# PLANET
## Networking & Communication

# User's Manual

## Internet Telephony PBX System

► **IPX-330 / IPX2100**
► **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-330/IPX-2100/IPX-2200/IPX-2500

Rev: 3 (May, 2018)

# Table of Contents

# Introduction

| Model | IPX-330 | IPX-2100 | IPX-2200 | IPX-2500 |
|---|---|---|---|---|
| | | | | |
| Extension User | 30 | 100 | 200 | 500 |
| Concurrent Call | 15 | 30 | 60 | 100 |
| Room Concurrent Call | 10 | 20 | 40 | 80 |
| Recording/Voicemail | 150 hrs | 150 hrs | 150 hrs | 7,500 hrs |
| HDD | - | - | - | Yes |
| Module | 0 | Slot 1/2 for Analog and GSM Module | | |

**Intuitive, Ease-of-Use IP PBX Management**

PLANET next-generation Internet Telephony PBX system is easy to set up and manage thanks to a perceptive web-based user interface and quick setup wizard. Since it is an asterisk-based equipment, the entire benefits of the pre-loaded SIP can be had as any other enterprise-level appliance would come equipped with the feature. The Internet Telephony PBX system is able to accept 30/100/200/500 user registrations, and easy to manage a full voice over IP system with the convenience and cost advantages. The equipment is best paired with PLANET color IP phones to get the function going smoothly and to have a seamless connection between the software and hardware.

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

The IPX-330 comes with 2 FXO ports and the IPX-2000 series integrates up to 8 calls via the IPX-21FO (4 FXO), IPX-21SL (2 FXO + 2 FXS) and IPX-21GS (4 GSM) modules to form a feature-rich PBX system that supports seamless communications between the existing PSTN calls, analog phones, IP phones and SIP-based endpoints.
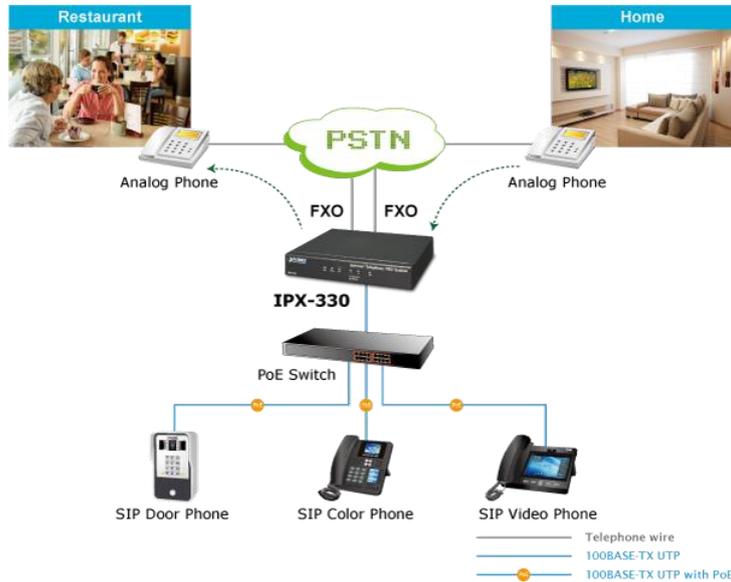


## Replacing Old PBX Easily without New Wiring

Cost-effective, easy-to-install and simple-to-use, the Internet Telephony PBX system converts standard telephones into 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.
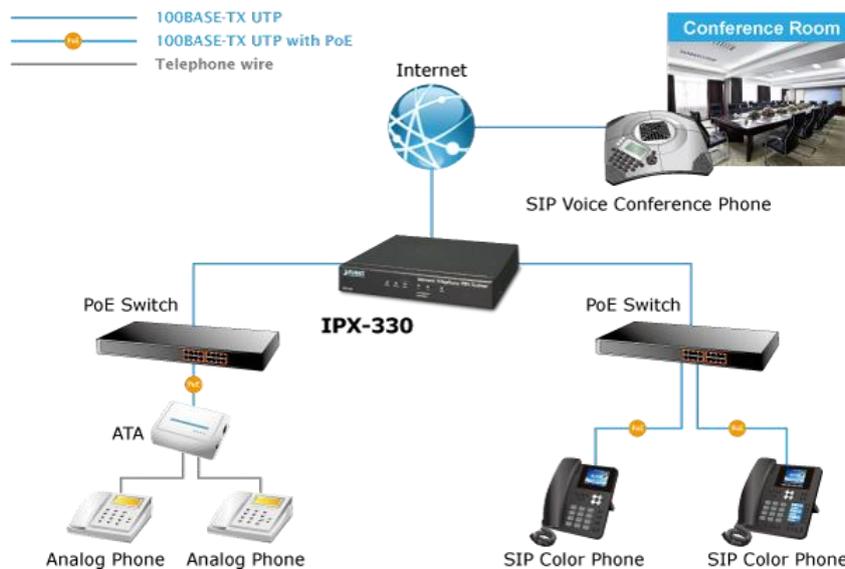


With the Internet Telephony PBX system, home users and companies are able to save the installation cost and extend their past investments in telephones, conferences and speakerphones. The Internet Telephony PBX

system can be the bridge between the traditional analog system and IP network without having to invest heavily.

## Distributed VoIP Network Infrastructure

In the new-generation communication age, the Internet Telephony PBX system supports IPv6 and VPN (client/server) connection to provide users with more flexible and advantageous communications products. With PLANET DDNS function, the Internet Telephony PBX system also helps users to apply and remember the login information easier. Its multiple-language feature helps user to quickly and easily manage the system. Moreover, the Internet Telephony PBX system 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 Internet Telephony PBX system is 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

The Fax to Email/Email to Fax service provided by the Internet Telephony PBX system allows users to transfer and receive faxes directly to or from your email inbox as file attachments. It is an easy and confidential way of receiving, storing and forwarding important fax documents, thus creating a paperless green office.

Green Office (Fax to Email / Email to Fax)

# Features

➢ **System Highlights**

- 15 concurrent calls and up to 30 registers (For IPX-330)
- 30 concurrent calls and up to 100 registers (For IPX-2100)
- 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 to perfect voice quality
- Fax to Email/Email to 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 Language of GUI for international business
- Web-based Control Panel for easy configuration and management of the system
- Strong security features protect your system from hacking
- Supports maximum 8 ports for FXO/GSM (on 2 slots)
- Supports 2 FXO interfaces (For IPX-330)
- Records voice and voicemail to external USB disk
- Supports Lync server

➢ **Codec and Protocol**

- SIP 2.0 (RFC3261), 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)
- DHCP Server/SNMP v1/v2
- IEEE 802.1Q of VLAN
- Supports both IPv6 and IPv4
- Manual configuration of static route table
- Troubleshooting (ping, traceroute)
- VPN Client (Supports N2N, L2TP, PPTP, Open VPN and IPSec)
- VPN Server (Supports PPTP, L2TP, Open VPN and IPSec)
- Refuses SIP Register DoS
- Refuses Abort Invite Dos
- Refuses SSH Login DoS
- Firewall/SRTP
- Enhances HTTPS connection

- ➢ **PBX Features**
    - ■ Black List
    - ■ BLF (Busy Lamp Field)
    - ■ CDR (Call Detailed Record)
    - ■ Conference Room (20 rooms)
    - ■ DID (Direct Inward Dialing Number)
    - ■ DISA (Direct Inward System Access)
    - ■ DND/Feature Codes/Flash Operation Panel
    - ■ Follow Me/Auto-Provision
    - ■ IVR (Interactive Voice Responses)
    - ■ Multi-language System Prompt
    - ■ Multiple-language GUI
    - ■ Phone Book/PIN Set
    - ■ LDAP Server for phonebook
    - ■ Record Files downloaded
    - ■ Ring Group/SIP Trunk
    - ■ Skype for SIP/Smart DID/System Log
    - ■ T.38 Fax (pass-through)/time-based rule
    - ■ Virtual Fax, Voicemail and Voice Mail to E-Mail

- ➢ **Call Features**
    - ■ Call Back, Call Forward, Call Group
    - ■ Call Hold, Call Paging and Intercom
    - ■ Call Park, Call Pickup, Call Queue
    - ■ Call Record, Call Route, Blind Transfer
    - ■ Attend Transfer, Call Waiting
    - ■ Caller ID, Dial by Name
    - ■ Customized IVR, On-hold Music, Transfer
    - ■ Three-way Conferencing, Video Call

# Package Contents

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

| Model | Package Description |
|---|---|
| IPX-330 | ● Quick Installation Guide x 1<br>● Power Adapter x 1 (12V DC)<br>● RJ45 x 1 |
| IPX-2100 | ● Quick Installation Guide x 1<br>● Power Adapter x 1 (12V DC)<br>● RJ45 x 1<br>● Bracket x 2 |
| IPX-2200 | ● Quick Installation Guide x 1<br>● Power Cord x 1 |
| IPX-2500 | ● RJ45 x 1<br>● Bracket x 2 |

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

# Physical Specifications of IPX-330

**Front Panel**
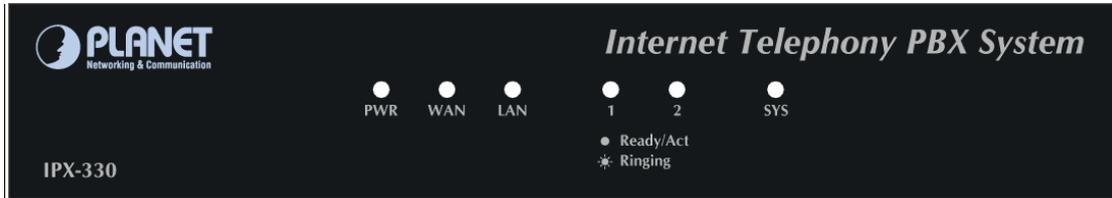


**Rear Panel**
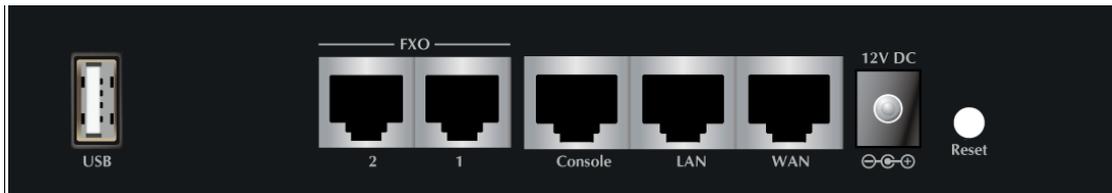


**LED definitions**

| Identification | Indication | Status | Specification |
|---|---|---|---|
| PWR | Power Status | On | Power on |
| | | Off | Power off |
| WAN | WAN Status | On | Connected but no data transmitting |
| | | Blink | Data transmitting |
| | | Off | Disconnected |
| LAN | LAN Status | On | Connected but no data transmitting |
| | | Blink | Data transmitting |
| | | Off | Disconnected |
| FXO 1~ FXO 2 | FXO Status | Red | Channel available |
| | | Off | Channel failure |
| SYS | System Status | On | System initiating |
| | | Blink | System is functioning |
| | | Off | System failure |

# Physical Specifications of IPX-2100

**Front Panel**



**Rear Panel**



**LED definitions**

| Identification | Indication | Status | | Specification |
|---|---|---|---|---|
| PWR | Power Status | On | | Power on |
| | | Off | | Power off |
| SYS | System Status | On | | System initiating |
| | | Blink | | System is functioning |
| | | Off | | System failure |
| WAN | WAN Data Status | On | | Connected but no data transmitting |
| | | Blink | | Data transmitting |
| | | Off | | Disconnected |
| LAN | LAN Data Status | On | | Connected but no data transmitting |
| | | Blink | | Data transmitting |
| | | Off | | Disconnected |
| 1-4 (SLOT1/2) | SLOT 1/2 Status | FXS | Green | Channel available |
| | | | Off | Channel failure |
| | | FXO | Red | Channel available |
| | | | Off | Channel failure |
| | | GSM | Red | Channel available |
| | | | Off | Channel failure |
| | | WCDMA | Red | Channel available |
| | | | Off | Channel failure |

# Physical Specifications of IPX-2200 / IPX-2500

**Front Panel**



**Front Panel of the IPX-2200**



**Front Panel of the IPX-2500**

**Rear Panel**



**Rear Panel of the IPX-2200/IPX-2500**

**LED definitions**

| Identification | Indication | Status | | | Specification |
|---|---|---|---|---|---|
| PWR | Power States | Green | | | Power On |
| | | Off | | | Power Off |
| SYS | System States | Wink | | | System is Running |
| | | Off | | | System Booting or Failed |
| WAN/LAN | WAN/LAN Interface States | Wink | | | Data Transmitting |
| | | Off | | | No Data Transmitting |
| 1-4 (Slot 1/2) | Slot 1 and Slot 2 States | FXS | | Green | Channel Loading Succeeded |
| | | | | Off | Channel Loading Failed |
| | | FXO | | Red | Channel Loading Succeeded |
| | | | | Off | Channel Loading Failed |
| | | GSM/WCDMA | | Red | Channel Loading Succeeded |
| | | | | Off | Channel Loading Failed |
| | | E1/T1 (PRI/R2) | L1 | Red | Module Loading Succeeded |
| | | | | Off | Module Loading Failed |
| | | | L2/L3 | Red/Off | CPE Signaling |
| | | | | Green/Off | NET Signaling |
| | | | | Off/Red | SS7 Signaling |
| | | | | Off/Green | R2 Signaling |
| | | | L4 | Green | Connected (No Alarm) |
| | | | | Red | Disconnected (Alarm) |
| | | BRI | | Red | TE Mode |
| | | | | Green | NT Mode |
| | | | | Off | Module Loading Failed |

**Physical interface description**

| 1 | **Power Switch** | Switch the power on or off |
|---|---|---|
| 2 | **Power Cord** | 100~240V AC, 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 | 1. Supporting 2 slots, user can buy expansion module like IPX-21FO (4 x FXO) or IPX-21GS (4 x GSM) for extending port service.<br>2. If IPX-21GS is used, please disable pin lock function. Otherwise, dial out or call in will not be available. |

# Plug-in modules for IPX-2100/IPX-2200/IPX-2500

● **IPX-21FO Module**

IPX-21FO module provides 4 FXO interfaces for connecting PSTN lines provided by the telecom. It can be installed on both slots of IPX-2100, IPX-2200 and IPX-2500 and provides maximal 8 FXO interfaces.

● **IPX-21GS Module**

IPX-21GS module provides 4 GSM channels; it can be installed on IPX-2100, IPX-2200 and IPX-2500 for making and receiving phone calls from GSM network. It is designed with SIM900 for global market; SIM900 is a quad-band GSM engine that works on frequencies GSM 850MHz, EGSM 900MHz, DCS 1800MHz and PCS 1900MHz.

● **IPX-21FS Module (by project order)**

The IPX-21FS module provides 4 FXS interfaces for connecting fax machines or analog phones. It can be installed on both slots of IPX-2100, IPX-2200 and IPX-2500 and provides maximal 8 FXS interfaces.

● **IPX-21SL Module (by project order)**



The IPX-21SL module provides 2 FXO and 2 FXS interfaces; it can be installed on both slots of the IPX-2100, IPX-2200 and IPX-2500. With the IPX-21SL module installed, it enables the IPPBX system with FXO to FXS lifeline feature. When there's power failure, you may still use the analog phone to make and receive phone calls.

● **IPX-21MA Module (by project order)**



The IPX-21MA (WCDMA) module provides 4 GSM/3G voice channels; it can be installed on IPX-2100, IPX-2200 and IPX-2500 for making and receiving phone calls from the GSM/3G network. It is designed with SIM5320 series module for global market, the SIM5320 is a dual-band WCDMA and quad-band GSM engine that works on frequencies like UMTS 850MHz, UMTS 900MHz, UMTS 1900MHz, UMTS 2100MHz, GSM 850MHz, EGSM 900MHz, DCS 1800MHz and PCS 1900MHz.

| | |
|---|---|
| **Note** | WCDMA modules for PLANET IP PBX series IPPBXs are only used for voice phone calls; they CANNOT be used for data transmission from 3G network. |

● **IPX-21BR Module   (by project order)**



The IPX-21BR module provides 4 BRI interfaces which can be configured to work in NT or TE mode. It can be installed on IPX-2200 and IPX-2500 but not IPX-2100. Regarding the IPX-2200 and IPX-2500, only one 4BRI module can be installed. And if it's going to be installed with other modules (IPX-21FO, IPX-21FS, IPX-21GS, IPX-21MA modules), it should be installed in Slot 2. The IPX-21BR cannot be installed with E1 module like the IPX-2200 or IPX-2500.

● **IPX-21PR Module (by project order)**



The E1/T1 module provided a RJ48 interface which could be configured to work in E1 (PRI-NET, PRI-CPE, R2, SS7 signaling) and T1 mode. You can install 2 E1/T1 modules on both IPX-2200 and IPX-2500. If it's going to be installed with other modules (IPX-21FO, IPX-21FS, IPX-21GS and IPX-21MA), it should be installed on Slot 2. And the E1/T1 module cannot be installed with BRI module like the IPX-2200 or IPX-2500.

# Specifications



| Product | IPX-330<br>Internet Telephony PBX system (30 SIP Users registrations) |
|---|---|
| **Hardware** | |
| LAN | 1 x 100BASE-TX RJ45 for LAN |
| WAN | 1 x 100BASE-TX RJ45 for WAN |
| Analog Ports | 2 x FXO |
| USB | Store data for external disk |
| Console | Console Interface |
| Reset Button | Reset to factory default |
| **Protocols and Standard** | |
| Standard | SIP 2.0 (RFC 3261), IAX2 |
| Request for Comments (RFCs) | RFC 793 TCP<br>RFC 826 ARP<br>RFC 1034, 1035 DNS<br>RFC 1631 NAT<br>RFC 2068 HTTP<br>RFC 2131 DHCP<br>RFC 2516 PPPoE |

| | RFC 3261, RFC 3311, RFC 3515 |
| --- | --- |
| | RFC 3265, RFC 3892, RFC 3361 |
| | RFC 3842, RFC 3389, RFC 3489 |
| | RFC 3428, RFC 2327, RFC 2833 |
| | RFC 2976, RFC 3263 |
| Voice Codec | G.711-Ulaw, G.711-Alaw, G.722, G.726, G.729, GSM, Speex and Opus |
| Video Codec | H.261, H.263, H.263+, H.264 and VP8 |
| 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>RFC 4733, SIP info, in-band and Auto |
| Protocols | SIP 2.0 (RFC 3261), TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE |
| **Internet Sharing** | |
| Network Features | DDNS client (Planet DDNS and easy DDNS)<br>DHCP server, SNMP v1/v2<br>IEEE 802.1Q of VLAN<br>IP assignment (DHCP/Static)<br>IPv4, IPv6<br>Manual configuration of static route table<br>Troubleshooting (ping, traceroute)<br>VPN server ,VPN client |
| Security Features | Refuse SIP Register DoS<br>Refuse Abort Invite Dos<br>Refuse SSH Login DoS<br>Firewall and enhances HTTPS connection |
| **Features** | |
| PBX Features | SIP Register with UDP/TCP/TLS<br>Phone Auto-Provision<br>One Touch Recording<br>Mobility Extension<br>Blacklist<br>BLF (Busy Lamp Field)<br>CDR (Call Detailed Record)<br>Conference Room<br>DID (Direct Inward Dialing Number)<br>DISA (Direct Inward System Access)<br>DNIS (Dialed Number Identification Service)<br>SRTP (Secure Realtime Transport Protocol) |

| | |
|---|---|
| | DND (Do Not Disturb) |
| | FOP (Flash Operation Panel) Status Monitoring |
| | IVR (Interactive Voice Responses) |
| | Follow Me, Call Spy and PIN Set |
| | Distinctive Ringtone |
| | Multi-language System Prompt |
| | Multiple Language of GUI |
| | Phone Book, Speed Dial |
| | LDAP Server for phonebook |
| | Record Files Download |
| | Ring Group, SIP Trunk |
| | Skype for SIP, Smart DID, System Log |
| | T.38 fax (pass-through), voicemail and voicemail to e-mail |
| | Time-based Rule |
| | PBX log, web access log and PBX debug log |
| Call Features | Call Back, Call Forward, Call Group |
| | Call Hold, Call Paging and Intercom |
| | Call Park, Call Pickup, Call Queue |
| | Call Record, Call Route, Blind Transfer |
| | Attend Transfer, Call Waiting |
| | Caller ID, Dial by Name |
| | Customized IVR, On-hold Music, Transfer |
| | Three-way Conferencing, Video Call |

## System Capacity

| | |
|---|---|
| System Capacity | 15 concurrent call legs<br>Up to 30 IP phone registers/extensions<br>Recording and Voicemail: 150 hours |

