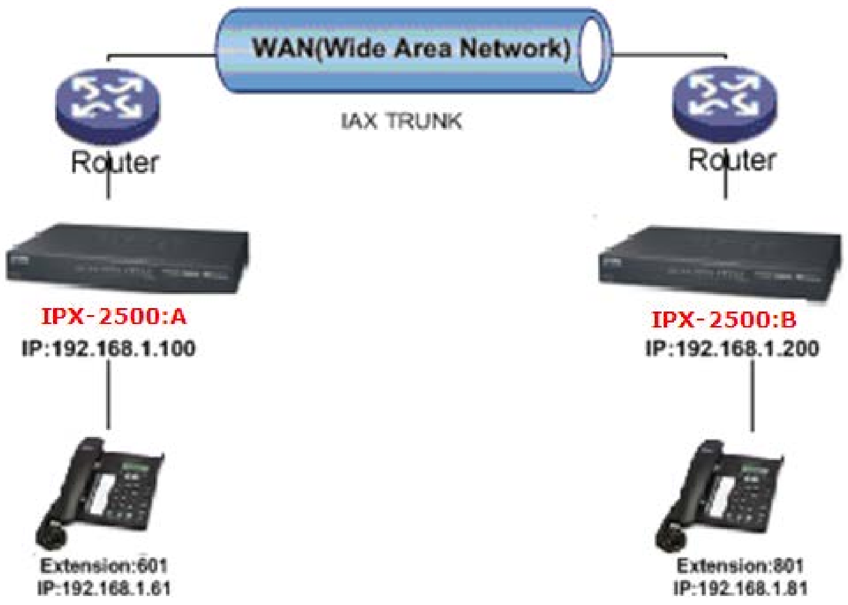
Register IPX-2500-B IP to a trunk of IPX-2500-A with authentication.
Configuration Rule:

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.

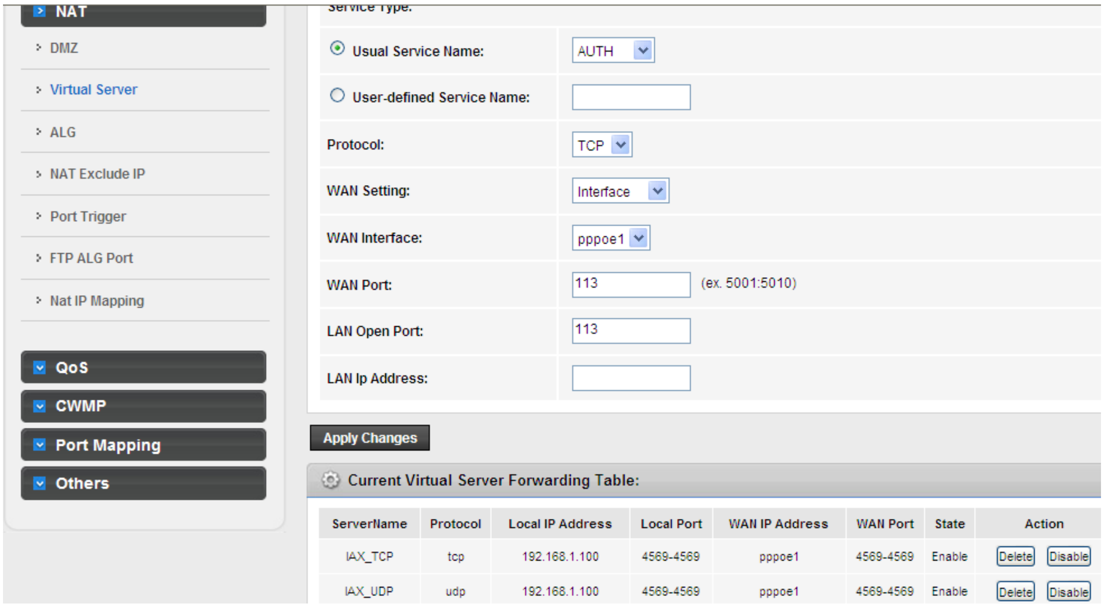



**Step 1:** Configure the mapping rule of IPX-2500-A on the router.

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

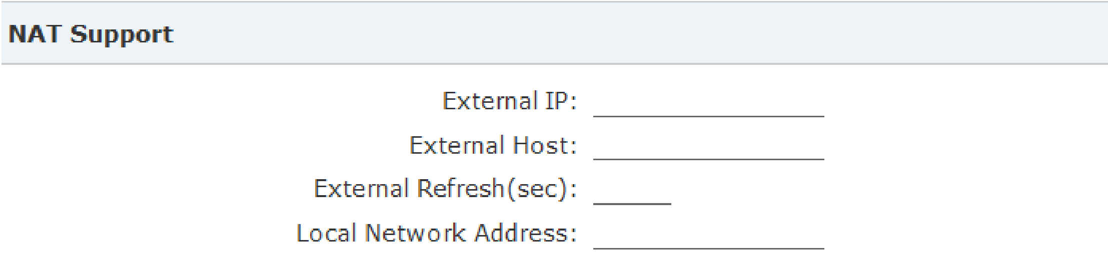
Create extension 601 on IPX-2500-A, extension 801 on IPX-2500-B, and create the correct outbound rule.

| 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.<br>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**:

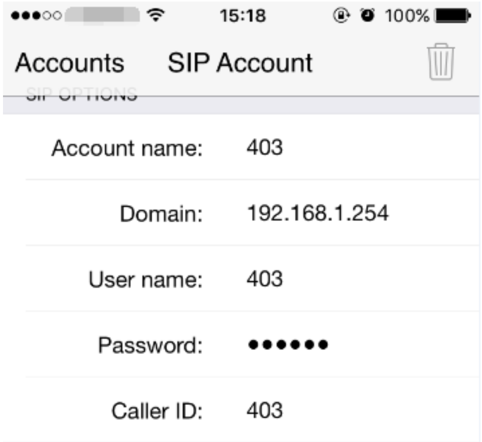
Tap + to create a new account.

**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**:

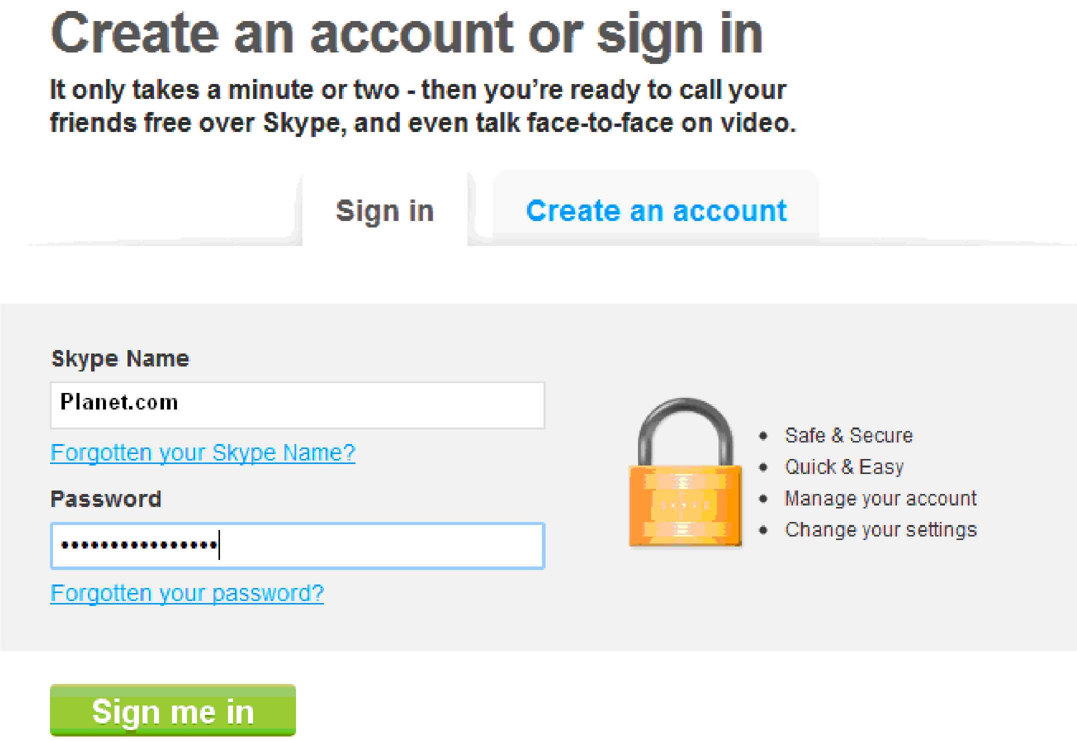
After entering the register credentials, tap Register to register to IP PBX system as an extension.