## Network and Configuration

| | |
|---|---|
| Access Mode | Static IP, DHCP |
| LED Indications | SYS: 1, LNK/Off<br>WAN: 1, LNK/Off<br>LAN: 1, LNK/Off<br>PWR: 1, LNK/Off<br>FXO: 1~2, Red |
| Dimensions (W x D x H) | 170 x 105 x 30 mm |
| Operating Environment | 0~40 degrees C, 5~95% humidity |
| Power Requirements | Input: 100 ~ 240V AC<br>Output: DC 12V 1A |
| EMC/EMI | CE, FCC Class B, RoHS |

Fully Functional **Asterisk**-based Advanced Internet Business Telephone System
100 SIP User Registrations Internet Telephony PBX System ▶ IPX-2100

| Product | IPX-2100<br>Internet Telephony PBX system (100 SIP Users registrations) |
|---|---|
| **Hardware** | |
| LAN | 1 x 100BASE-TX RJ45 for LAN |
| WAN | 1 x 100BASE-TX RJ45 for WAN |
| 2 Slots | Supports maximum 8 ports (FXS/FXO/GSM) |
| USB | Store data for external disk |
| Console | Console Interface |
| Reset Button | Reset to factory default |
| **Protocols and Standard** | |
| Standard | SIP 2.0 (RFC 3261), IAX2 |
| Protocols | RFC 793 TCP<br>RFC 826 ARP<br>RFC 1034, 1035 DNS<br>RFC 1631 NAT<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 |

| | RFC 3428, RFC 2327, RFC 2833 |
|---|---|
| | RFC 2976, RFC 3263 |
| Voice Codec | G.711-Ulaw, G.711-Alaw, G.722, G.726, G.729, GSM, Speex and Opus |
| Video Codec | H.261, H.263, H.263+, H.264 and VP8 |
| Fax over IP | T.38 Fax (pass-through)<br>Note: T.38 support is dependent on fax machine, SIP provider and network/ transport resilience |
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| **Internet Sharing** | |
| Network Features | DDNS client (Planet DDNS and easy DDNS)<br>DHCP server/SNMP v1/v2<br>IEEE 802.1Q of VLAN<br>IP assignment (DHCP/Static)<br>IPv4, IPv6<br>Manual configuration of static route table<br>Troubleshooting (ping, traceroute)<br>VPN server ,VPN client |
| Security Features | Refuse SIP Register DoS<br>Refuse Abort Invite Dos<br>Refuse SSH Login DoS<br>Firewall and enhances HTTPS connection |
| **Features** | |
| PBX Features | SIP Register with UDP/TCP/TLS<br>Phone Auto-Provision<br>One Touch Recording<br>Mobility Extension<br>Black List<br>BLF (Busy Lamp Field)<br>CDR (Call Detailed Record)<br>Conference Room<br>DID (Direct Inward Dialing Number)<br>DISA (Direct Inward System Access)<br>DNIS (Dialed Number Identification Service)<br>SRTP (Secure Realtime Transport Protocol)<br>DND (Do Not Disturb)<br>FOP (Flash Operation Panel) Status Monitoring<br>IVR (Interactive Voice Responses) |

| | Follow Me, Call Spy and PIN Set |
|---|---|
| | Distinctive Ringtone |
| | Multi-language System Prompt |
| | Multiple Language of GUI |
| | Phone Book, Speed Dial |
| | LDAP Server for phonebook |
| | Record Files Download |
| | Ring Group, SIP Trunk |
| | Skype for SIP, Smart DID, System Log |
| | T.38 fax (pass-through), voicemail and voicemail to e-mail |
| | Time-based Rule |
| | PBX log, web access log and PBX debug log |
| Call Features | Call Back, Call Forward, Call Group |
| | Call Hold, Call Paging and Intercom |
| | Call Park, Call Pickup, Call Queue |
| | Call Record, Call Route, Blind Transfer |
| | Attend Transfer, Call Waiting |
| | Caller ID, Dial by Name |
| | Customized IVR, On-hold Music, Transfer |
| | Three-way Conferencing, Video Call |

### System Capacity

| | |
|---|---|
| System Capacity | 30 concurrent call legs |
| | Up to 100 IP phone registers/extensions |
| | Recording and Voicemail: 150 hours |

### Network and Configuration

| | |
|---|---|
| Access Mode | Static IP, DHCP |
| LED Indications | SYS: 1, LNK/Off |
| | WAN: 1, LNK/Off |
| | LAN: 1, LNK/Off |
| | PWR: 1, LNK/Off |
| | SLOT: FXO/GSM (Red), FXS (Green) |
| Dimensions (W x D x H) | 343 x 154 x 35 mm |
| Operating Environment | 0~40 degrees C, 5~95% humidity |
| Power Requirements | Input: 100 ~ 240V AC |
| | Output: DC 12V 2A |
| EMC/EMI | CE, FCC Class B, RoHS |

| Product | IPX-2200<br>Internet Telephony PBX system (200 SIP Users registrations) |
|---|---|
| **Hardware** | |
| 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<br>Store data for external disk |
| Audio In/Out | For external paging |
| 2 Slots | Supports maximum 8 ports (FXO/GSM) |
| **Protocols and Standard** | |
| Standard | SIP 2.0 (RFC 3261), IAX2 |
| Protocols | RFC 793 TCP<br>RFC 826 ARP<br>RFC 1034, 1035 DNS<br>RFC 1631 NAT<br>RFC 2068 HTTP<br>RFC 2131 DHCP<br>RFC 2516 PPPoE<br>RFC 3261, RFC 3311, RFC 3515 |

| | RFC 3265, RFC 3892, RFC 3361 |
|---|---|
| | RFC 3842, RFC 3389, RFC 3489 |
| | RFC 3428, RFC 2327, RFC 2833 |
| | RFC 2976, RFC 3263 |
| | TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE |
| Voice Codec | G.711-Ulaw, G.711-Alaw, G.722, G.726, G.729, GSM, Speex and Opus |
| Video Codec | H.261, H.263, H.263+, H.264 and VP8 |
| 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>RFC 4733, SIP info, in-band and Auto |
| **Internet Sharing** | |
| Network Features | DDNS client (Planet DDNS and easy DDNS)<br>DHCP server/SNMP v1/v2<br>IEEE 802.1Q of VLAN<br>IP assignment (DHCP/Static)<br>IPv4, IPv6<br>Manual configuration of static route table<br>Troubleshooting (ping, traceroute)<br>VPN server ,VPN client |
| Security Features | Refuse SIP Register DoS<br>Refuse Abort Invite Dos<br>Refuse SSH Login DoS<br>Firewall and enhances HTTPS connection |
| **Features** | |
| PBX Features | SIP Register with UDP/TCP/TLS<br>Phone Auto-Provision<br>One Touch Recording<br>Mobility Extension<br>Black List<br>BLF (Busy Lamp Field)<br>CDR (Call Detailed Record)<br>Conference Room<br>DID (Direct Inward Dialing Number)<br>DISA (Direct Inward System Access)<br>DNIS (Dialed Number Identification Service)<br>SRTP (Secure Realtime Transport Protocol)<br>DND (Do Not Disturb)<br>FOP (Flash Operation Panel) Status Monitoring |

| | |
|---|---|
| | IVR (Interactive Voice Responses) |
| | Follow Me, Call Spy and PIN Set |
| | Distinctive Ringtone |
| | Multi-language System Prompt |
| | Multiple Language of GUI |
| | Phone Book, Speed Dial |
| | LDAP Server for phonebook |
| | Record Files Download |
| | Ring Group, SIP Trunk |
| | Skype for SIP, Smart DID, System Log |
| | T.38 fax (pass-through), voicemail and voicemail to e-mail |
| | Time-based Rule |
| | PBX log, web access log and PBX debug log |
| Call Features | Call Back, Call Forward, Call Group |
| | Call Hold, Call Paging and Intercom |
| | Call Park, Call Pickup, Call Queue |
| | Call Record, Call Route, Blind Transfer |
| | Attend Transfer, Call Waiting |
| | Caller ID, Dial by Name |
| | Customized IVR, On-hold Music, Transfer |
| | Three-way Conferencing, Video Call |

## System Capacity

| | |
|---|---|
| System Capacity | 60 concurrent call legs |
| | Up to 200 IP phone registers/extensions |
| | Recording and Voicemail: 150 hours |

## Network and Configuration

| | |
|---|---|
| Access Mode | Static IP, DHCP |
| LED Indications | SYS: 1, LNK/Off |
| | WAN: 1, LNK/Off |
| | LAN: 1, LNK/Off |
| | PWR: 1, LNK/Off |
| | SLOT: FXO/GSM (Red), FXS (Green) |
| Dimensions (W x D x H) | 343 x 154 x 35 mm |
| Operating Environment | -10~45 degrees C |
| | 10~80% humidity |
| Power Requirements | AC 100V-240V, DC 12V/2A, 50/60Hz |
| EMC/EMI | CE, FCC Class B, RoHS |

| Product | IPX-2500<br>Internet Telephony PBX system (500 SIP Users registrations) |
|---|---|
| **Hardware** | |
| 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<br>Store data for external disk |
| Audio In/Out | For external paging |
| 2 Slots | Supports maximum 8 ports (FXO/GSM) |
| **Protocols and Standard** | |
| Standard | SIP 2.0 (RFC 3261), IAX2 |
| Protocols | RFC 793 TCP<br>RFC 826 ARP<br>RFC 1034, 1035 DNS<br>RFC 1631 NAT<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<br>TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE |
| Voice Codec | G.711-Ulaw, G.711-Alaw, G.722, G.726, G.729, GSM, Speex and Opus |

| Video Codec | H.261, H.263, H.263+, H.264 and VP8 |
|---|---|
| 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>RFC 4733, SIP info, in-band and Auto |

**Internet Sharing**

| Network Features | DDNS client (Planet DDNS and easy DDNS)<br>DHCP server/SNMP v1/v2<br>IEEE 802.1Q of VLAN<br>IP assignment (DHCP/Static)<br>IPv4, IPv6<br>Manual configuration of static route table<br>Troubleshooting (ping, traceroute)<br>VPN server ,VPN client |
|---|---|
| Security Features | Refuse SIP Register DoS<br>Refuse Abort Invite Dos<br>Refuse SSH Login DoS<br>Firewall and enhances HTTPS connection |

**Features**

| PBX Features | SIP Register with UDP/TCP/TLS<br>Phone Auto-Provision<br>One Touch Recording<br>Mobility Extension<br>Black List<br>BLF (Busy Lamp Field)<br>CDR (Call Detailed Record)<br>Conference Room<br>DID (Direct Inward Dialing Number)<br>DISA (Direct Inward System Access)<br>DNIS (Dialed Number Identification Service)<br>SRTP (Secure Realtime Transport Protocol)<br>DND (Do Not Disturb)<br>FOP (Flash Operation Panel) Status Monitoring<br>IVR (Interactive Voice Responses)<br>Follow Me, Call Spy and PIN Set<br>Distinctive Ringtone<br>Multi-language System Prompt<br>Multiple Language of GUI<br>Phone Book, Speed Dial<br>LDAP Server for phonebook |
|---|---|

| | Record Files Download |
|---|---|
| | Ring Group, SIP Trunk |
| | Skype for SIP, Smart DID, System Log |
| | T.38 fax (pass-through), voicemail and voicemail to e-mail |
| | Time-based Rule |
| | PBX log, web access log and PBX debug log |
| Call Features | Call Back, Call Forward, Call Group |
| | Call Hold, Call Paging and Intercom |
| | Call Park, Call Pickup, Call Queue |
| | Call Record, Call Route, Blind Transfer |
| | Attend Transfer, Call Waiting |
| | Caller ID, Dial by Name |
| | Customized IVR, On-hold Music, Transfer |
| | Three-way Conferencing, Video Call |

## System Capacity

| System Capacity | 100 concurrent call legs |
|---|---|
| | Up to 500 IP phone registers/extensions |
| | Recording and Voicemail: 7500 hours |

## Network and Configuration

| Access Mode | Static IP, DHCP |
|---|---|
| LED Indications | SYS: 1, LNK/Off |
| | WAN: 1, LNK/Off |
| | LAN: 1, LNK/Off |
| | PWR: 1, LNK/Off |
| | SLOT: FXO/GSM (Red), FXS (Green) |
| Dimensions (W x D x H) | 343 x 154 x 35 mm |
| Operating Environment | -10~45 degrees C |
| | 10~80% humidity |
| Power Requirements | AC 100V-240V, DC 12V/2A, 50/60Hz |
| EMC/EMI | CE, FCC Class B, RoHS |

# Installation Procedure

## New System Setup Wizard

**Step 1.** Connect a PC directly with the IP PBX on WAN or LAN port. Default IP of WAN port is **172.16.0.1**, and default IP of LAN port is **192.168.0.1**. Please change the PC IP address to 172.16.0.xx if you have connected WAN or 192.168.1.xx if you have connected LAN.

**Step 2.** Start a web browser. The browser should be IE11, Firefox or Google Chrome.

**Step 3.** Enter the default URL of WAN port **"https://172.16.0.1"** or enter URL of LAN port "https://192.168.0.1" into the browser address bar to access its Web interface.

**Step 4.** Now you should be made known by a security warning. Please ignore the warning and proceed to access the IP PBX Web interface.

**Step 5.** By now you should have seen the **Quick Installation Wizard**. Please follow the wizard to go through a few steps of basic settings. Or you can skip the installation wizard.

● **Step 1: Wizard Language**
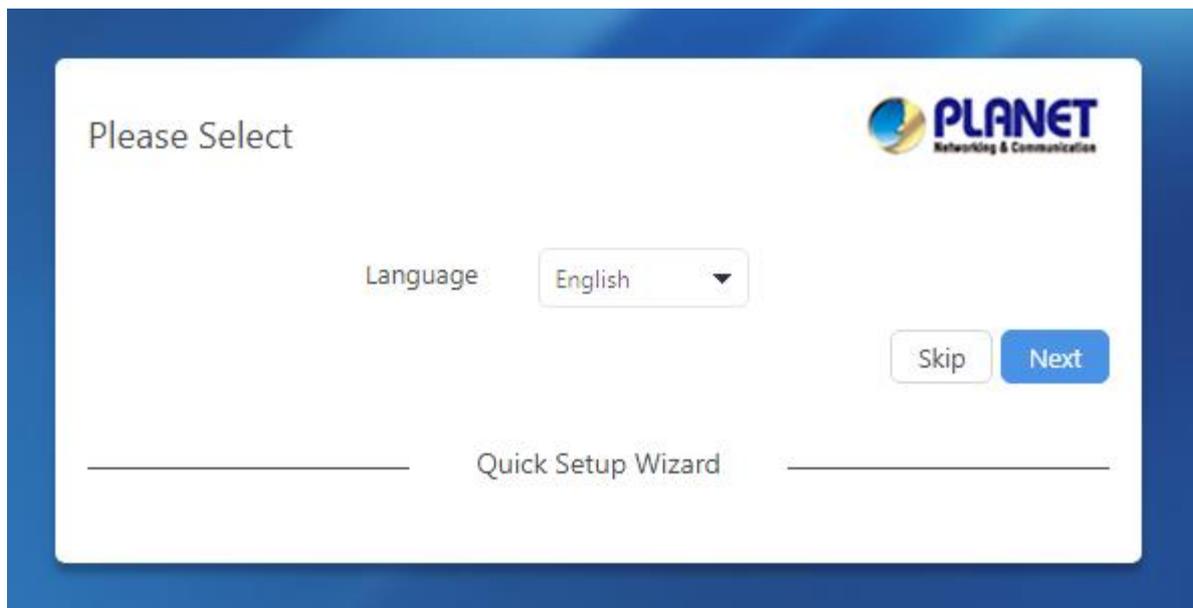


You may skip the wizard here by clicking on [ Skip ] button. Or choose your native language to begin with the wizard, if it's not in the list then choose one that you familiar with. Once done, please click on [ Next ] button to continue.

● **Step 2: Network Settings**
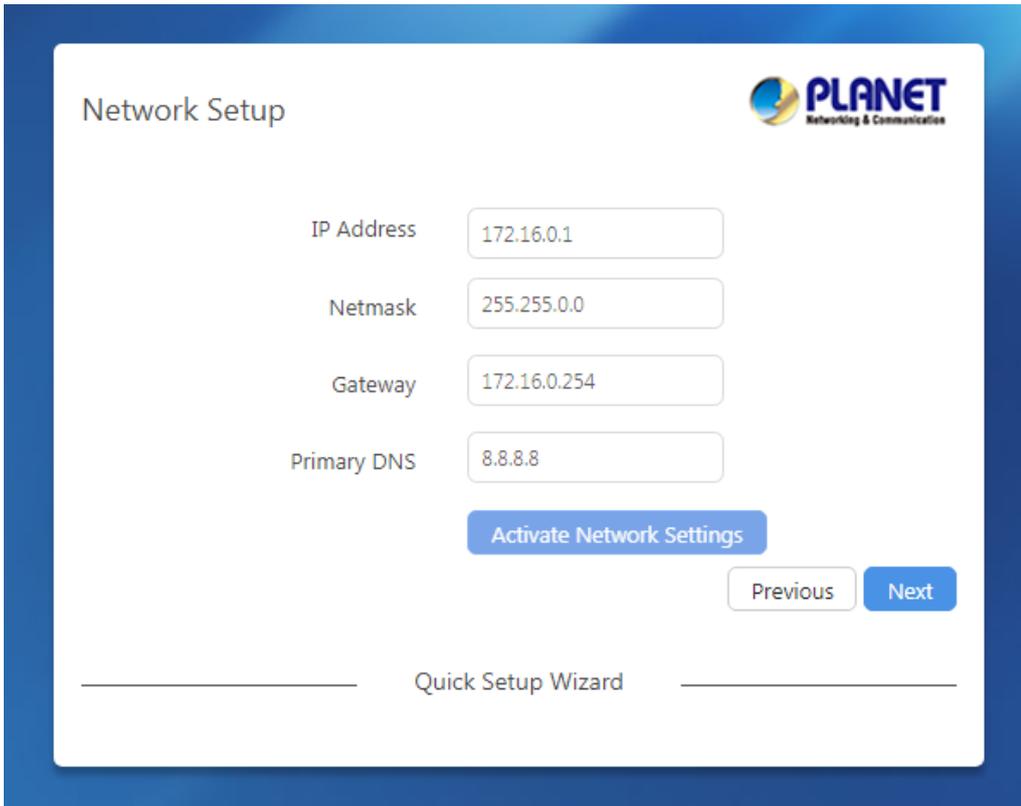


Change the network profiles per your local WAN environment. Once done, click on the [Activate Network Settings] button, it will take several seconds and you will be directed to the new page with new IP address automatically. Once done, please click on [Next] button to continue.

● **Step 3: Location**
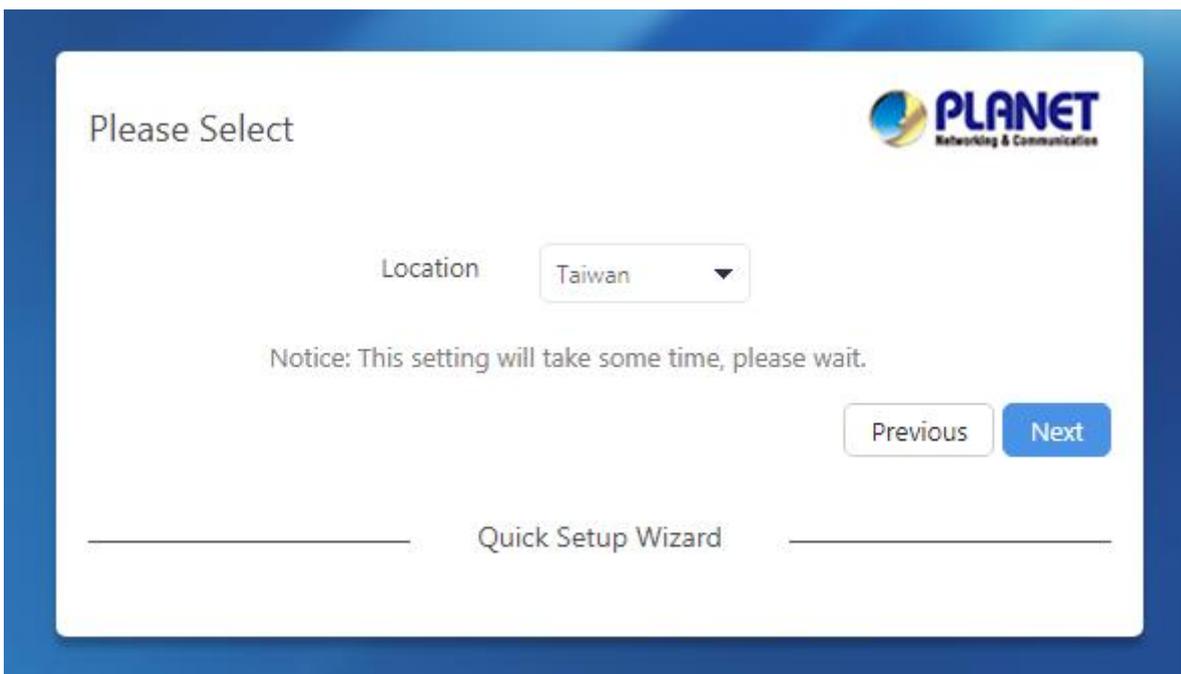
Select the country/region you live in, the location you selected will also tell the IPPBX system which time zone and tone zone you want the IPPBX to use. And some other regional settings will be defined as well.