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

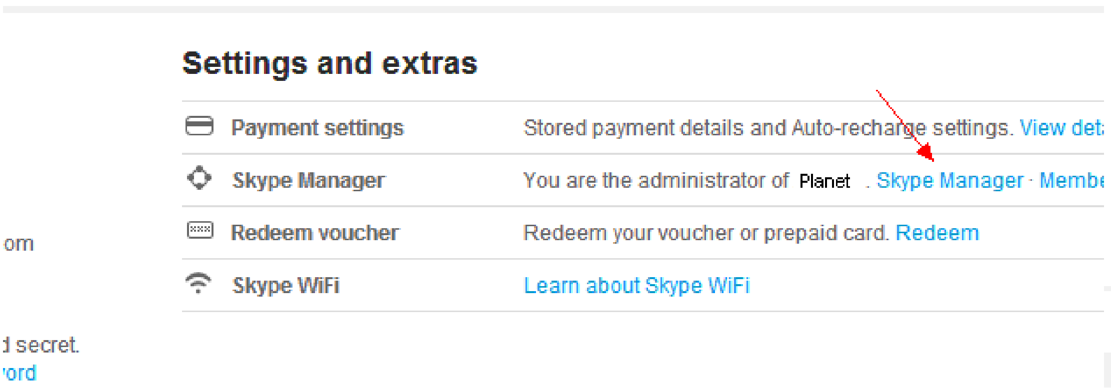
| | |
|---|---|
| Note | The fee of your business account is much more than **€50** when you use the account for the first time. |

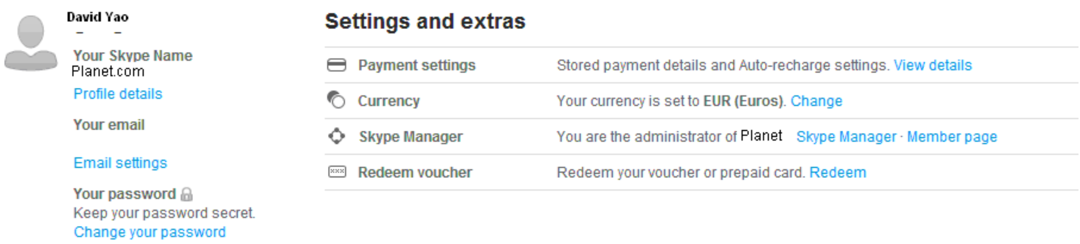
**1** https://login.skype.com

Sign in with the business account.

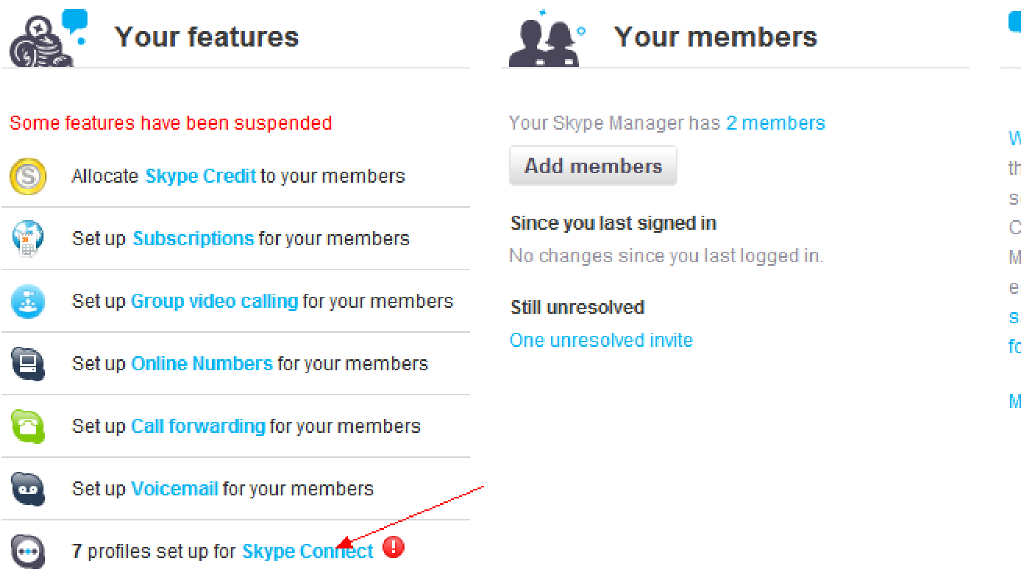


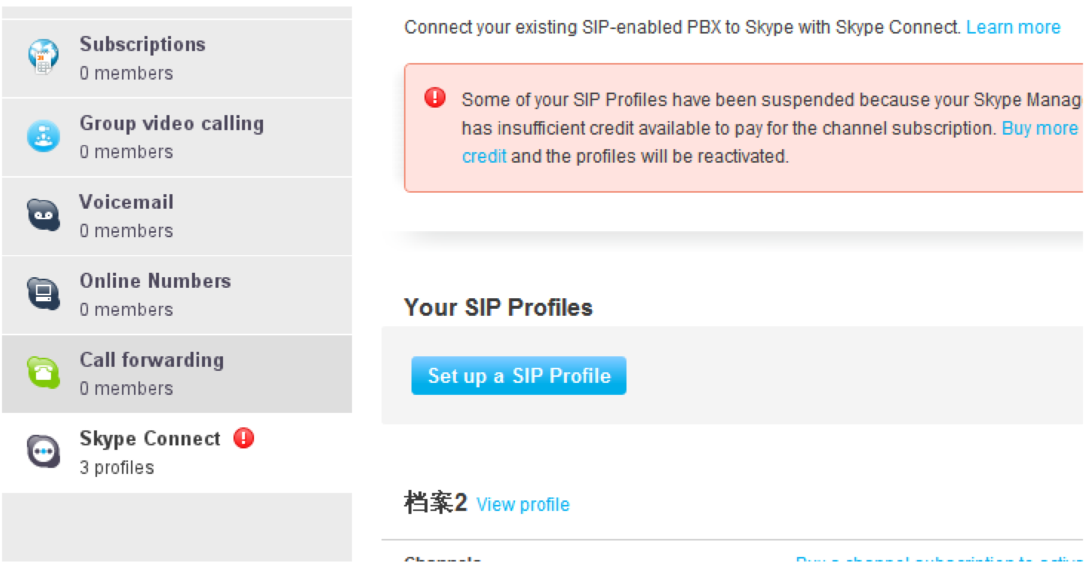

**2** When you have signed in, at the end of this page, you will find the **Skype Manager**, Please click it.