● **Step 4: Module Settings**
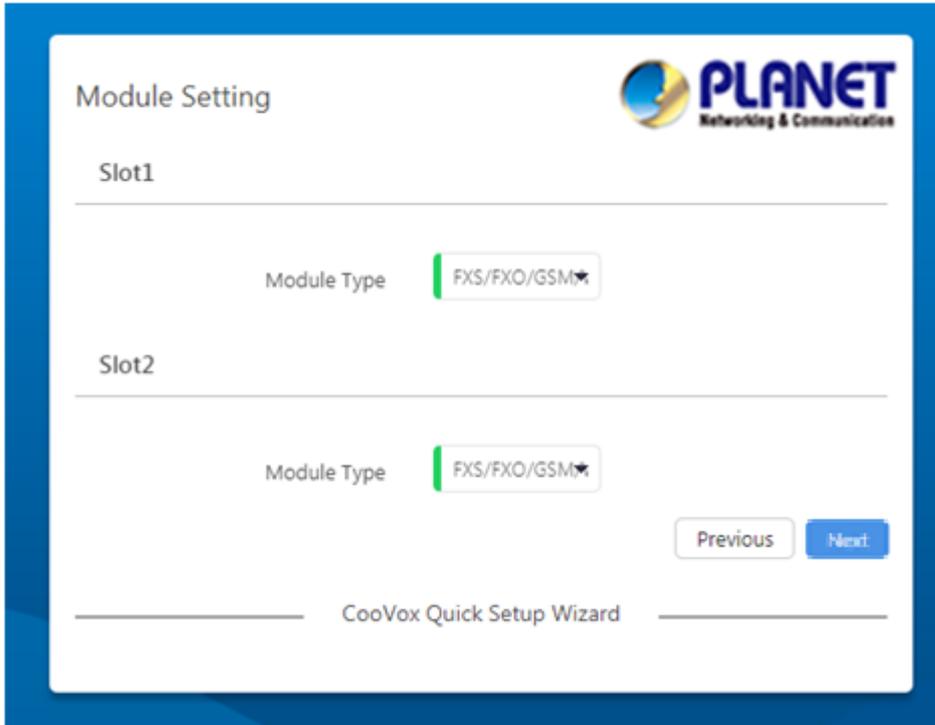


This step is only for IPX-2200 and IPX-2500. If you have installed FXS/FXO/GSM/WCDMA modules on both Slots then just simply click on [Next] button to continue. If you have installed E1/T1/BRI modules, please select the module type accordingly. For more configurations please refer to Module Settings.

● **Step 5: Create Departments and Extensions**
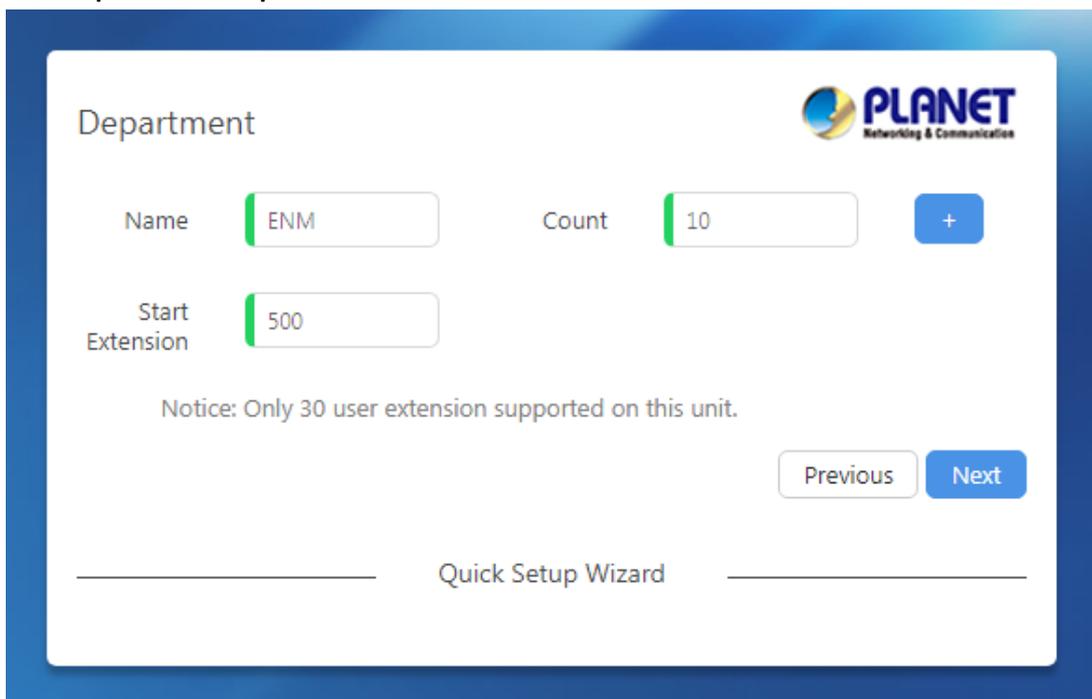
According to your company organization structure you can add departments and extensions here per each department.

Specify the department name and number of the members, and then click on [ + ] button to add another department the same way. Once done, specify a start extension number and click on [ Next ] button to continue.

● **Step 6: Specify Operator Extension**
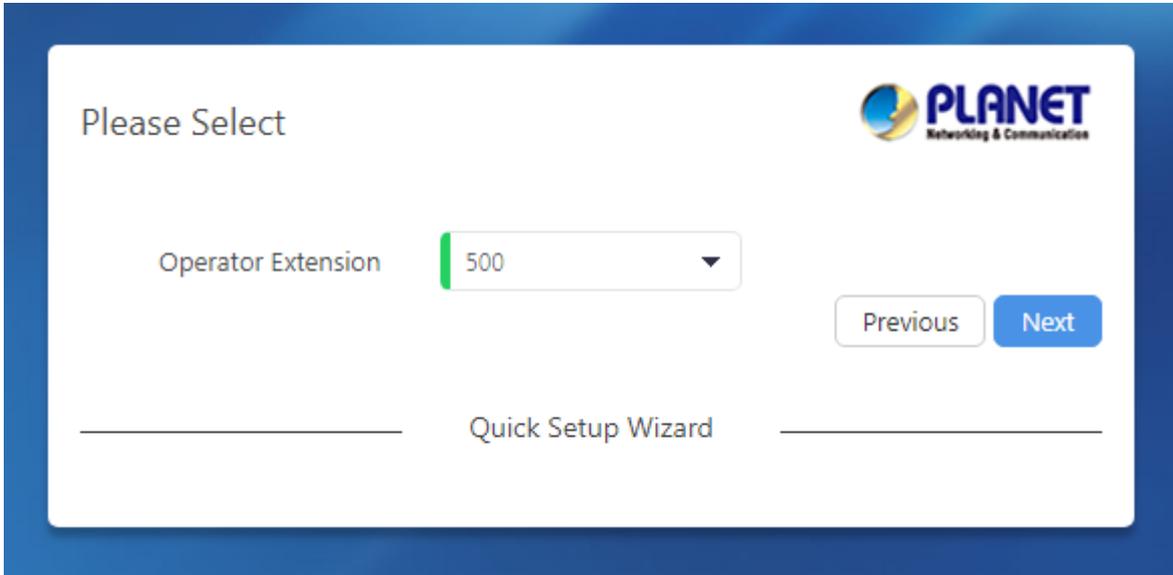


The operator extension is usually the extension number of the company receptionist or secretary. Please specify the extension number here, in certain circumstances calls will be sent directly to this extension for information.

● **Step 7: Create Admin Account**

Admin account can be customized that you can use any user name you prefer. For security reasons, please do not use weak administrator credentials.

● **Step 8: Create Operator Account**
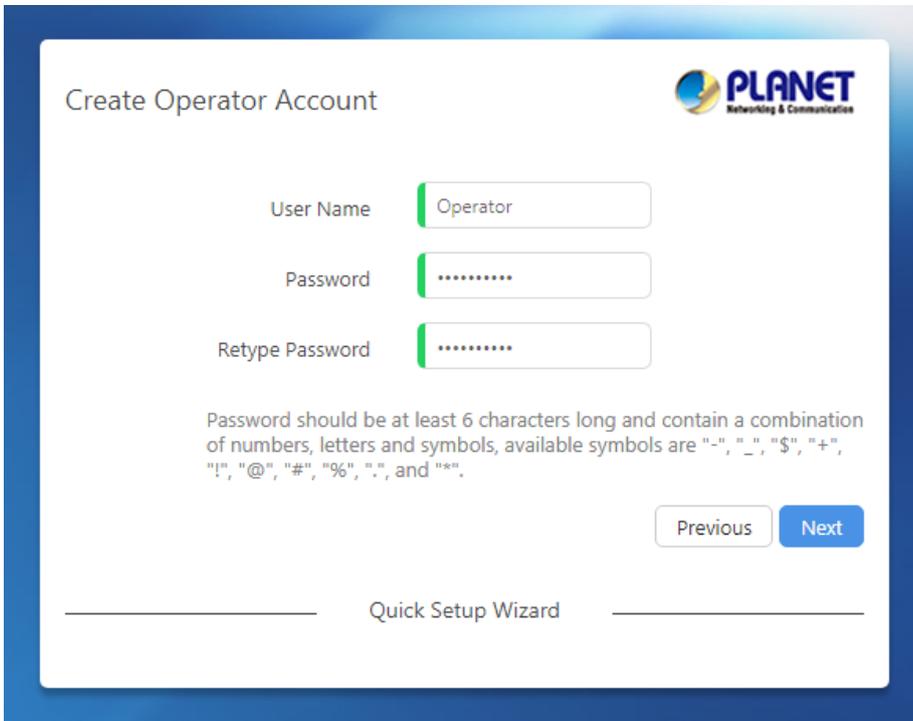


Operator account has limited permissions compare to Admin user. It can be used by the operator user to manage extension numbers of the company, setup calling features, manage call logs and recordings.

"Enable AMI" option can be used to enable CTI (Computer Telephony Integration) connectivity using the operator account. Please keep it disabled if there's no such requirement.

● **Step 9: Mail Server Settings**
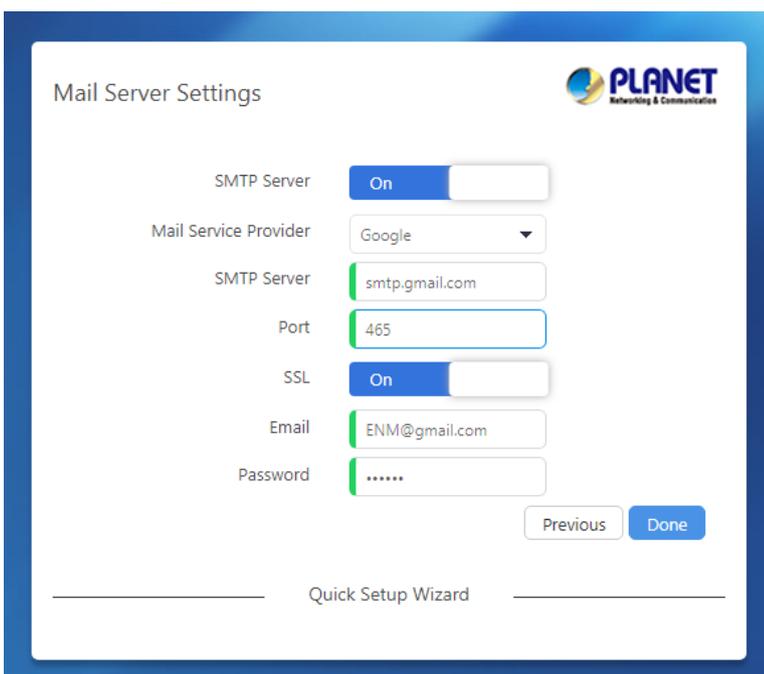
Mail server will be used to send out email notifications from the IPPBX system. Please select a mail service provider from the list. Supported mail service providers' default settings will be filled in automatically. If the mail service provider of yours is not in the list please choose "Other".

Once done, please click on [Done] button, now system will reboot for new settings take effect. If IPX-330 or IPX-2100, you'll have to wait around 4 minutes then refresh the page. If IPX-2200 or IPX-2500, it will take around 2 minutes to reboot, after this please refresh the page. Now you should see the login page like the screenshot shown below.



Please use the admin credentials you defined during the installation wizard to sign in.

The admin user name and password are defined by during the installation wizard process. If you have skipped the wizard, you'll need to log in with the default username and password which are both "**admin**". Once you have modified the network profile during the quick installation wizard or after admin login, then you can power off and deploy the IPPBX system to your local LAN, and then power it on for further configurations.

**(Default IP) For increased security, please use https to access IP PBX.**

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

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

# Extensions

Path: *Telephony -> Extensions*

Extensions and departments should have been created during the Quick Setup Wizard process. You may manage extensions and departments here on this screen. If you have skipped the Quick Setup Wizard, you may create them here on this screen as well.

## Departments

Path: *Telephony -> Extensions -> Departments*

Department concept is new on IP PBX series v3.0 version. Extensions are grouped by your company's actual organizational structure.

A department is equal to a Pick-up group.

A department is equal to a Call groups (Ring Group).

If you have created departments and extensions from Quick Setup Wizard, you should see all your departments and extensions here.



If you wish to create a new department, please click on the **Add** button. Specify the department name and select member extensions and then submit.

If you wish to modify department settings, please click on the [pencil] button, or click on the [trash] button to remove a department.

- You may change the department name from the **Department Name** textbox.
- In the Ring Strategy dropdown list select a desired ring strategy of how to ring the department (Ring Group) extensions upon incoming calls.
  - **Ring All**: Ring all available member extensions until one answers (default).
  - **Leastrecent**: Ring the extension of the member who has least recently received a call.
  - **Fewestcalls**: Ring the extension of the member who has taken the fewest number of calls.
  - **Random**: Ring the extension of a random member.
  - **RRMemory**: The system remembers which member was last called and begins the round robin with the next agent.
  - **Linear**: Starting with the first member, ring the extension of each member in turn until the call is answered.
  - **RROrdered**: Same as RRMemory, except the department member order is preserved.
- In **Destination if no answer** dropdown list select a call destination for the inbound calls when no one answers the call.
- You may add/remove members of your department from the **Select Department Members** field.

# IP Extensions

Path: *Telephony -> Extensions -> IP Extensions*

IP extensions are user extensions that can be registered on various SIP/IAX2 endpoints, including desktop IP phones, softphone for Windows/Android phones/iPhones/Linux and some other endpoints that support SIP/IAX2 protocol.



The extensions are created through the quick setup wizard, to check or modify the extension properties please click the  button.

- **User Name**: Alias of this extension which can be the name of the extension user.
- **Password**: The password is used for the phones to register or by the extension user to sign in the extension web portal. This can be set manually or can be generated by the IPPBX system. Auto generated password consists of numbers, letters and special characters.
- **Email**: Email address of this extension user.
- **Outbound CID**: Choose a number to show the external called party. This feature only works with E1/T1, BRI and SIP trunks if the telco/ITSP (Internet Telephony Service Provider) allows this number to be passed.
- **Music On Hold**: When the user holds a call the caller will listen to music, and the music can be set up here.
- **Mobile Number**: The mobile phone number of the extension user.
- **Dial Permission**: Defines which type of numbers the extension can dial.
- **Language**: If the extension user is not a native speaker of the system default language, you may set a different language for this user.



Voicemail: Voicemail box could be enabled or disabled for this user.
- **Voicemail Password**: The password used to access voicemail by voicemail feature codes.
- **Remote Extension**: If you want the extension can be used out of the LAN this option needs to be enabled.

Before doing this please ensure the extension uses a strong password.

- **Simultaneous Register Count**: The extensions could be registered on up to 5 different SIP endpoints at the same time, by default the value is 2. When there are already 2 registers the 3$^{rd}$ register will be responded with a 403 error.
- **Video Call**: You may enable/disable video call support of this extension.
- **Video Codecs**: Supported video codecs are H.261, H.263, H.263+, H.264, VP8.
- **Web Portal**: If enabled, users can use their extension number and password to login to the IPPBX system web GUI.
- **Call Recording**: This is an automated recording option, you may choose to automatically record the inbound, outbound or both inbound and outbound calls from/to this extension.
- **Call Spy**: Call Spy feature allows the phone calls of this extension to be monitored from other extensions, please refer to Call Spy Feature Codes for how to monitor phone calls.
- **Register Expiration**: Default register expiration time, default value 120 seconds.
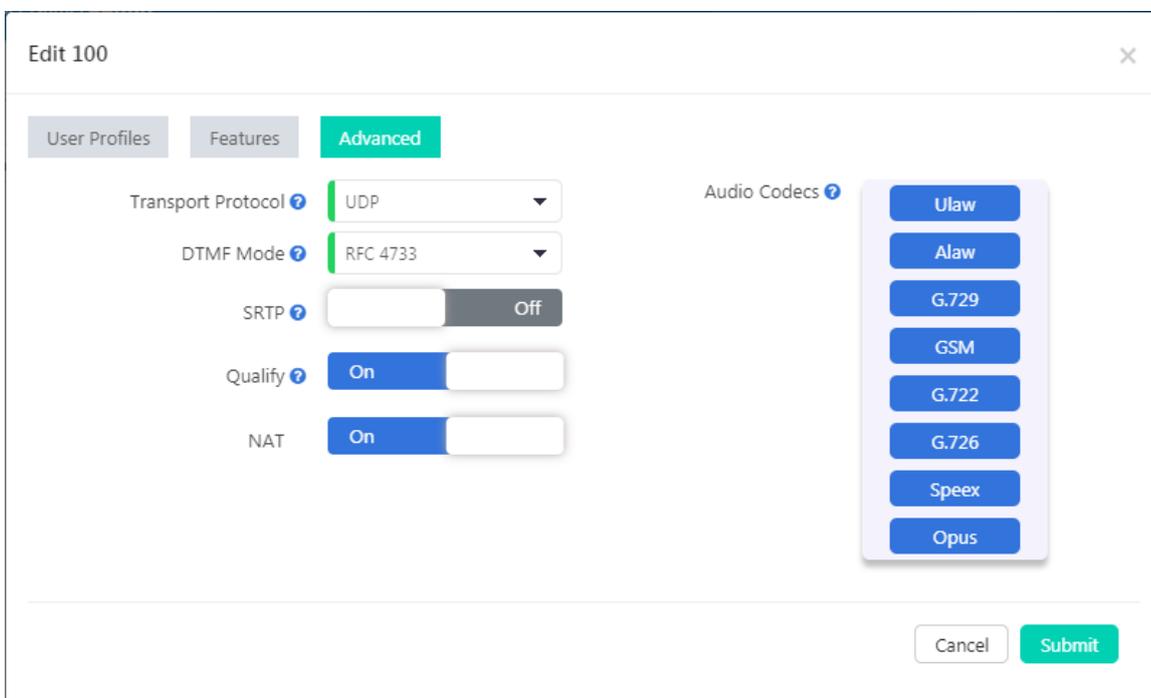


- **Transport Protocol**: The transport protocol to be used by SIP signaling. By default it uses UDP protocol, if you choose to use TCP or TLS please make sure the SIP IP phone or softphone uses the same protocol. Otherwise you'll get "403" error on SIP register.
- **DTMF Mode**: Defines how the system detects DTMF tones, the default setting is RFC4733, it can be changed if necessary.
- **SRTP**: Secure Real-time Transport Protocol (SRTP) encrypts the RTP traffic to secure your VoIP phone calls. Before enabling this option you need to ensure the SIP endpoint can also support SRTP.
- **Qualify**: The IPPBX system sends a SIP OPTIONS command regularly to check that the device is still online.
- **NAT**: Enable this option if extension user or the phone is behind a router.
- **Audio Codecs**: PLANET IPPBX system supported audio codecs such as G.711 (ulaw, alaw), G.722, G.726, G.729, GSM, Opus and Speex. The sequence of the audio codecs listed here is the sequence of the audio codecs to be used for negotiating the media of a phone call to be established.

If you want more extensions to be created or if you had skipped the quick setup wizard so no extension had been created, you may click on Add button to add a new extension or click on Bulk Add button to create bulk extensions.