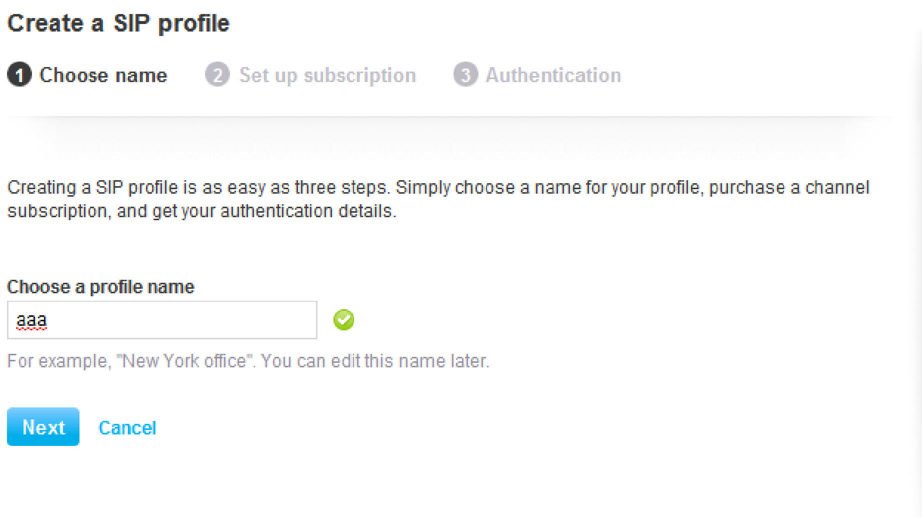



3 Please click the **Skype connect**

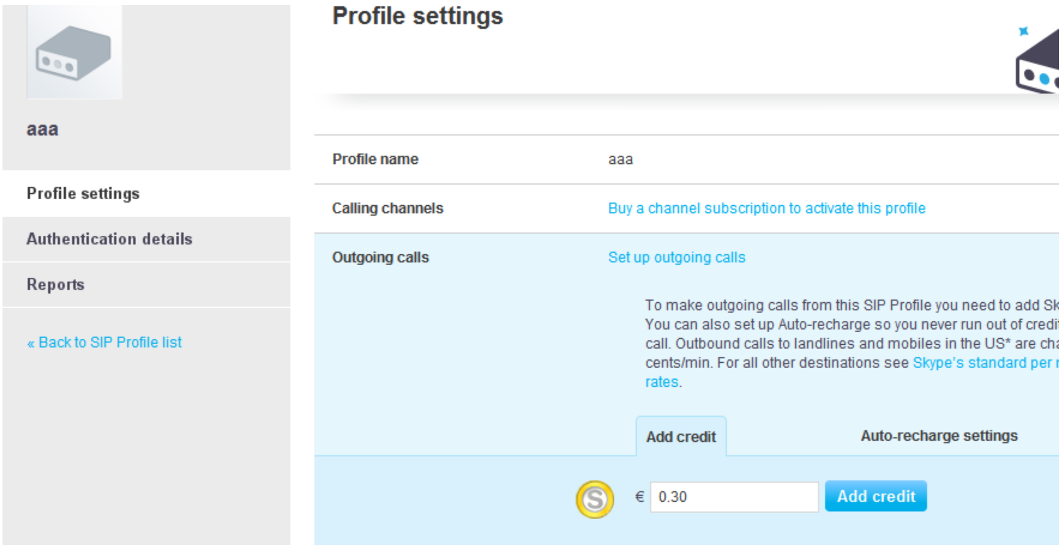


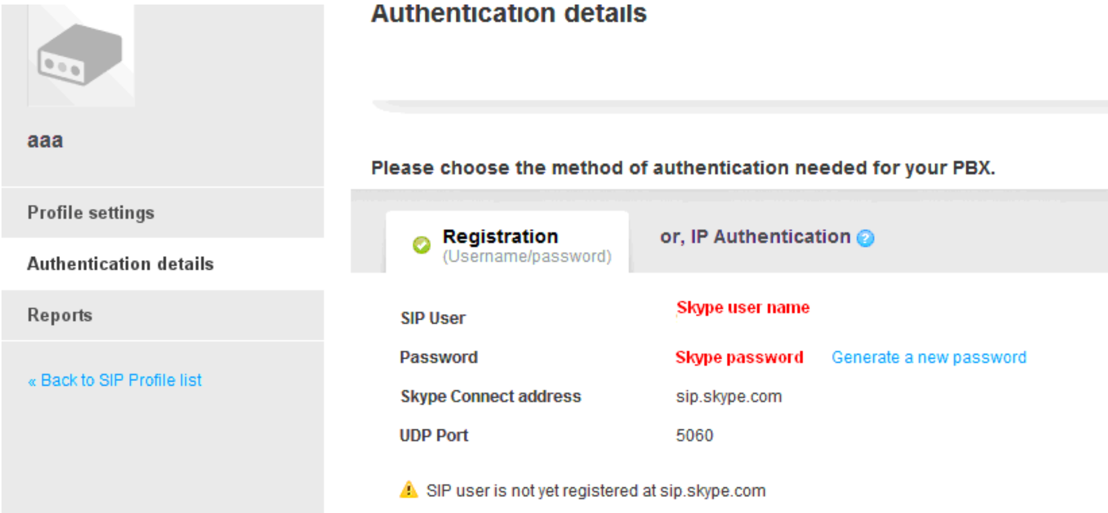