The extensions' properties could be set while you are creating them.
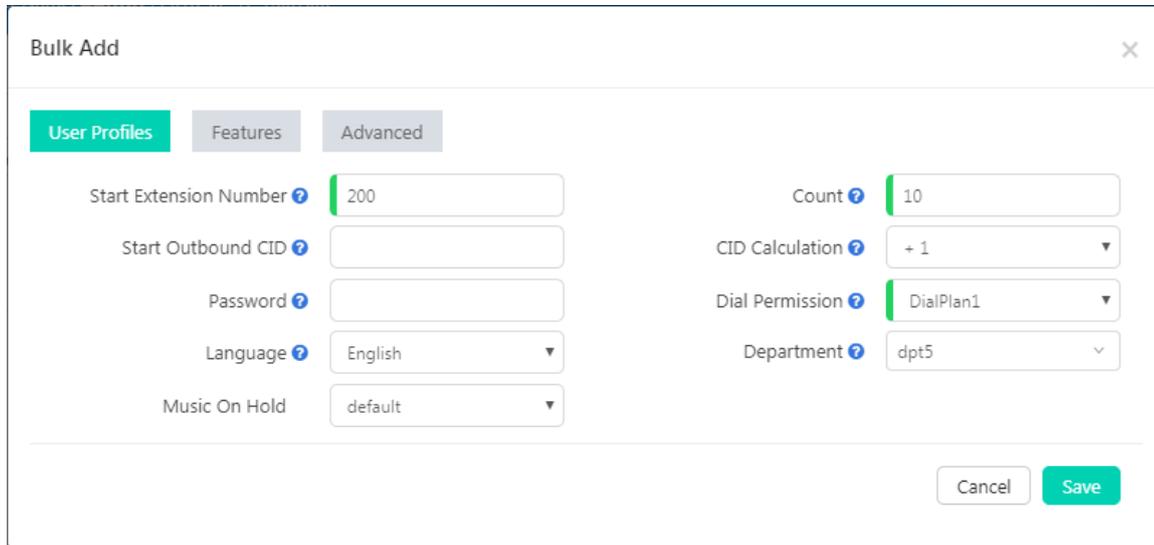


- Define a **Start Extension Number** and the number of extensions to be created in the **Count** field.
- If you want to associate outbound CID numbers to the extensions, you'll need to specify the first CID number in the **Start Outbound CID** field and in the **CID Calculation** field specify the calculating of the following CID numbers. Otherwise leave these two blanks.
- In the **Password** field you may leave it blank so the created extensions will use random passwords or you can define a password so the created extensions will share the same password.

As for other options, you may configure accordingly per your demands. The features/options configured will be applied to all newly created extensions.

# Analog Extensions

Path: *Telephony -> Extensions -> Analog Extensions*

Analog extensions are generated automatically by the IPPBX system if FXS interfaces are detected. All you have to do is attaching analog phones or fax machine to the FXS interface, the analog extensions can be used directly for phone calls, no more additional settings required.

| | Departments | | IP Extensions | | Analog Extensions | | Phone Provisioning | | Expansion Box |
|---|---|---|---|---|---|---|---|---|---|
| Channel ❓ | Extension ❓ | | Name | Department Name | Call Recordings | Dial Permission | Options | | |
| 5 | 005 | | 005 | | Disabled | DialPlan1 | ✏️ | | |
| 6 | 006 | | 006 | | Disabled | DialPlan1 | ✏️ | | |
| 7 | 007 | | 007 | | Disabled | DialPlan1 | ✏️ | | |
| 8 | 008 | | 008 | | Disabled | DialPlan1 | ✏️ | | |

4 Total

The **Channel** column and **Extension** column list the FXS interfaces and the corresponding extension numbers. They are generated by the IPPBX system which cannot be modified.

You may click on the ✏️ button to change settings if necessary.

Edit 005                                    ✕

| | |
|---|---|
| Alias | 005 |
| Department | dpt5 ⌄ |
| Call Recording ❓ | Disabled ▾ |
| Language ❓ | English ▾ |
| Dial Permission ❓ | DialPlan1 ▾ |
| Input Volume ❓ | -10 ——●—— 0 —— 10 |
| Output Volume ❓ | -10 ——●—— 0 —— 10 |

Cancel    Submit

- **Alias** can be defined to identify this analog extension.
- **Department** can be selected to add this analog extension to one of the departments. Or could change a department it belongs to.
- **Call Recording** could be enabled to record Inbound, Outbound or Both direction phone calls if necessary.
- The **Language** option determines the language type of the system prompt that the user will listen to/hear
- **Dial Permission** controls which dial rules the user can use to make phone calls.
- **Input Volume** could be used to adjust the input gain of this analog extension.
- **Output Volume** could be used to adjust the output gain of this analog extension.

# Inbound Control

Path: *Telephony -> Inbound Control*

The Inbound Control section is where you define how IPPBX 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.

## Inbound Destinations

A call destination in IPPBX system might be an IVR menu that instructs the callers to press certain digits to route their calls, a queue to wait for specific telephone services, a ring group to call a number of user extensions, or virtually any other type of process to route the call in whatever way is desired. A call may have several destinations throughout its lifespan.

Below is a list of call destinations available in IPPBX system:

- **IVR**: Strictly speaking, IVR is not an actual call destination. The calls landed on IVR menu will finally be directed to other call destinations even another IVR menu and this is how you create multi-layer IVRs.
- **Extension**: to ring a specific extension.
- **Voicemail**: direct the caller to the voicemail box of a specific extension.
- **Trunk**: direct the inbound call to a selected trunk and dial a specified number, it's mostly used when the IPPBX system is applied as a gateway.
- **Department (Ring Group)**: to ring the extensions of a department at this same time, in other words, department equals to a ring group. If you want to exclude some extensions from the ring group, please disable "Ring Group Member" option from the department edit screen.
- **Conference**: to send the caller to a specific conference room.
- **Queue**: to send the caller to a specific call queue.
- **Paging Group**: to send the caller to a paging group, so the caller could make a public address to the paging group members externally.
- **Hangup**: to hangup the call directly.
- **Fax**: to handover the call to IPPBX system and handled by Inbound Fax settings.

## IVR

Path: *Telephony -> Inbound Control -> IVR*

IVR, or Interactive Voice Response, is responsible for the menus people hear and respond to when they call up a company or business and hear the words for example: "press 1 for sales, press 2 for marketing, press 0 to speak to the operator,".

Before configuring IVR menus you will first need to create inbound call destinations, for example, **Extensions**, **Departments** (ring groups), **IVR prompts**, **Call Queues**, etc.

If you want to create multi-layer IVR menus, you may need to create the sub-layers at first.

In order to create an IVR menu, please click on the [Add] button, you'll see a popup dialog as below:



- In the **Name** blank a name is required to identify this IVR menu.
- In the **Number** blank a number had been created for this IVR menu for user being able to dial this number and test the IVR options.
- In **Voice Prompts** drop-down list, select a pre-recorded voice prompts for this IVR menu. The prompts will be played to the callers as they enter the IVR. The voice prompts must be uploaded from the *Audio Library -> IVR Prompts* screen.
- In **Loop Count** drop-down list, select the number of times to playback this IVR prompts before callers pressing a key.
- **Dial Extension** switch could be enabled for callers to dial specific numbers upon this IVR menu if they already knew which number should be dialed, so the callers don't have to listen to all the options of this IVR.
- If **Dial Extension** is enabled a default **Dial Permission** named **Extension** will be applied for callers being able to dial internal extensions upon this IVR menu, if you wish callers could dial some more numbers you may select another dial permission here. (Not Recommended)
- **Events** are the IVR options to be configured according to the instructions you have specified in the selected IVR prompts. Available key presses could be set from **0** to **9**, **\*** and **#**. If the caller presses the key which are not specified and it will be handled by the "Invalid Key" option.

# Time Conditions

Path: *Telephony -> Inbound Control -> Time Conditions*

Time conditions in IPPBX allow you to control what happens to inbound calls both during and outside (weekends/holidays) normal business hours.

Time condition settings include Time Rule, Weekday and Holiday settings.

- Time Rule:
- Weekdays:
- Holidays:

To create a time rule you need first set up weekdays and holidays.

To set up weekdays you may modify the default one or create a new one by clicking on **Add** button.



This example shows the company opens from Monday to Friday. On each weekday, it opens from 9 am to 12 pm, after a 2-hour break then opens from 2 pm to 6 pm. Any other time durations unspecified will be considered as non-business hours.

In order to exclude holidays from the weekdays, you'll also have to set up holidays.



Please ensure you add all upcoming holidays to the holiday list.

Now you have all prerequisites to set up a time rule.

Now you could apply this time rule to the **Inbound Routes**.
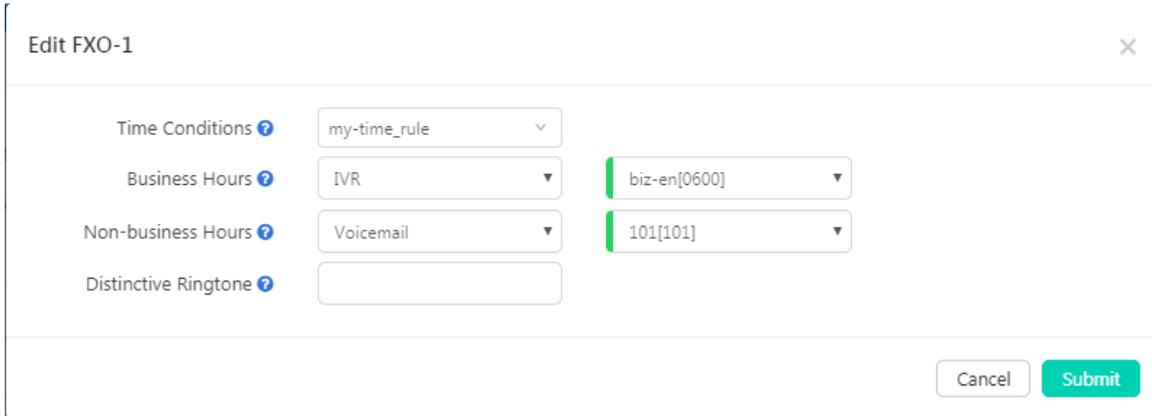
# Inbound Routes

Path: ***Telephony -> Inbound Control -> Inbound Routes***

The Inbound Routes settings tell your IPPBX system where to send those inbound calls coming in from the trunks. Calls can be sent to a variety of destinations, including extensions, departments (ring groups), call queues, IVRs, DISAs, conferences, paging groups, voicemail, fax, etc.

The Inbound Routes are configured per each trunk. You may set different inbound destinations for different trunks.



Please click on  button to configure inbound routes for each trunk.

In the **Time Conditions** drop-down list, select a time rule for this trunk, so the inbound calls will be routed on time basis.

In the **Business Hours** field you should select a calling destination for business hours, and in the **Non-business Hours** field select a calling destination for non-business hours.

**Distinctive Ringtone** is optional, if needed, you may specify the ringtone name of the phone, so when the callers call in from this trunk the phone will ring this specific ringtone. It requires the phone supporting distinctive ringtone feature.

This is how you configure inbound routes for a trunk, you may configure the same inbound routes for other trunks or use different inbound route settings per your requirements.

# Direct Routing

Path: *Telephony -> Inbound Control -> Direct Routing*

You may set up Direct Routing based on the DID numbers of your VoIP/E1/T1/BRI trunk lines and the phone numbers of the external callers. Direct Routing has higher priority than time conditions and other general inbound routes.

Direct Routing based on DID numbers will cause the inbound calls which dialed the specified DID number to a specific call destination without the limitation of any other inbound settings.

To add a Direct Routing rule based on DID number, please click on the "Add" button as shown below.
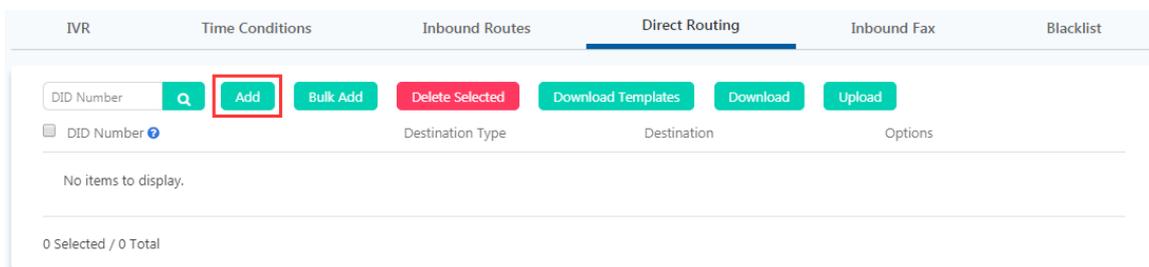


In the popup window, specify one of your DID numbers, and assign a call destination for all inbound calls by calling this DID number.
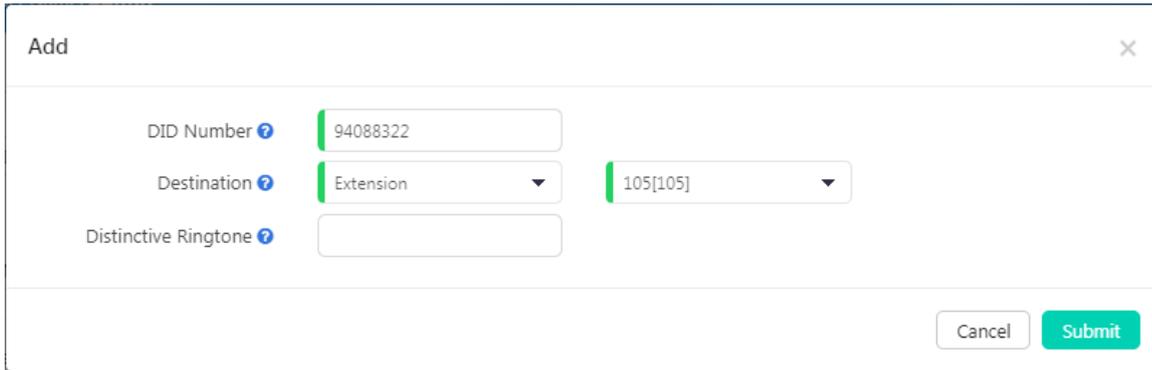
In the above example, 94088322 is one of your DID numbers, you may configure it with an extension number, when someone calls this number the call will then directly go to the selected extension.

In the **Distinctive Ringtone** field you may specify the ringtone name of the phone, so when the callers call the DID number and the call goes to this extension the phone will ring this specific ringtone. It requires the phone supporting distinctive ringtone feature.

To add a Direct Routing rule base on the caller's number, please click "Add" button as shown below.
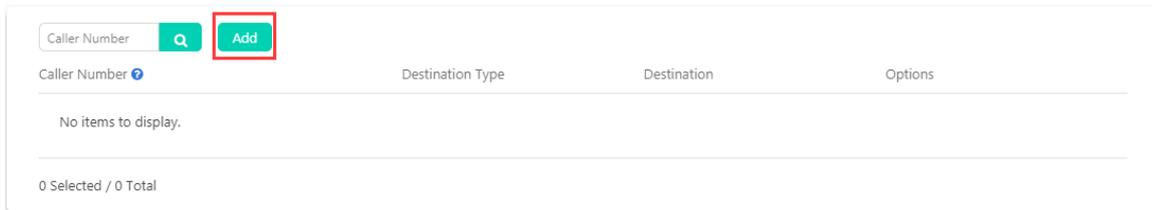


In the popup window, specify the caller's number, and assign a call destination for inbound calls from this external phone number.



Once this Direct Routing is created, all phone calls coming from the number 69032354 will then all go to extension 106, no matter when and from which trunk the call is coming in.

# Inbound Fax

Virtual Fax feature on IPPBX system has the ability to automatically detect incoming fax and send the fax to one of the destinations as below:

- **Save to System**
- **Send as Email**
- **Fax Machine**

If the **Fax Destination** is set to **Save in System**.
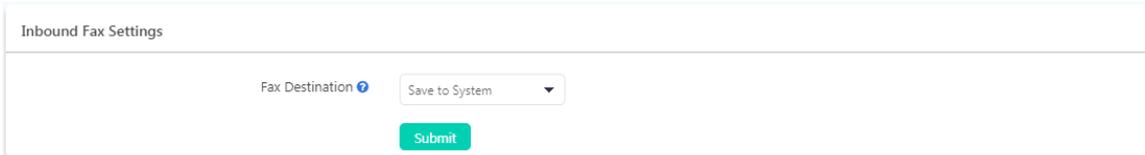
| Inbound Fax Settings | |
|---|---|
| Fax Destination ❓ | Save to System ▾ |
| | Submit |

The received faxes will be saved in the IPPBX system internal storage. Admin user and operator user are able to check faxes on **Reports -> System Logging -> Fax Logs** page.

If the **Fax Destination** is set to **Send as Email**.
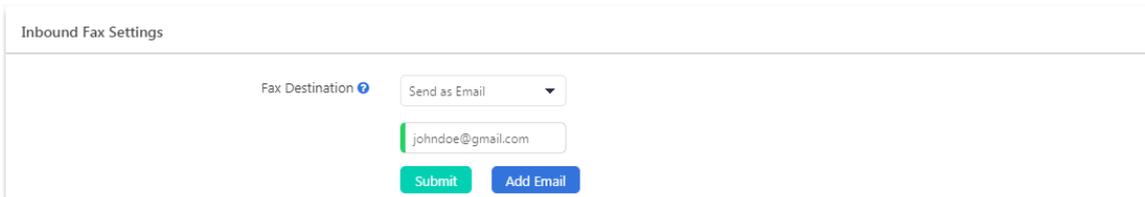
| Inbound Fax Settings | |
|---|---|
| Fax Destination ❓ | Send as Email ▾ |
| | johndoe@gmail.com |
| | Submit   Add Email |

You'll have to give an Email address for the IPPBX to send the receive faxes to this Email address. Each fax will be attached to the Email in .tif format.

If your IPPBX system has been equipped with FXS ports, you may set the **Fax Destination** to **Fax Machine**.
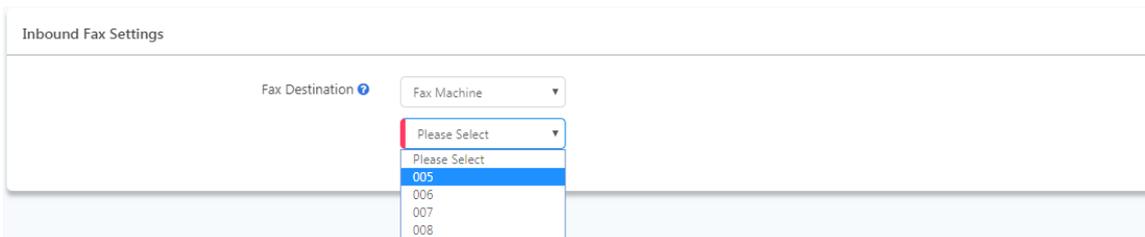
| Inbound Fax Settings | |
|---|---|
| Fax Destination ❓ | Fax Machine ▾ |
| | Please Select ▾ |
| | Please Select |
| | 005 |
| | 006 |
| | 007 |
| | 008 |

In the dropdown list, select the analog (FXS port) extension number which has the fax machine attached. When inbound fax detected, it will be directed to the fax machine.

# Blacklist

Path: **Telephony -> Inbound Control -> Blacklist**

Blacklist feature allows you to create a list of numbers that are not allowed to call in to the IPPBX system. Blacklist could be managed by both the admin user and operator user. The extension user could also add numbers to the system blacklist by using Blacklist Feature Codes.



By specifying a number in the top right number blank, you may add a number to the system blacklist.

To add blacklist numbers by using a template file, please click on **Download Templates** button to download the template file and edit with MS Excel, once done upload it from this page, the numbers you added in the template file will be added to the system blacklist.

If you want to share the blacklist numbers on other IPPBX systems, you may download it by clicking the **Download** button to download all blacklist numbers in a file and upload on other IPPBX systems.

# Outbound Control

By default if you've not configured any outbound control settings, the extension users are not able to make outbound phone calls yet. Please follow the instructions of this chapter to configure the IPPBX system for outbound phone calls.

## Trunks

A trunk on an IPPBX system is essential for extensions to be able to make outbound phone calls. On IPPBX system, the trunks will be detected and generated automatically at the first time of the system initialization.

### FXO/GSM/WCDMA Trunks (IPX-21FO/IPX-21GS/IPX-21MA)

Path: ***Telephony -> Outbound Control -> Trunks***
On the IPPBX front panel, red LED indicates the RJ11 interface is FXO. You should attach the telephone wire from your telecom socket to the FXO ports.
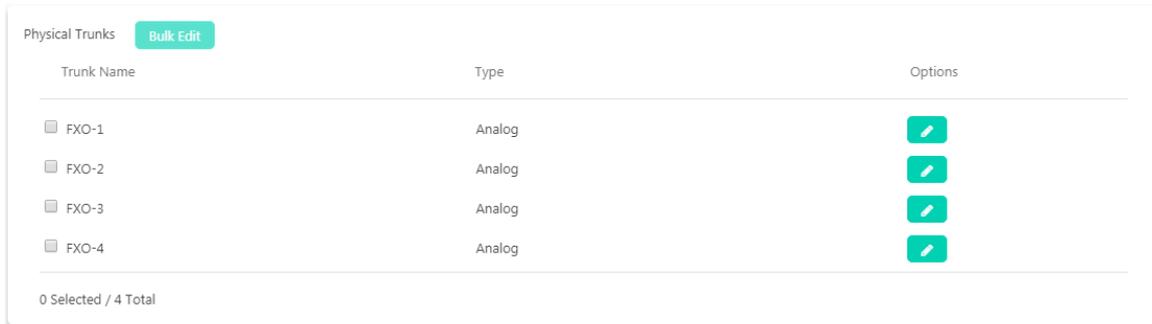
| Physical Trunks | Bulk Edit | | |
|---|---|---|---|
| Trunk Name | Type | | Options |
| ☐ FXO-1 | Analog | | ✎ |
| ☐ FXO-2 | Analog | | ✎ |
| ☐ FXO-3 | Analog | | ✎ |
| ☐ FXO-4 | Analog | | ✎ |
| 0 Selected / 4 Total | | | |

If needed you may edit the trunk settings by click on the ✎ button, or you may select the same type of trunks and click on ⬚ Bulk Edit button to edit settings of the trunks together.
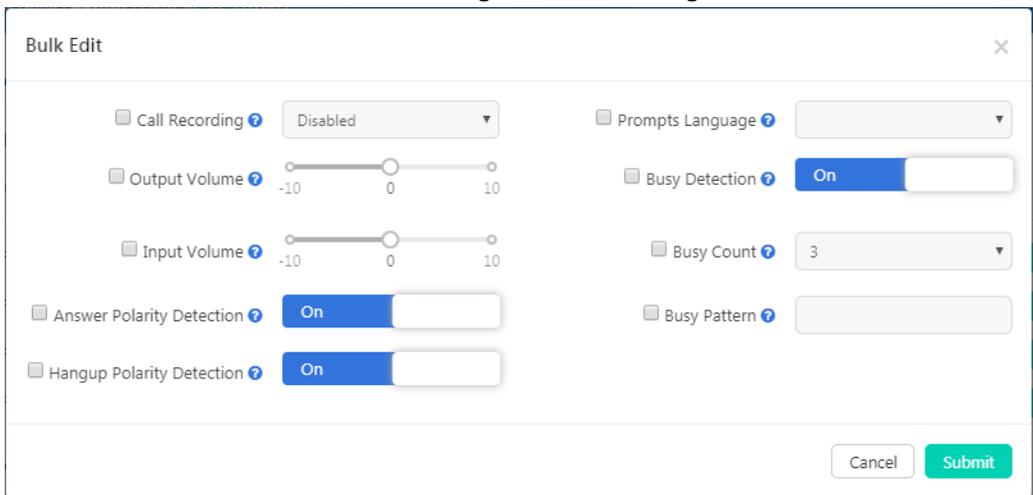
**Bulk Edit** ✕

| | | |
|---|---|---|
| ☐ Call Recording ❓ | Disabled ▼ | ☐ Prompts Language ❓ ▼ |
| ☐ Output Volume ❓ | -10   0   10 | ☐ Busy Detection ❓ On |
| ☐ Input Volume ❓ | -10   0   10 | ☐ Busy Count ❓ 3 ▼ |
| ☐ Answer Polarity Detection ❓ | On | ☐ Busy Pattern ❓ |
| ☐ Hangup Polarity Detection ❓ | On | |

Cancel   Submit

Select the parameters you wish to configure before modifying them. Usually if the trunks are working fine please do not change these settings.

● **Call recording**: To enable or disable call recording on the trunk/trunks. To enable recording you have options

to record inbound calls only, outbound calls only or both inbound and outbound calls.

- **Output Volume**: Sets the volume of the outgoing calls from the FXO channels.
- **Input Volume**: Sets the volume of the incoming calls from the FXO channels.
- **Answer Polarity Detection**: When enabled, FXO (FXS signaled) ports watch for a polarity reversal to mark when an outgoing call is answered by the remote party.
- **Hangup Polarity Detection**: In certain countries, a polarity reversal is used to signal the disconnection of a phone line. If enabled, the calls will be considered "hang up" on a polarity reversal.
- **Prompts Language**: You can choose a desired language of the system voice prompts to play to the incoming calls from this trunk. For example, if the call is not answered or the user is busy, the IPPBX system will notify the caller to leave a voice message in the language you set.
- **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, and it's configurable only if Busy Detection is enabled.
- **Busy Pattern**: If busy detection is enabled, it is also possible to specify the cadence of your busy signal.

## E1 Trunks (IPX-21PR)

Path: ***Telephony -> Outbound Control -> Trunks***

If your IPX-2200 or IPX-2500 IPPBX system has E1 module installed, you'll have an E1 trunk available for inbound and outbound phone calls.

| Physical Trunks **Bulk Edit** | | | |
|---|---|---|---|
| Trunk Name | Remark | Type | Options |
| ☐ FXO-32 | | Analog | ✏ |
| ☐ FXO-33 | | Analog | ✏ |
| ☐ FXO-34 | | Analog | ✏ |
| ☐ FXO-35 | | Analog | ✏ |
| ☐ PRI-1 | | Digital | ✏ |

0 Selected / 5 Total

Click on the ✏ button to configure the E1 (PRI) trunk when needed.

**Edit  PRI-1**                                                          ✕

| | | | |
|---|---|---|---|
| Remark | | | |
| Call Recording | Disabled | Overlap Dial | Yes |
| Outbound CID | | Reset Interval | 3600 |
| Pri Indication | Inband | Switch Type | EuroISDN (common in Eurc |
| Prompts Language | English | Fax Detect | On |
| Dial Permission | Default | Preferred Outbound CID ❓ | Extension |

Cancel    **Submit**

- **Call recording**: To enable or disable call recording on the trunk/trunks. To enable recording you have options to record inbound calls only, outbound calls only or both inbound and outbound calls.
- **Overlap Dial**: Overlap dialing mode (sending overlap digits).
- **Outbound CID**: The number you want to display to the called party.
- **Reset Interval**: To set the time in seconds between restart of unused B channels.
- **Pri Indication**: To enable this to report Busy and Congestion on a PRI using out-of-band notification.
- **Switch Type**: To set the type of PRI switch being used by the telephony provider.
- **Prompts Language**: Custom a system voice prompts language for the callers calling in from this trunk.
- **Fax Detect**: Enable/disable fax detection on this trunk.
- **Dial Permission**: Custom dial permission for this trunk, by default it uses the "Extension" dial permission. Configure only if this trunk is used for PBX integration, so calls coming from the other side can dial out from this IPPBX trunk directly. DO NOT change unless you fully understand how this feature works.
- **Preferred outbound CID**: To set preferred outbound CID of this trunk of the extensions.

## BRI Trunks (IPX-21BRv2)

Path: ***Telephony -> Outbound Control -> Trunks***

If your IPX-2200 or IPX-2500 IPPBX system has 4BRI module installed, you'll have 4 BRI trunks available for inbound and outbound phone calls.



Click on the [✏] button to configure the BRI trunk when needed.

- **Call recording**: To enable or disable call recording on the trunk/trunks. To enable recording you have options to record inbound calls only, outbound calls only or both inbound and outbound calls.
- **Overlap Dial**: Overlap dialing mode (sending overlap digits).
- **Outbound CID**: The number you want to display to the called party.
- **Reset Interval**: To set the time in seconds between restart of unused B channels.
- **Pri Indication**: To enable this to report Busy and Congestion on a BRI using out-of-band notification.
- **Switch Type**: To set the type of PRI switch being used by the telephony provider.
- **Prompts Language**: Custom a system voice prompts language for the callers calling in from this trunk.
- **Fax Detect**: To enable/disable fax detection on this trunk.
- **Dial Permission**: Custom dial permission for this trunk, by default it uses the "Extension" dial permission. Configure only if this trunk is used for PBX integration, so calls coming from the other side can dial out from this IPPBX trunk directly. DO NOT change unless you fully understand how this feature works.
- **Preferred outbound CID**: To set preferred outbound CID of this trunk of the extensions.

## VoIP Trunks

Path: ***Telephony -> Outbound Control -> Trunks***
Asterisk PBX can be registered as a SIP user agent to a SIP proxy (provider). If you have subscribed to a VoIP service from an ITSP (Internet Telephony Service Provider), then with the account details provided by them you can configure a VoIP trunk on your IPPBX system for the user extensions to share and make outbound phone calls.
To implement your SIP trunk account on the IPPBX system, you'll need to create a VoIP trunk.
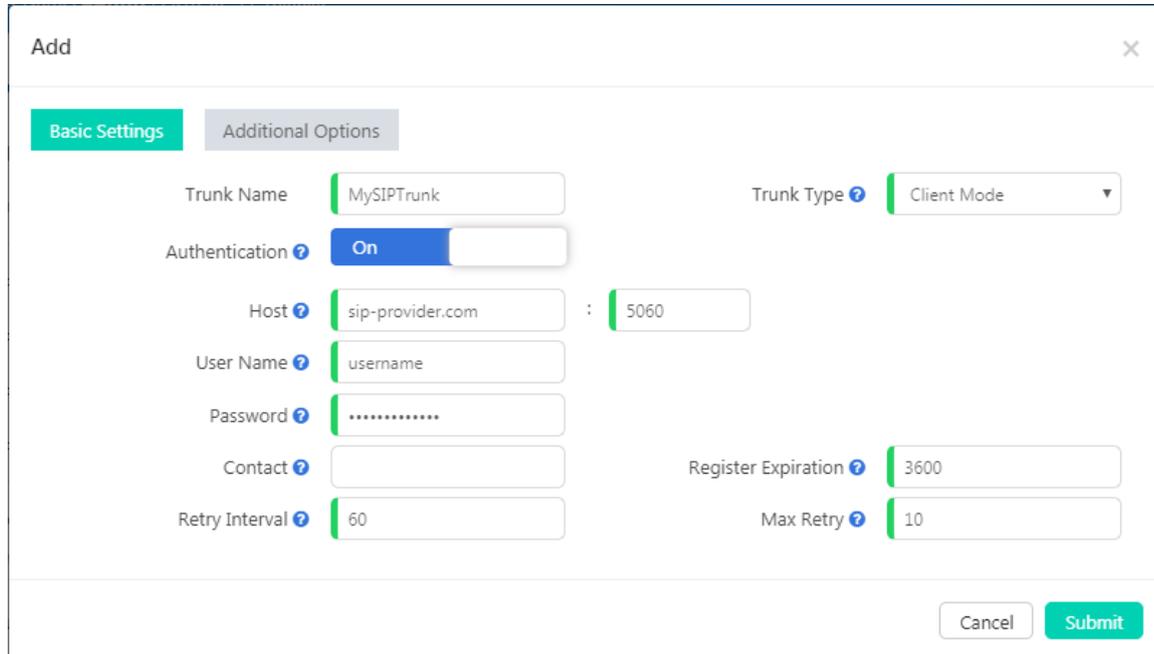


Most of the trunk settings will be given by the service provider, settings that are not mentioned by the provider you may leave them blank or use default values.
- **Authentication**: If the service provider doesn't require a username and password for this account to register to their server then you can enable this option.
- **Host**: The SIP server domain or IP address.
- **Username**: Username provided by VoIP Provider.

- **Password**: Password provided by VoIP Provider.
- **Contact**: Contact user to use in an outbound call request through this trunk.
- **Retry Interval**: Once registration expired, retry interval is the number of seconds system will wait before attempting to send another register request to the server.
- **Trunk Type**: In practical applications, client mode VoIP trunks are the most commonly used to connect to the VoIP providers for low cost, long distance and international phone calls, while server mode is only used when users want to do SIP trunking between IPPBXs.
- **Registration Expiration**: Expiration time of registration in seconds.
- **Max Retry**: Defines how many times the IPPBX system will attempt to register to the server before permanently giving up.

More advanced settings.
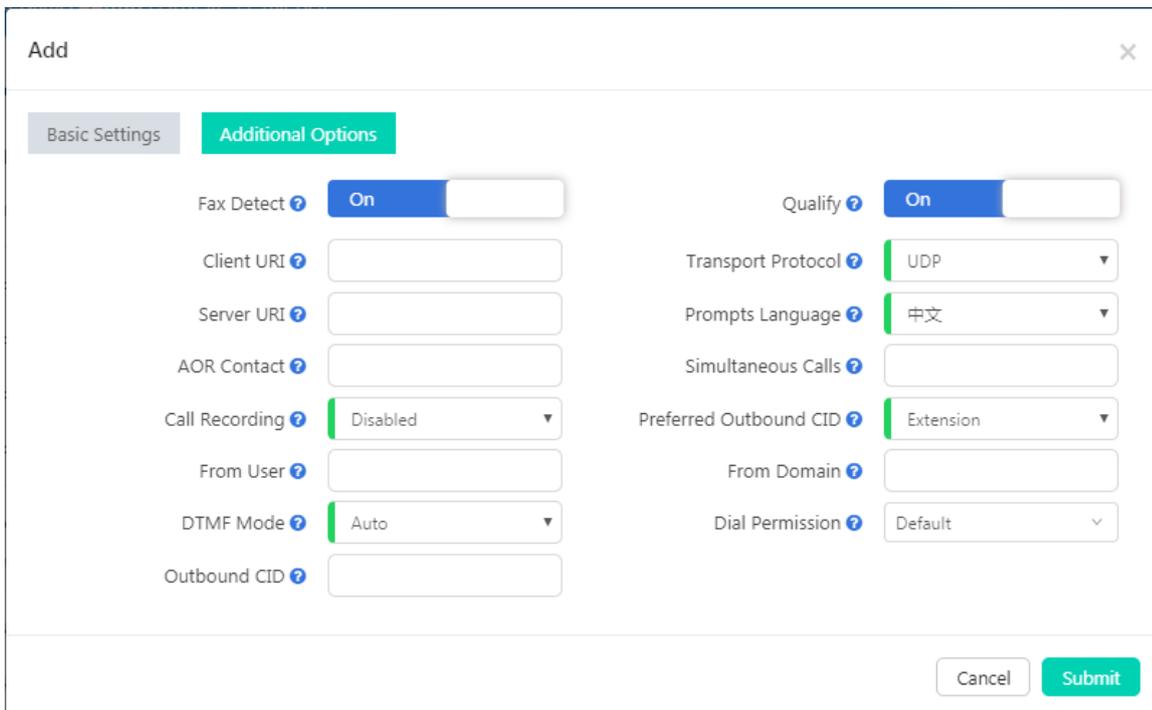


- **Fax Detection**: Enable/disable fax detection on this trunk.
- **Client URI**: Client SIP URI used when attempting outbound registration (e.g. SIP:22199518@sip.example.com:5060).
- **Server URI**: SIP URI of the server to register against (e.g. sip:sip.example.com:5060).
- **AOR Contact**: Address of records, it uses the same format as the client URI.
- **Call Recording**: Enable/disable call recording on this trunk. If enabled, all phone calls going in or out will all be recorded.
- **From User**: Username to use in "From" header for sending outbound call requests to this trunk.
- **DTMF Mode**: Used to inform the system how to detect the DTMF key press. Choices are Inband, rfc4733, SIP info and Auto.
- **Outbound CID**: The number you want to display to the called party while dialing out through this trunk. It depends on the service provider whether it works or not.
- **Qualify**: Qualify will cause the server sending SIP OPTIONS command regularly to check that the device is still online.
- **Transport Protocol**: To set the VoIP trunk to use UDP, TCP or TLS as the transport protocol, in most cases the

providers use UDP as default transport protocol.

- **Prompts Language**: You can choose a desired language of the system voice prompts to play for the incoming calls from this trunk. For example, if the call is not answered or the user is busy, the IPPBX system will notify the caller to leave a voice message in the language you set.
- **Simultaneous Calls**: This option will limit the number of simultaneous outbound calls can be made through this trunk, leave it blank as not limited.
- **Preferred Outbound CID**: To set preferred outbound CID of this trunk of the extensions.
- **From Domain**: Your service provider's domain name.
- **Dial Permission**: Custom dial permission for this trunk, by default it uses the "default" dial permission. Configure only if this trunk is for branch office integration, so calls coming from the other side can dial out from this IPPBX trunk directly. DO NOT change unless you fully understand how this feature works.

# Dial Rules

Path: ***Telephony -> Outbound Control -> Trunks***

On the IPPBX system you can setup different dial rules, for users to dial numbers in different format/pattern and cause the IPPBX system to call out through different trunk lines. For example, users dial the numbers with a prefix 9 to call out through the CO lines (land lines). Or dial the numbers with a prefix 00 to call out through the VoIP lines (SIP trunks).

Click on Add button to create a dial rule, below is an example dial rule.



- First of all in the **Dial Rule Name** blank specify a name to identify this dial rule.
- In the **Time Conditions** dropdown list, you may select a time condition for this dial rule, so this dial rule will only be available to be used at business hours.

- **Prepend** option is used to always add specific digit/digits in front of the actual dialed number after the **Dial Prefix** is deleted. These extra digits will be sent along with the actual number to the service provider to exchange. For example, if you want to always add an area code in front of the dialed number, you can spcify the area code in front of the dialed number, you can specify the area code here, otherwise leave this field blank.
- **Dial Prefix** is the first digit users have to dial while they want to make calls through the trunk/trunks selected in this dial rule. The system will strip the prefix from the number that is sent to the trunk.
- **Dial Patterns** act like a filter for matching numbers dialed with trunks. The various patterns you can enter are similar to Asterisk's definition of them:
    - **X** — Refers to any digit between 0 and 9
    - **N** — Refers to any digit between 2 and 9
    - **Z** — Any digit that is not zero. (E.g. 1 to 9)
    - **.** — Wildcard. Match any number of anything. Must match *something*.
- **Pin Set** is a collection of PIN codes for granting outbound phone calls.
- **Call Method** sets how to use the selected trunks for outbound phone calls.
    - **Linear**: Always take the first available trunk, if the first trunk is busy it will try the second trunk, if the second trunk is busy it will try the third, and so on.
    - **Linear Cycle**: Always take the next trunk, the trunk which the last had taken will not be used, it will call out through the next one directly.
- Double click one of the trunks or drag-and-drop to move the trunks from **Available Trunks** field to **Selected Trunks** field. The selected trunks will be used by this dial rule for outbound phone calls.

| | |
|---|---|
| Note | If you want all users to use the same dial rule for outbound phone calls, a dial prefix may not be necessary. But please make sure all available trunks should be included in the **Selected Trunks** field, otherwise unselected trunks will never be used.<br>If you want to set different dial rules please make sure the dial rules use different dial prefixes. |

# Dial Permissions

Path: ***Telephony -> Outbound Control -> Dial Permissions***

A dial permission consists of outbound dial permissions (dial rules) and internal dial permissions. Each extension number had been assigned with a dial permission. Dial rules are created for dial outbound phone calls, internal dial permissions are used for controlling extension number from using local phone system features.

You may create several different dial permissions. By assigning the extension numbers with different dial permissions you may limit the extension users to dial certain outbound phone calls and use certain local phone system features.

Click on [ Add ] button to create a new dial permission or you may use the default dial permission.

- In the **Dial Rules** section by moving the dial rules from the **Available Rules** field to the **Selected Rules** field to enable the dial rules in this dial permission. In the above given example, 2 dial rules had been enabled. The "Dial-FXO" rule is used to make phone calls through CO lines (land lines). The "dial-sip-trunk" rule is used to make phone call through the SIP trunk. So if you assign this dial permission to the extension users, they will be able to make outbound phone call both through CO lines and the SIP trunk.
- In the **Internal Permissions** section by switching the internal call features on/off to enable/disable the call features.
  - **Extension**: Allow/Disallow dialing other extension numbers.
  - **Paging & Intercom**: Allow/Disallow dialing paging & intercom group numbers.
  - **Conference**: Allow/Disallow using conference feature.
  - **Parked Call**: Allow/Disallow answering the parked calls.
  - **DISA**: Allow/Disallow using DISA feature.
  - **Pickup Exten**: Allow/Disallow pickup phone calls on other extensions.
  - **Feature Codes**: Allow/Disallow using feature codes.
  - **Queue**: Allow/Disallow dialing the call queue numbers.
  - **IVR**: Allow/Disallow dialing IVR extensions.
  - **Spy**: Allow/Disallow spying on other extensions' phone calls.
  - **Audio Console**: Allow/Disallow using feature code to broadcast through audio out interface (IPX-2200 and IPX-2500 only).

By default all extensions use the default dial permission "DialPlan1", if you have created new dial permissions, please don't forget to assign them to the extensions from *Telephony -> Extensions -> IP Extensions* and *Telephony -> Extensions -> Analog Extensions* (if there are analog extensions) page.

# PIN Sets

Path: **Telephony -> Outbound Control -> PIN Sets**

Pin sets can be used to secure your IPPBX system phone services and in particular for outbound dial rules and DISA.