## 4 Create a SIP profile




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

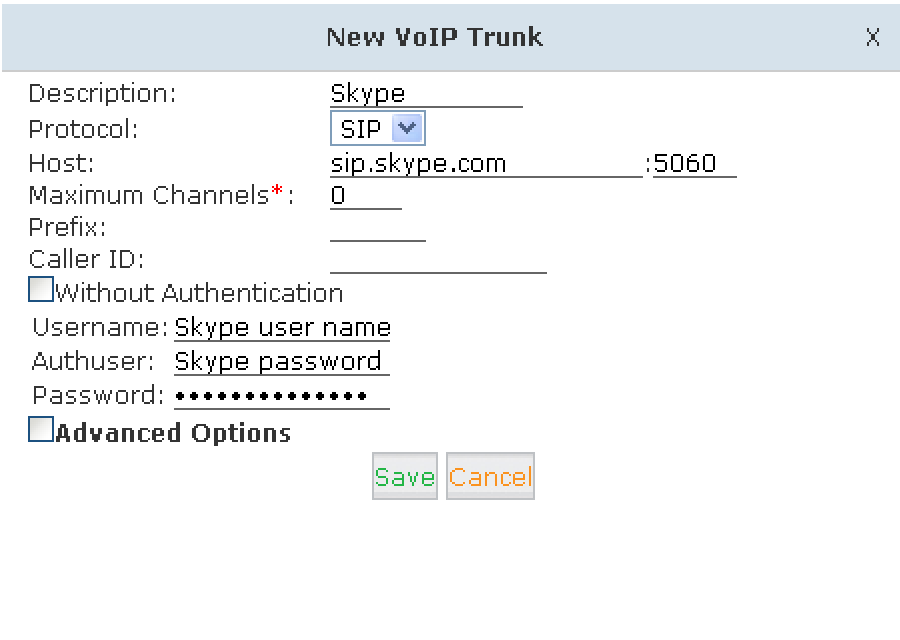
### 5.1 Build one SIP trunk with Skype for SIP account