Each PIN Set consists of a series of PIN Codes.
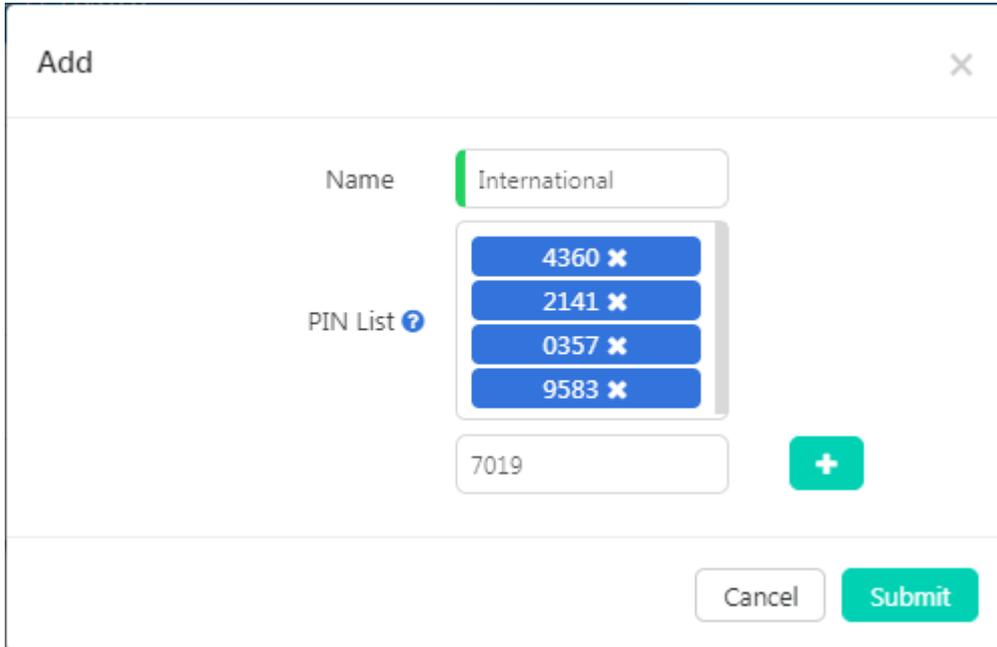


The PIN codes could be any digits that you want, but usually recommended it to be 3 to 5 digits meaningless numbers.

You could distribute these PIN codes out to each of the extension users or several of them to share a same PIN per your demand. If the PIN set is implemented on a dial rule or DISA, the IPPBX system will ask them to enter one of those PIN codes before they can call out.

The PIN codes also can be used to query call logs and recordings, so even if the extension user dialed a number from another extension if PIN code is used you'll know who actually made that call.

# Audio Library

## Music On Hold

Path: **Telephony -> Audio Library -> Music On Hold**

Music On Hold (MOH) is commonly known on an IPPBX system allows audio files (such as WAV or MP3 files) to be uploaded to the IPPBX system and played back when a caller is placed on hold or is waiting in a queue.

Audio files are managed by folder basis. You may use the system default MOH folder as on hold music or you may create new folders and upload your custom music files. Please first click on "Create New Folder" to create a new MOH folder.
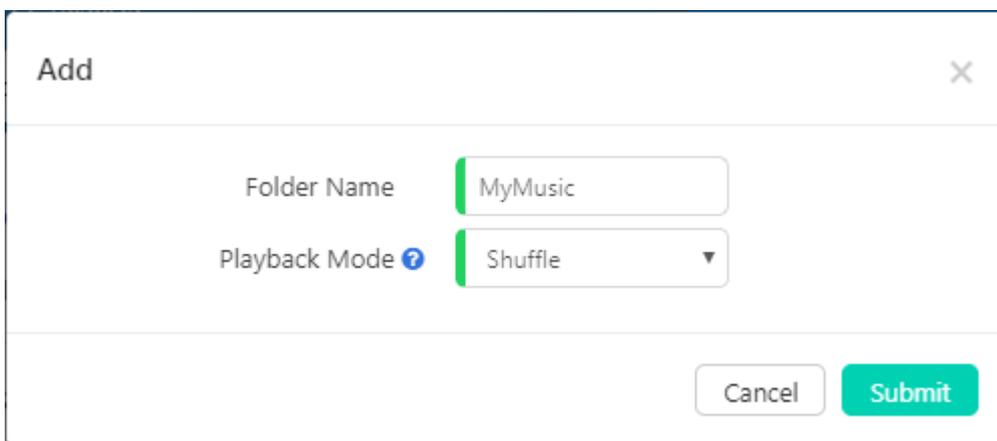


Give this folder a name and set the playback mode as shuffle (random playback) or in turn (playback in order). Once done click on "Submit".

Now click on [image] button to upload audio files to the newly created folder one by one.

Supported File Format: MP3, WAV(8KHz, 16bit, Mono)

## IVR Prompts

Path: **Telephony -> Audio Library -> IVR Prompts**

To configure an IVR menu on IPPBX system you'll first need to record your IVR prompts, these IVR prompts will communicate with the callers about the menu options that they have e.g. press one for sales.

Always be sure that the recorded IVR prompts will match the options to be set up in the IVR. If you change your IVR options, don't forget to change your recording!

The IVR prompts are pre-recorded and then uploaded to the IPPBX system.

| File | Format | Play ? | Options |
|------|--------|--------|---------|
| biz-cn | mp3 | | |
| biz-en | mp3 | | |
| language | mp3 | | |
| nobiz-cn | mp3 | | |
| nobiz-en | mp3 | | |

5 Total
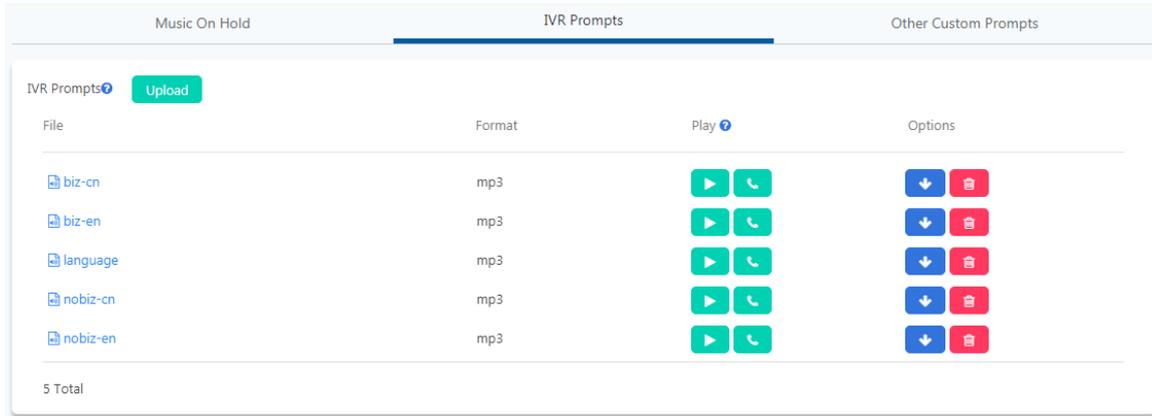
The recorded audio could be MP3 or WAV (16bit, 8KHz, Mono) format.

After uploading, you may playback on the web by clicking ▶ button or playback on a phone by clicking on the 📞 button.

# Custom Prompts

Path: *Telephony -> Audio Library -> Other Custom Prompts*

Custom prompts are to be used by call queue, call forward and some other advanced features, where customized voice prompts required.

You could record the voice prompts in MP3 or WAV (16bit, 8000Hz, mono) format and upload here. Then when you setup call queue periodic announcements you could select the customized voice prompts, or when you setup call forward notify message you could set the IPPBX system to notify callers before forwarding their calls.

# Advanced Features

## Call Forward

Path: **Telephony -> Advanced Features -> Call Forward**

Call forward allows calls to an extension to be forwarded to a specific internal extension number or an external phone number.

Before configuring call forward for user extensions, there are 2 advanced options you may want to know about.

| Advanced Options | |
|---|---|
| Passthrough Caller ID ❓ | Off |
| Notify Caller before Forwarding ❓ | Off |
| | Save |

● By enabling **Passthrough Caller ID** option to cause the IPPBX system passing the actual caller ID to the target number while forwarding.

● **Notify Caller before Forwarding** option allows you to choose a voice prompts to be played to the caller to notify caller that the call will be forwarded. The voice prompts is uploaded from **Telephony -> Audio Library -> Other Custom Prompts** page. If this option is not enabled, the call will be forwarded without notifying the caller.

To configure call forward please click on the [ Add ] button. And follow the explanations to complete the configurations as below.

| Add | × |
|---|---|
| Forward Extension | 131(131) ▼ |
| Forward Type ❓ | Forward All ▼ |
| Number ❓ | 965302387 |
| Enable | On |
| | Cancel   Submit |
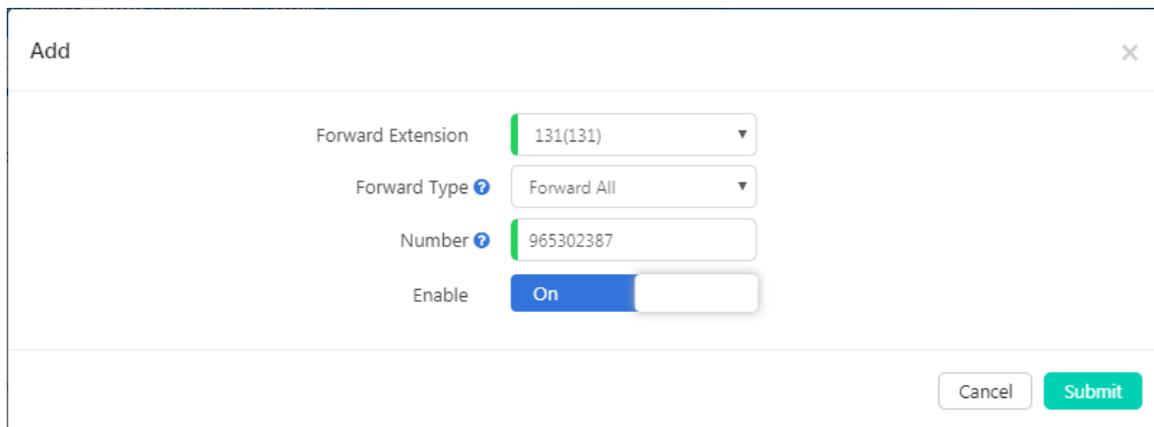
● In the **Forward Extension** dropdown list select the extension to be configured with call forward.
● In **Forward Type** dropdown list select the condition of when to forward the incoming calls.
● In the **Number** field specify the number to receive the forwarded phone calls. If it's another internal extension number, just fill in with that extension number. If it's an external number, you'll have to specify the dial prefix in front of the actual number. In this case, the actual number is 65302387, the dial prefix is 9.

In the forward list, you may disable or enable items based on requirements.

Call forward could be configured by Admin user and the operator user, and even by extension user from extension user web portal or by extension users from their phones by feature codes, please refer to Call Forward feature codes.

# Follow Me

Path: **Telephony -> Advanced Features -> Follow Me**

The Follow Me feature allows you to set a list of numbers for an extension number that the extension user may possibly be contacted on. Therefore, if someone calls the extension and the user is not available then Follow Me will work through the list calling each of the numbers in turn until the user is contacted or the list is exhausted. Before configuring Follow Me there are 2 advanced options you may want to know about.



- **Always Take Call**: If enabled the forwarded call will be answered directly by a destination number, if disabled, when the forwarded call reaches the destination number, IPPBX system will give the destination number choices to decline or answer the call.
- **Record Caller Name**: Ask the caller to say his/her name and record the name, so it can be announced to each destination number.

Add a follow me feature like below.

- Select the **Extension** which will be configured with Follow Me.
- **Ring Duration (Sec)**: To set the time in seconds to ring the extension before Follow Me process starts.
- **Follow Me List**: The list of numbers to be reached in order.
- **Number** and **Timeout (Sec)**: The number to be reached and the time to ring this number before trying the next one. If the number is an external number, don't forget to add a dial prefix in front of it.

Take the above settings as example, when extension 121 gets an incoming call, if it's not answered in 30 seconds, the call will be forwarded firstly to 122 and ring this extension for 30 seconds, if still not answered, it will try number 65302387 (9 is the dial prefix, not part of the number) for another 30 seconds. If extension 122 answered the call then 65302387 will not be called. If the call didn't answer by any of the numbers listed in **Follow Me List**, the Follow Me process will end and the caller will be disconnected.

# Wake Up Call

Path: ***Telephony -> Advanced Features -> Wake Up Calls***

Wake Up Call feature could be used to schedule reminders to the user extensions. Wakeup calls could be scheduled by admin user from admin Web interface, by operator user from operator Web interface, or could be scheduled by extension users by dialing Wake Up Call feature codes.

To schedule a wakeup call from admin user Web interface, please click on Add button, in the popup window set the time of the wakeup call and select the extension/extensions to be called at the scheduled time point.
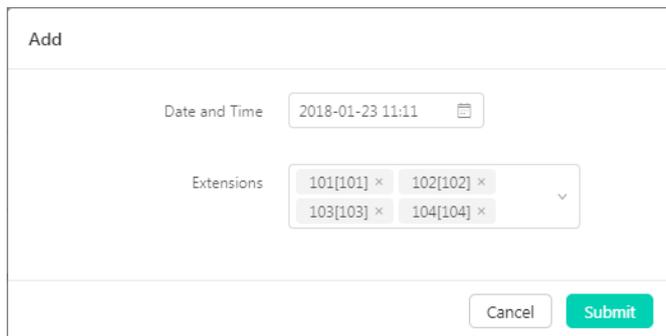


- Click on **Date and Time** field to schedule the date and time for the wakeup call.
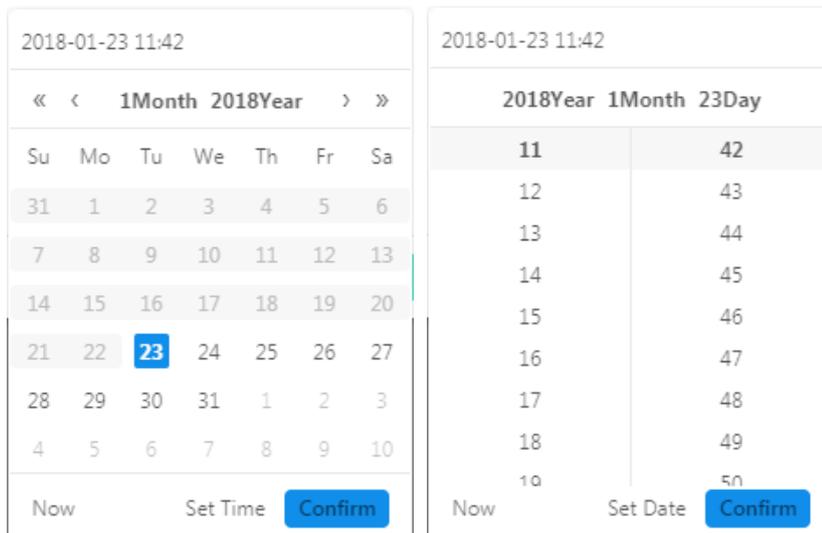


Click on **Now** to fill in the blank with the current date and time, and then you can modify as per your requirements.

Or you may choose a date and then click on **Set Time** to set the exact time of the scheduled wakeup call.

● In **Extensions** field you could select one or more extensions as you wish.

When it's time for wakeup call, IPPBX system will ring the selected extension/extensions. After user answering the wakeup call, IPPBX system will give user options to "C**onfirm"** wakeup call or to "**Snooze**". If user press a key to confirm wakeup call then the schedule is completed, if user chooses to snooze, then the wakeup call will ring back later as per the user selected by key press.

If a wakeup call is not answered, system will try to ring back in the next minute, and will retry 2 times, after which system will consider the wakeup call completed, and the schedule will be canceled.

# Conference

Path: *Telephony -> Advanced Features -> Conference*

Conferences allow two or more callers to be joined together so that all parties on the call can hear one another. Conferences are also referred as Conference Bridges or Conference Rooms.

There are 10 conference numbers for internal extension users to dial to join conference calls. You can also set conference as a destination in inbound routes to allow outside callers to reach the conferences.

| Call Forward | Follow Me | Wake Up Call | Conference | DISA | Paging & Intercom | ❯ |

| Extension ❓ | Guest Password ❓ | Admin Password ❓ | Leader Wait ❓ | Announce Caller ❓ | Call Recording ❓ | Options |
|---|---|---|---|---|---|---|
| 90 | 1234 | 2345 | Yes | Yes | Yes | ✏️ |
| 91 | 1234 | 2345 | No | No | No | ✏️ |
| 92 | 1234 | 2345 | No | No | No | ✏️ |
| 93 | 1234 | 2345 | No | No | No | ✏️ |
| 94 | 1234 | 2345 | No | No | No | ✏️ |
| 95 | 1234 | 2345 | No | No | No | ✏️ |
| 96 | 1234 | 2345 | No | No | No | ✏️ |
| 97 | 1234 | 2345 | No | No | No | ✏️ |
| 98 | 1234 | 2345 | No | No | No | ✏️ |
| 99 | 1234 | 2345 | No | No | No | ✏️ |

10 Total

There are totally 10 conference numbers could be used. Only users who dial the same conference number could hear one another.

There are options for each conference for you to customize the conference feature. Please click the ✏️ button to change the options if needed.

- **Guest Password** is for ordinary conference users, only the users who enter the correct password can join in the conference.
- **Admin Password** is for conference admin, only the user/users who enter the admin password will become the conference administrator. Conference admin can invite other numbers to join in the conference by using Conference feature codes.
- **Dial Permission** could be used by the conference admin user to dial other numbers and invite them to join in the conference. By default, all conferences use Internal dial permission, which means by default the conference admin could only invite internal extension numbers to join in the conference, if inviting external number is necessary, please select a valid dial permission which could be used to dial external numbers.
- **Leader Wait**, if enabled, the conference will start when the conference admin entered. Before conference admin joining in, all other participants will be waiting with background music on.
- **Announce Callers** option causes the IPPBX system to notify all conference participants about new participants join-in. Before a new participant joining in the conference, the IPPBX system will ask the participant to say his/her name, once done, system playback the recorded name to other participants and at the same time, new participant joins in.
- **Call Recording** option determines whether the conferences to be held in this "conference room" should be recorded or not.

# DISA

Path: **Telephony -> Advanced Features -> DISA**

Direct inward system access (DISA) allows an outside caller to dial directly into the PBX system and access the system's features and facilities remotely.

It's useful if you want people to be able to, for example take advantage of the low rate for international calls that you have available on your system, or to allow outside callers to be able to use the paging or intercom features of the system. Always protect this feature with strong password/passwords, the passwords need to be set on PIN Sets page.

To add a DISA feature follow the explanations below.
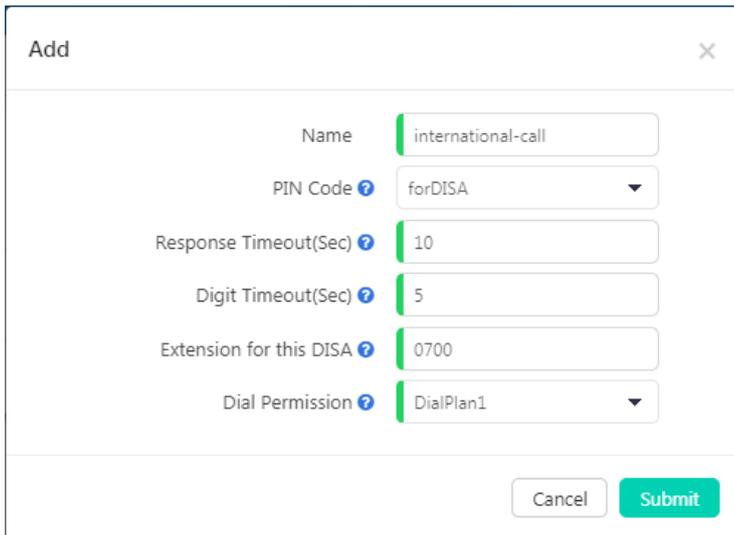
- In the **PIN Code** dropdown list select a valid PIN Set. The PIN codes of this PIN set will be used to authorize all callers using the system features and facilities.
- **Response Timeout(sec)**:The maximum waiting duration before hanging up if the dialed number is incomplete or invalid. Defaulted 10 seconds
- **Digit Timeout(sec)**:The maximum interval time between digits when typing extension number. Defaulted 5 seconds.
- **Extension for this DISA**: If you want to access DISA by dialing an extension, you can define an extension number for this DISA.
- **Dial Permission**: Select a dial permission for this DISA so callers will be able to make outbound phone calls using the trunks on the IPPBX system.

# Paging & Intercom

Path: ***Telephony -> Advanced Features -> Paging & 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 an extension number that by calling the number will simultaneously page/intercom a group of phones.

To create a **Paging & Intercom** group, please click on the [Add] button, a popup window will show up as below.
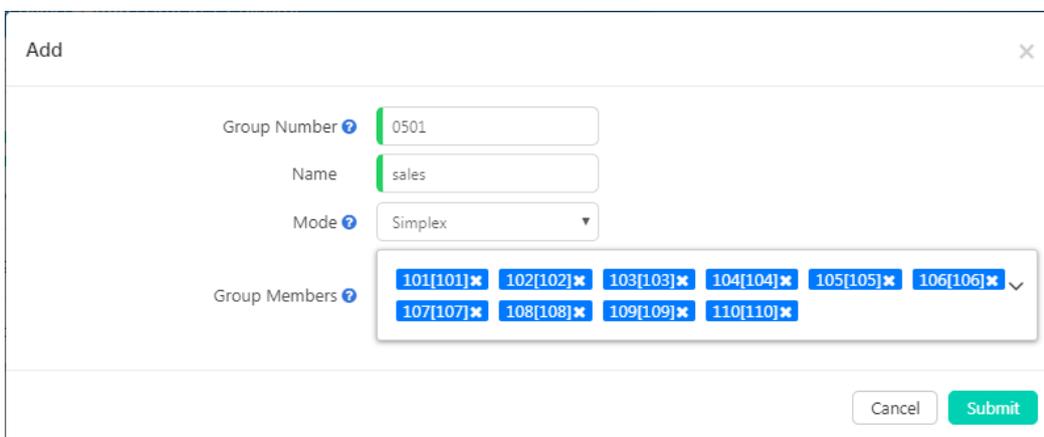


- In the **Group Number** field, a default group number is given. The number could be changed within the Paging

Group Extension Number Range listed on **Telephony -> Preferences -> Global PBX Options** page, Extension Ranges section.

● In the **Name** field a name should be given to identify this paging group.

● In the **Mode** dropdown list, if "Simplex" is selected, calling the group number will page on the group members, if "Duplex", the group members are able to talk back to the caller (intercom).

● In the **Group Members** field, select the desired user extensions, make sure all extensions you selected are desktop based IP phones, otherwise if the phone is an analog one, paging/intercom will not work.

Except group paging and intercom, extension users could also paging/intercom an individual extension by using feature codes, please refer to introductions in Other feature codes section.

# Call Queue

Path: **Telephony -> Advanced Features -> 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. Once a call queue has been created, it can be assigned to specific extensions and configured to feature greetings, messages, and hold music.

To create a call queue, please click on the [Add] button, a popup window will show up as below:
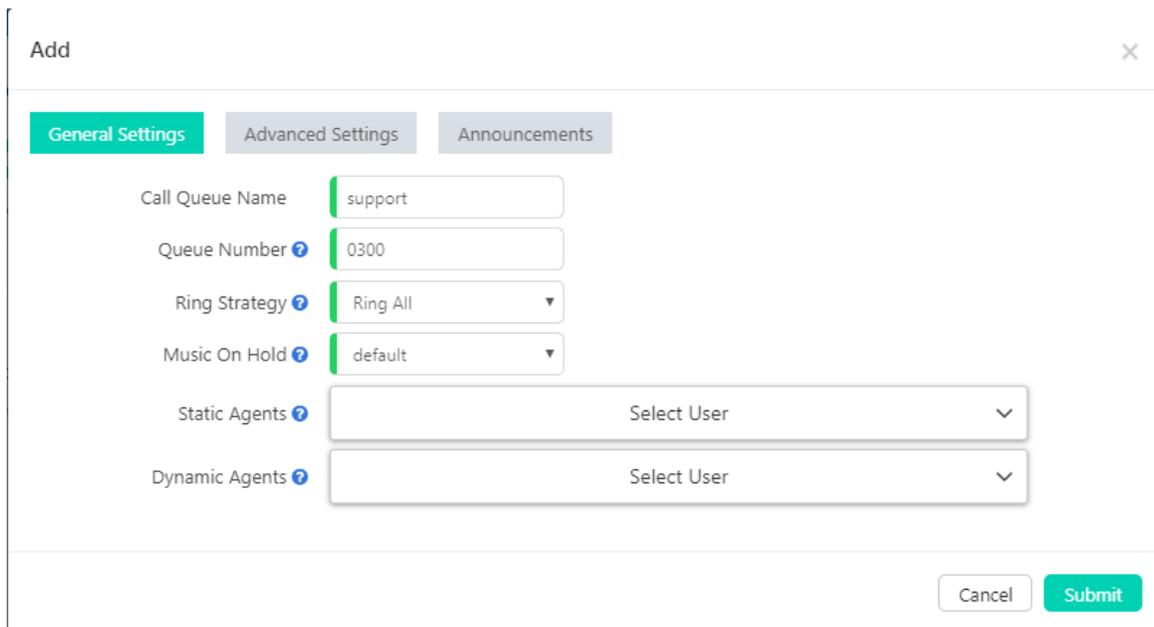


First please complete the **General Settings**.

● In **Call Queue Name** field specify a name to identify this queue.

● In **Queue Number** field a default number is given. The number could be changed within the Paging Group Extension Number Range listed on **Telephony -> Preferences -> Global PBX Options** page, Extension Ranges section.

● **Ring Strategy** sets the method how you wish the queue agent extensions to ring when there's incoming call to this queue.
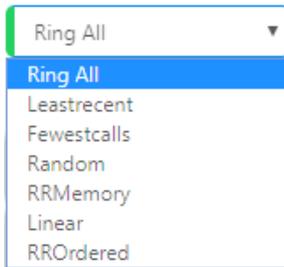
- ■ **Ring All**: Ring all available agents until one answers (default).
- ■ **Leastrecent**: Ring the extension of the Agent who has least recently received a call.
- ■ **Fewestcalls**: Ring the extension of the Agent who has taken the fewest number of calls.
- ■ **Random**: Ring the extension of a random Agent.
- ■ **RRMemory**: The system remembers which agent was last called and begins the round robin with the next agent.
- ■ **Linear**: Starting with the first agent, ring the extension of each agent in turn until the call is answered.
- ■ **RROrdered**: Same as RRMemory, except the queue member order is preserved.
- ● In the **Music On Hold** dropdown list select a music folder as hold music when callers are waiting in the queue.
- ● **Static Agents** are extensions that are assumed to always be in the queue. Static agents do not need to "log in" to the queue, and cannot "log out" of the queue.
- ● **Dynamic Agents** are extensions that can log in and out of the queue. Extensions selected here will NOT automatically be logged in to the queue.

More advanced options for call queue is available, please click on [Advanced Settings] button to show advanced options, they are optional but might be useful to improve the callers' experiences.



- ● **Auto Fill** if it's set to be Yes, and multiple agents are available, the PBX will send one call to each waiting agent (depending on the ring strategy). Otherwise, it will hold all calls while it tries to find an agent for the

- top call in the queue, making the other callers wait.
- **Agent TimeOut** specifies the number of seconds to ring an agent's extension before sending the call to the next Agent (based on Ring Strategy).
- If an agent's extension rings and the agent fails to answer the call, **Auto Pause** option can automatically pause that agent to stop them receiving further calls from the queue.
- **Wrap Up time** is the amount of time in seconds that an agent has to complete work on a call after which the call is disconnected.
- If **Report Hold Time** is enabled, it will report to the agent about how long the caller had been waiting in the queue.
- The value of **Max Callers** limits the maximum amount of callers can wait in the queue (Default is 0 -- unlimited). When the maximum number of callers in the queue is reached, subsequent callers will be sent to the **If no answer** destination.
- If **Add Queue Name Caller ID** option is enabled, when an incoming call is distributed to an agent the queue name will be displayed on the phone screen along with the caller ID. So a call queue agent knows which call queue the call is coming from. This feature is helpful if an agent belongs to multiple call queues.
- Calls that have been waiting in the queue for **Max Wait Time(Sec)** will be sent to the **If no answer** destination. If left blank, there will not be any time limitation of waiting time.
- **Join Empty** option allows callers to enter the queue when no agents are available. If this option is not enabled, callers will not be able to enter the queue without available agents - callers will be sent to the **If no answer** destination.
- **Leave When Empty** option if it's enabled and calls are still in the queue when the last agent logs out, the remaining callers in the Queue will be transferred to the If no answer destination. This option cannot be used with Join Empty at the same time.

You may set the system to playback announcements to the callers while they are waiting in the queue. Please click on the **Announcements** button to setup customized announcements.

- **Caller Position Announcements** is used to tell the callers how they've been waiting and the position in the queue.
  - **Announce Hold Time:** Announce to the callers of the time they have been waiting, the first minute callers waiting in the queue will not hear such announcements.
  - **Announce Position:** If set to be Yes, the system will announce the position of the caller is currently waiting in the queue.
  - **Broadcast Frequency(Sec):** To defines how often to announce queue position and estimate hold time.
- **Periodic Announcements** can be used to periodically playback a voice prompts to the callers waiting in the queue.
  - Repeat Frequency(Sec): The time interval to repeat this periodic announcements.
  - Announcements Prompts: To select a voice prompts to be periodically played to the waiting callers.
- **Destination if no answer** sets the final destination for the callers if no one answers the call when they were in the queue.

After setting up the queue, you may use internal extensions (non-agent extensions) to call the queue number to verify the queue settings.

# Smart DID

Path: *Telephony -> Advanced Features -> Smart DID*
With Smart DID feature, the IPPBX system has the ability to route an inbound call directly to an extension if the extension had previously called or tried to call the external number. It is convenient for the called party to make a call back and be directly routed to the extension that called them without going through the IVR menu or reception desk.
For example, extension 100 called external number 1234567, no matter this number answered or not, when the number tries to ring back, the call will go directly to extension 100.
If you want this to happen, please use the **Enable Smart DID** switch to turn on this feature.
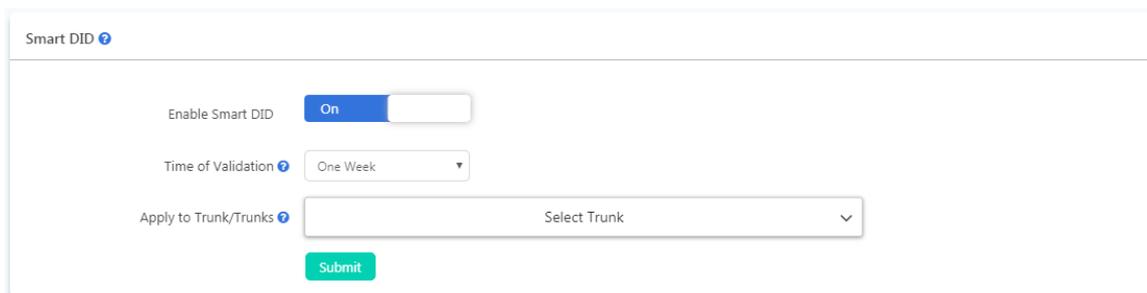


In **Time of Validation** dropdown list choose how long the system to save these outbound call records. When the records expired, the inbound calls will be routed according to you inbound routes settings.
In **Apply to Trunk/Trunks** field, you have to select the trunk/trunks Smart DID feature will be applied to.

# Phonebook

Path: *Telephony -> Advanced Features -> Phonebook*

Phonebook feature for IPPBX is just like a contact list on the mobile phones. You may add contacts to the IPPBX system, when the contacts calling in, on the ringing user extension phone screen will display the caller number and the contact name you have added before. If the number didn't match any contacts in the phonebook, then only caller number will be displayed on the ringing phone screen.

You may click on the [Add] button to add a new contact from the popup window.
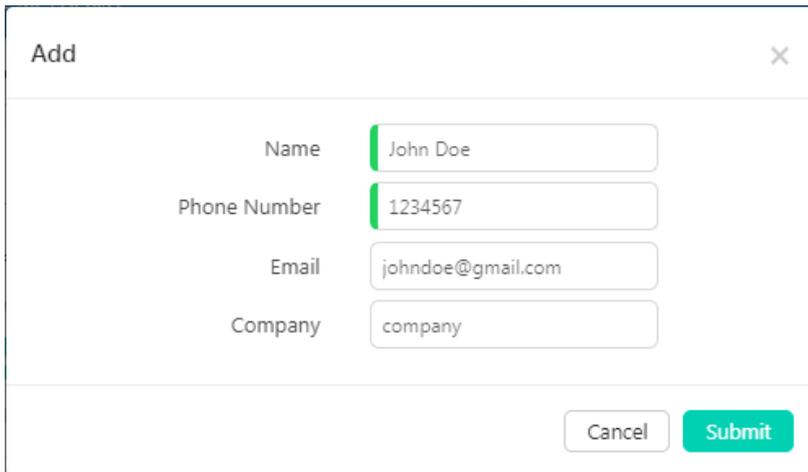


Or you may export the phonebook template file to add the contacts by MS Excel and then upload the file to generate contacts.

Contacts could be added by admin user from admin web interface, by operator from operator web interface and by extension user from extension user web portal.

A contact added by admin user and operator user is visible to all extension users, but a contact added by an extension user is only visible to the user who added it and the admin and operator user, other extensions won't be able to see it.

# LDAP

Path: *Telephony -> Advanced Features -> Phonebook*

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. An LDAP server has been pre-configured on IP PBX which is mainly used to centralize manage the phonebook.

If you are using VIP-1120PT or VIP-2140PT, LDAP configurations will be automatically configured when you were provisioning (no matter by PNP or Quick Register Code) the phones. No more LDAP configurations required. You may browse IPPBX phonebook directly from the IP phone LDAP menu.

If you are using IP phones from other manufacturers you'll need to manually configure LDAP configurations using the LDAP server credentials as below.
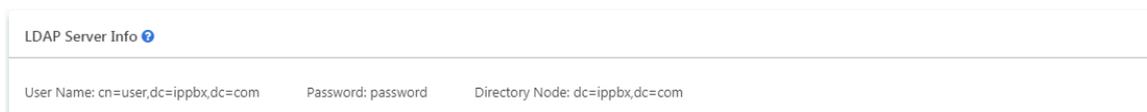
# Feature Codes

Path: **Telephony -> Feature Codes**

Feature codes can be dialed from user extensions to enable and disable certain features or to achieve some call features. For example, enable and disable call forward, transfer incoming calls, check voice messages, etc.

## Voicemail feature codes



Dial *60 and you will enter the main menu of voicemail feature, by specifying the extension number and voicemail password of the required extension then you can check its voicemail and you can do this for any extension by following the system voice guidance.

By dialing *61 from an extension and entering the voicemail password for this extension you can follow the voice guidance to check voicemail of your own extension. Or alternatively, you can configure some advanced options for your voicemail box.

## Call Pickup feature codes

Call pickup feature codes allow users to pick up calls that are not directed to them by dialing a feature code *8 or **.



If there's an incoming call ringing on an extension that belongs to your department, you may dial the general call pickup feature code *8 (end with #) to pick up the call. While if there are 2 ringing extensions in your department, by dialing *8 will pick up the first incoming call. If you need to pick up the second incoming call or if you don't know which call came first, you may use direct call pickup feature code.

Direct call pickup feature code could be used to pick up an incoming call on a specific extension, no matter the extension is from the same department or from another department. Just dial ** following by the extension number (end with #) you'll be able to pick up the incoming call on that specific extension.

# Call Parking feature codes

Call parking feature allows anyone who has received a call to park the call on an extension, allowing any other user to access the parked call.

| Call Parking ❓ | |
|---|---|
| Parking Number: | *4 |
| Call Retrieve Number: | 41-49 |

To park a call, extension user could dial *4 during a live call, and then listen as the system tells you where you can retrieve the call (usually extension 41). The second call will be parked on 42, and it continues to park on orderly.

To retrieve the parked calls, user should dial the retrieve number given by the IPPBX system. And this could be done by any extension.

A call could be parked for maximally 120 seconds before it goes back to the extension which parked it. And the parking lot (call retrieve numbers) could be monitored by BLF. It's helpful if the operator wants to know if there are calls parked on the IPPBX system.

# Call Transfer feature codes

Call Transfer is used to transfer a call in progress to some other destinations. 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 referred to "Supervised Call Transfer".

- Blind call transfer - Where the call is transferred to the other destinations without intervention (the other destination could ring out and may not be answered for instance).

| Call Transfer ❓ | |
|---|---|
| Blind Transfer: | # |
| Attended Transfer: | *2 |
| Abort Transfer: | * |

In a live call, you can press # key and the IPPBX system prompts "Transfer", you then enter the number to transfer to, this call will be transferred instantly and the user can hang up. If the transferred number doesn't answer this call then it will go to voicemail.

If blind transfer sometimes seems inappropriate, you may use attended transfer feature. In a live call, you can press *2 and the IPPBX system prompts "Transfer", you then enter the number to transfer to, after someone answers your call, you can introduce this call and hang up at which point the call is transferred.

In an attended transfer, if the third party rang for 15 seconds without answering, the extension user will go back to the caller and the transfer is terminated. You may also manually abort the transfer by pressing * when the third

party is still ringing.

# Blacklist feature codes

Black list feature codes allow the extension users to add external phone numbers to IPPBX system blacklist from their phones, consequently the numbers added will not be able to dial in to the IPPBX system.

Adding blacklist numbers from phone by using feature codes is the same as adding blacklist numbers from admin and operator UI.

| Blacklist ❓ | |
| --- | --- |
| Blacklist the Last Caller: | *76 |
| Blacklist a Number: | *75 |
| Whitelist a Number: | *075 |

Blacklist the last caller allows you to dial *76 to directly add the last caller's number to the IPPBX blacklist.

You may also dial *75 (end with #) and follow the voice prompts to specify the number you wish to blacklist to add numbers to the IPPBX system blacklist.

To remove numbers from black list (whitelist a number), you can dial *075 (end with #) and follow the voice prompts to specify the number you wish to whitelist.

# Call Spy feature codes

Call Spy allows users to dial the spy feature codes following by an extension number to listen to the call conversation in real-time.

| Call Spy ❓ | |
| --- | --- |
| Normal: | *90 |
| Whisper: | *91 |
| Barge: | *92 |

- Normal Spy: For example, extension 410 is talking to someone on the phone, you can dial *90410 (end with #) to listen to their conversation, however, neither speaker will be able to hear you.
- Whisper Spy: Whisper spy is also known as coaching. For example, a new employee is talking to the customer on the phone, their supervisor can dial *91 following by the employee's extension number (end with #) to listen to their conversation. The supervisor can talk to the new employee only without the customer hearing the conversation.
- Barge Spy: Barge spy is similar to an instant 3-way conference call. While an extension user is talking to someone else on the phone, you can dial *92 following by their extension number (end with #) to talk to both of the speakers.

| Note | Before you can spy on an extension, please enable "Call Spy" option on the extension edit popup window. |
|------|--------------------------------------------------------------------------------------------------------|

# Call Queue feature codes

Call queue feature codes are for call queue agent extensions only. They are meaningless to the non-agent extensions.

| Call Queue ❓ | |
|---------------|------|
| Agent Login: | *62 |
| Agent Logout: | *062 |
| Agent Pause: | *95 |
| Agent Unpause: | *095 |

**Agent Login** and **Agent Logout** are for dynamic agents to login or out of the call queue. And for both static agents and dynamic agents, they can dial *95 to suspend their extensions temporarily, new calls will not be distributed to their extensions, until they dial *095 to resume.

# Conference feature codes

Conference feature codes are used by conference admin for inviting participants to join in a conference or for creating a conference during a normal phone call.

| Conference ❓ | |
|---------------|------|
| Invite Participant: | 0 |
| Return with Participant: | ** |
| Return without Participant: | *# |
| Create Conference: | *0 |

When in a conference room, if the conference admin user presses 0 they will get a dial tone for inviting others to participate in this conference.

If the invited party agrees to join in the conference, conference admin user can dial ** to return to the conference with invited party.

If the invited party doesn't want to join in the conference, conference admin user can press *# to return to the conference without the invited party.

During a live call the extension user can press *0 to create a dynamic conference room. The other side will automatically enter the conference as an ordinary participant while the extension user who created this conference will be requested to enter the conference password to enter. Usually the user needs to enter the conference admin user password as the user needs to invite others to join in the conference.

| | After a dynamic conference is created, in reality you have entered a static conference room (by default 90 is the first available conference room). You are able to use conference admin menu to invite others to the conference and also others can dial 90 to enter this conference. |
|---|---|
| Note | |

# Wakeup call feature codes

Except configuring Wakeup Calls from admin and operator web user interface, extension users could request wakeup calls from their phones directly by feature codes.

| Wakeup Call ❷ | |
|---|---|
| Cancel All Wakeup Calls: | *055 |
| Direct Wakeup Call Request: | *55* |
| Wakeup Call Main Menu: | *55 |

- **Cancel All Wakeup Calls**: By dialing this code to cancel all requested wakeup calls.
- **Direct Wakeup Call Request**: Add a wakeup call directly by dialing this feature code followed by a specific date and time in 8-digit number format, for example, feature code is *55*, you can dial *55*08010730 to add a wakeup call of 7:30am on August 1st.
- **Wakeup Call Main Menu**: Advanced wakeup call menu for adding, viewing and canceling wakeup calls.

# Call Forward feature codes

Call forward could be configured from admin and operator web user interface. With the following feature codes, extension users can activate or deactivate call forward directly from their phones without configuration on the Web GUI.

| Call Forward ❷ | |
|---|---|
| Forward All Activate : | *71 |
| Deactivate All: | *071 |
| Activate Forward on Busy: | *72 |
| Deactivate Forward on Busy: | *072 |
| Activate Forward on No Answer: | *73 |
| Deactivate Forward on No Answer: | *073 |

For example, a IPPBX requires prefix 9 to call outbound, and the number you want to forward the calls to is 85337096.
- **Forward All Activate**: Dial *71985337096, press 1 to confirm.
- **Deactivate Forward All**: Dial *071.