Provider Type: Custom Trunk

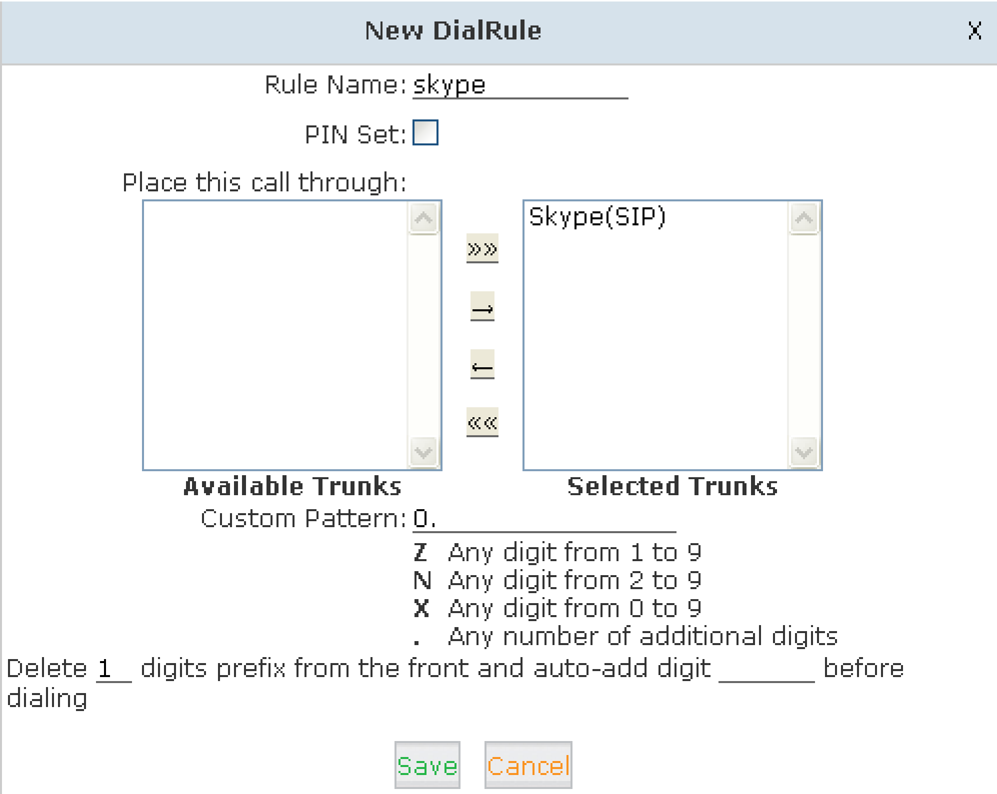
Host: sip.skybe.com

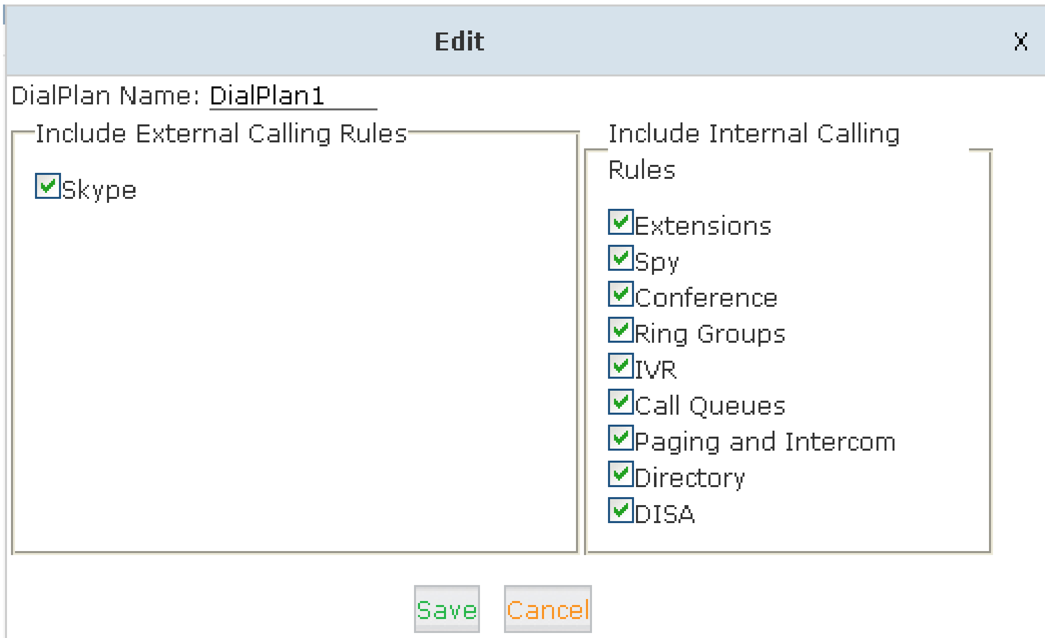
User name: the user name you defined in Authentication detail

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