- **Activate Forward on Busy**: Dial *72985337096, press 1 to confirm.
- **Deactivate Forward on Busy**: Dial *072.
- **Activate Forward on No Answer**: Dial *73985337096, press 1 to confirm.
- **Deactivate Forward on No Answer**: Dial *073.

## DND feature codes

DND (Do Not Disturb) could be set on the IP phones from the phone level, if the phone doesn't have DND feature you may use the DND feature code to set DND from IPPBX system level. Any phone connected to the IPPBX system can use the DND feature code, no matter it's IP phone, analog phone or softphone.

| DND ❓ | |
| --- | --- |
| DND Activate: | *74 |
| DND Deactivate: | *074 |

Simply dial *74 to enable DND, if you hear a beep sound that means DND is on. Once DND enabled, the extension will only be able to make calls, and inbound calls will be rejected.

Make sure when you are ready to receive inbound calls, dial *074 to deactivate DND.

## Other feature codes

| Others ❓ | |
| --- | --- |
| Announce WAN Port IP: | **11 |
| Announce LAN Port IP: | **12 |
| Announce Extension Number: | **13 |
| One Touch Recording: | *1 |
| Intercom: | *50 |
| Paging: | *51 |
| Audio Console: | *911 |

- **Announce WAN Port IP**: By dialing this code you'll hear the system announce the IP address of the IPPBX WAN interface.
- **Announce LAN Port IP**: By dialing this code you'll hear the system announce the IP address of the IPPBX LAN interface.
- **Announce Extension Number**: By dialing this code you can check the extension number of your phone, either it's an IP phone or analog phone.
- **One Touch Recording**: One Touch Recording is also known as Record on Demand. It allows users to record phone calls selectively. In a live call conversation, an extension user can use feature code *1 to record this call.

With this feature, you don't have to configure recording all calls for the extensions which may cause heavy system resource use if some call recordings are not required.

● **Intercom**: The intercom feature code allows you to intercom one extension only. You don't have to create a "Paging and Intercom" group for only one extension if you intend to intercom with only that extension.

● **Paging**: The paging feature code allows you to page one extension only. It's the same as the intercom feature code, the only difference between paging feature code and intercom feature code is by using intercom feature code both sides can talk to each other but using paging feature code, only the caller can talk to the called party.

● **Audio Console**: A speaker could be connected to the 3.5mm audio out (available on IPX-2200 and IPX-2500) interface, extension user could dial *911 to deliver a public address from the audio out interface.
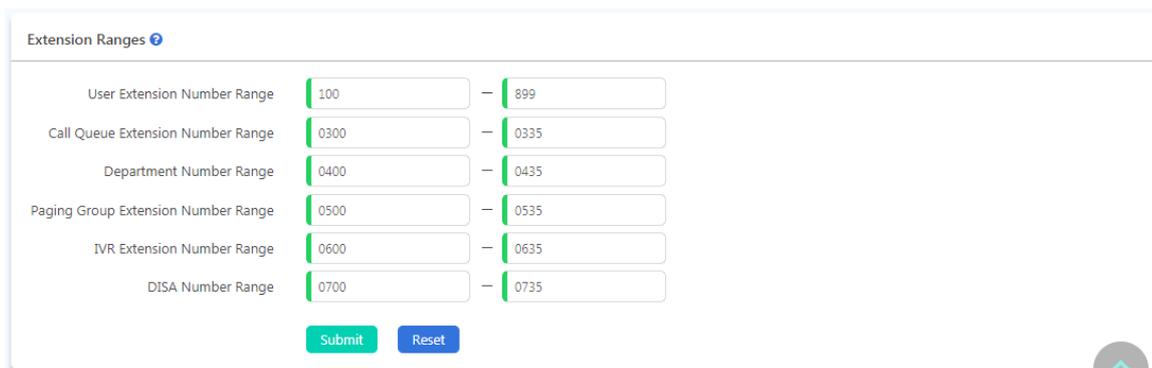
# Preferences

## Global PBX Options

Path: ***Telephony -> Preferences -> Global PBX Options***
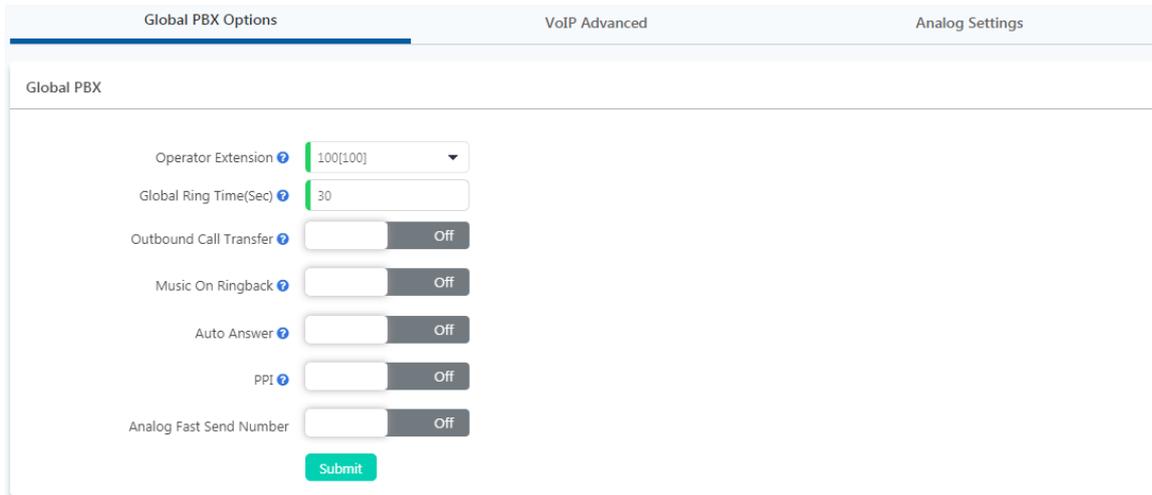


- **Operator Extension**: Choose an extension to be operator extension. When an incoming call has been directed to voicemail, then by pressing '0' the caller will be put through to the operator extension.
- **Global Ring Time**: If it's not specifically configured, an incoming call will ring the extension for the time given here.
- **Outbound Call Transfer**: Allow outbound phone calls to be transferred, if enabled it might cause phone call problem in certain situations. For example, an outbound phone call had been placed to another IVR system, the keypress might be recognized as transfer request on your own IPPBX system.
- **Music On Ringback**: If enabled, callers will hear music instead of ringback tone when calling extensions.
- **Auto Answer**: Auto-answer enables the IPPBX to automatically answer the inbound calls from analog ports.
- **PPI**: The P-Preferred-Identity (PPI) 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.
- **Analog Fast Sent Number:** This parameter will always add a # at the end of the numbers dialed through the PSTN lines, it will cause the carrier to forward the calls immediately instead of waiting till digits timeout. If your PSTN line carrier supports this feature, you may enable it for faster outbound phone calls via PSTN.
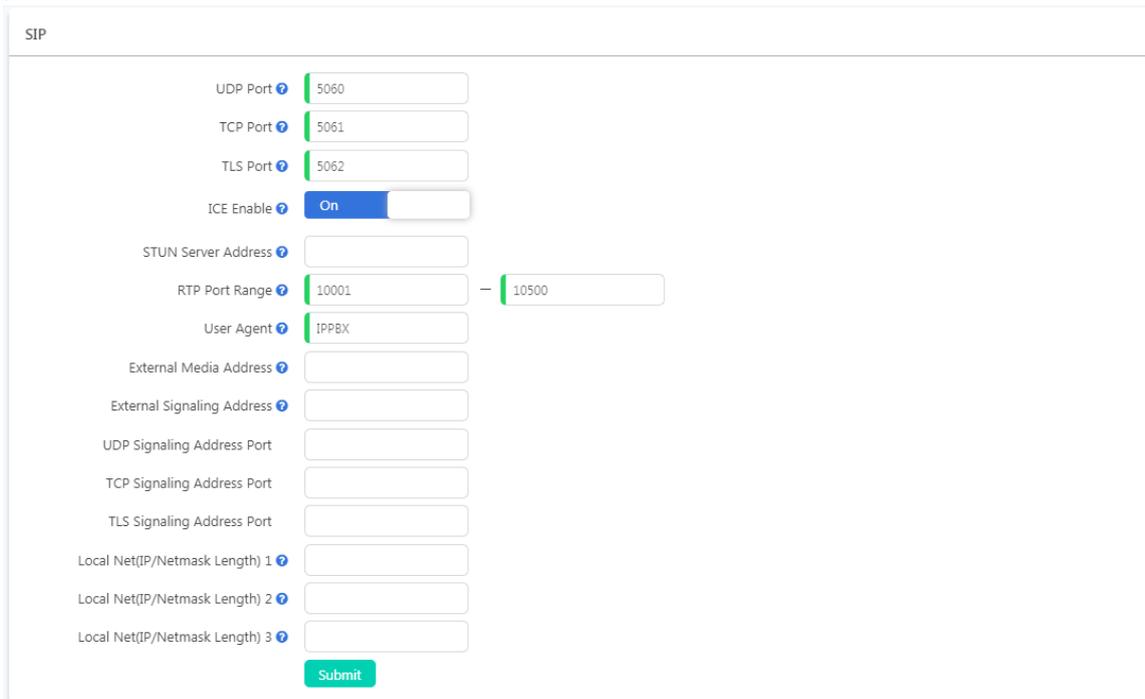
The user extension number and system extension number ranges are defined here to avoid any conflicts within the IPPBX system. You can modify these number ranges as per your requirements.

# VoIP Advanced

Path: **Telephony -> Preferences -> VoIP Advanced**
Global SIP settings allow you to configure some general and advanced options for the IPPBX system global SIP preferences.



- **UDP Port**: SIP over UDP service port. By default IPPBX system uses UDP as SIP transmission protocol. Port number can be changed here if required. If changed on the IPPBX system, you'll also have to change on the SIP clients.

- **TCP Port**: If the IP phones support TCP protocol, you can register SIP extensions over TCP protocol on port 5061.

- **TLS Port**: If the IP phones support TLS protocol, you can register SIP extensions over TLS protocol on port 5062.

- **ICE Enable**: This is specific to clients that support NAT traversal for media via ICE, STUN, TURN. By default, please keep it enabled, otherwise WebRTC won't work, **STUN Server Address** can be left blank.

- **STUN Server Address**: By default please keep it blank, if you got available STUN server, please specify the valid server address, otherwise an invalid STUN server address will cause phone system exception.

- **RTP Port Range**: The UDP ports used by the IPPBX system to carry RTP voice stream. Do not change the port range or you may encounter audio issue with phone calls.

- **User Agent**: The default user agent string also contains the Asterisk version. If you don't want to expose it, change the user agent string here.

- **External Media Address**: If you want to map your IPPBX system to the Internet, you should specify the static

public IP address or domain name here.

● **External Signaling Address**: This is similar to External Media Address except that the External Signaling Address is looked up regularly (every 10s).

● **UDP Signaling Address Port**: Port number of SIP signaling with UDP transport protocol on the public network.

● **TCP Signaling Address Port**: Port number of SIP signaling with TCP transport protocol on the public network.

● **TLS Signaling Address Port**: Port number of SIP signaling with TLS transport protocol on the public network.

● **Local Net (IP/Netmask Length)**: Your local network address/addresses.

| | |
|---|---|
| **Note** | If you are going to map your IPPBX system to the Internet, the following configurations should be done.<br>1. SIP port mapping on your router (one of the following: UDP: 5060; TCP: 5061; TLS: 5062).<br>2. RTP port mapping on your router (UDP: 10001 to 10500).<br>3. Specify External Media Address and External Signaling Address.<br>4. Specify your local network address/addresses.<br>5. For extensions remote registration, enable "Remote Extension" on extension edit popup window.<br>Mapping your IPPBX to the Internet will be risky, for security precautions please always use strong passwords. |

**IAX ❓**

| | |
|---|---|
| UDP Port | 4569 |

Submit

IAX2 extension support had been enabled by default for all extensions. And IAX2 works on UDP port 4569, you may modify the port number if required.

Asterisk supports different QoS settings at the application level for various protocols on both signaling and media. The Type of Service (TOS) byte can be set on outgoing IP packets for various protocols. The TOS byte is used by the network to provide some level of Quality of Service (QoS) even if the network is congested with other traffic.

**Type of Service ❓**

| | |
|---|---|
| TOS for Signaling packets: | CS3 |
| TOS for RTP audio packets: | ef |
| TOS for RTP video packets: | AF41 |
| COS Priority for Signaling packets: | 3 |
| COS Priority for RTP audio packets: | 5 |
| COS Priority for RTP video packets: | 4 |

# Analog Settings

Global Analog Settings are used for configuring the IPPBX system to seamlessly work with the telephone lines from your telecommunications providers.



- **Caller ID Signaling**: The signaling type applied on the PSTN lines to pass caller ID.
  - ■ **Bell-US**—Also known as Bellcore FSK. Used in the Canada, China, Hong Kong and US.
  - ■ **DTMF**—Dual Tone Multi-Frequency. Used in Denmark, Finland and Sweden.
  - ■ **V23**—Mostly used in UK.
  - ■ **V23-Japan**—Mostly used in Japan.
- **Caller ID Start**: Defines when the caller ID starts.
  - ■ **Ring**—Caller ID starts when a ring is received.
  - ■ **Polarity**—Caller ID starts when polarity reversal starts.
  - ■ **Polarity(India)**—Can be used in India.
  - ■ **Before Ring**—Caller ID starts before a ring received.
- **CID Buffer Length**: The buffer length can be used to store caller ID info.
- **Ring Debounce**: Sets the minimum time in milliseconds to debounce extraneous ring events.
- **DTMF Hits Begin**: Sampling matching value of DTMF caller ID digits, you can choose 1 to 5 digits been matched then to consider it as part of the Caller ID.
- **DTMF Misses End**: Sample matching value of DTMF caller ID digits, you can choose 1 to 5 digits been mismatched then to consider it's not part of the caller ID.
- **Detect Caller ID After**: Sets the IPPBX to detect Caller ID after how many rings been detected.
- **Opermode**: Set the Opermode for FXO Ports.
- **ToneZone**: Select the tone zone of your country.
- **Send Caller ID After**: Certain 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).
- **Denoise**: The denoise parameter will help on noise reduction of the noisy analog lines, especially when gains have been increased on the lines.
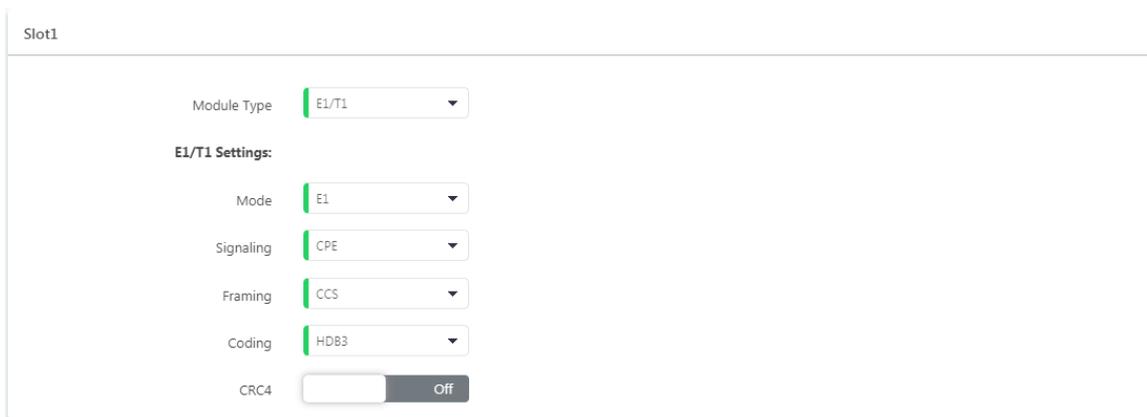
# Module Settings

Path: *Telephony -> Preferences -> Module Settings*

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/WCDMA on both telephony module slots. So if you don't have E1 and BRI modules installed then you don't have to configure module settings.

## E1 PRI Signaling

E1 module can be installed on both Slot 1 and Slot 2. To ensure IPX-2200/IPX-2500 IPPBX can detect and drive E1 module in the Module Type field, you should choose "E1/T1".



- **Mode**: Sets the module to work as E1 mode.
- **Signaling**: Sets the module to work with PRI CPE or NET, CPE is used on the client side, NET is used on the telephony provider side.
- **Framing**: By default CPE and NET use CSS (Common Channel Signaling).
- **Coding**: By default HDB3.
- **CRC4**: A method of checking for errors in transmitted data on E-1 trunk lines. Enable it only if the telephony provider implemented CRC4 on their E1 lines.

These configuration parameters should be given by the telephony provider, please configure these parameters correctly according to what they give to match the switching equipment being used by the telephony provider.

Once the configurations had been done, save and reboot the IPPBX system. In the meantime you attach the E1 line to the E1 interface. After rebooting you should get LED indications with L1 red, L2 red, L3 off and L4 green of a successful PRI CPE connection. For more details of the LED indications please check PLANET IPX-2200 and IPX-2500 chapter in the LED Indication section.

If in the deployment you got someone else connection status you should check with the telephony provider to confirm the configuration parameters. Or check with them if the line had been activated by them and ready for phone calls. If you need any help from PLANET, please contact PLANET Support for help.

## T1 PRI Signaling

To configure PLANET E1 telephony module to work in T1 mode, please choose **T1** in the **Mode** dropdown list. And then configure T1 related parameters given by the telephony provider.
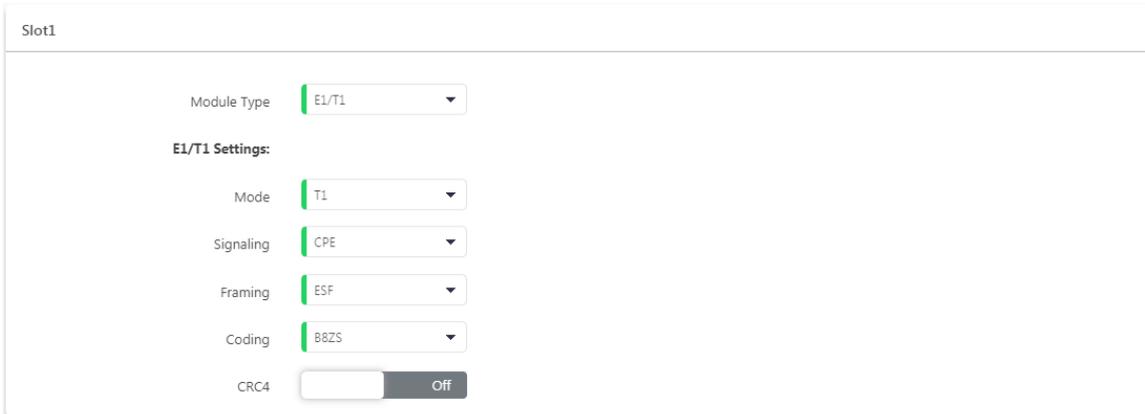


T1 runs on same signaling types as E1 mode. And T1 uses different Framing and Coding methods, configure these parameters according to the details provided by the telephony provider. In most cases CRC4 is not needed for T1 circuit, enable it only when the provider requires it.

After configurations been done, save and reboot the IPPBX system. In the meantime you attach the T1 line to the T1 interface. After rebooting you should get LED indication with L1 red, L2 red, L3 off, L4 green to indication PRI CPE signaling. For more details of the LED indications please check IPX-2200 and IPX-2500 chapter in the LED Indication section.

## MFC/R2 Signaling

In the E1 settings section and Signaling field by selecting R2 you are able to configure E1 R2 signaling.
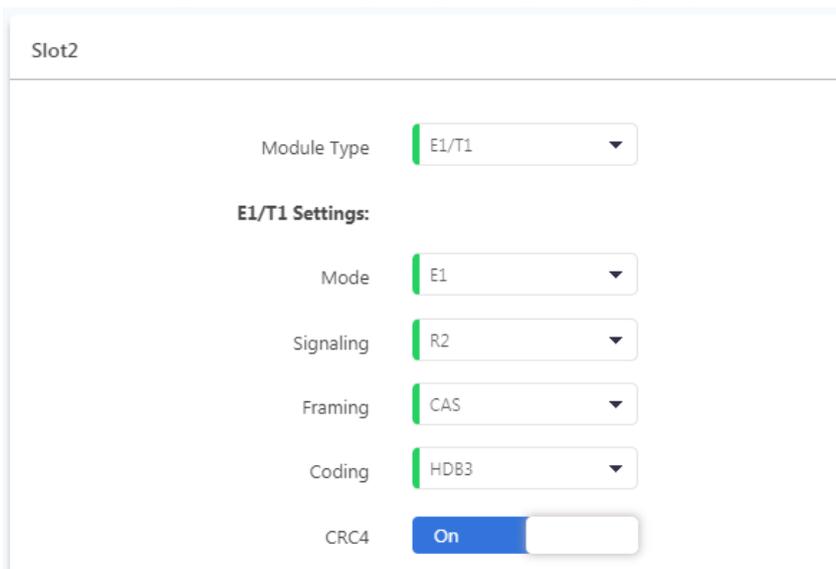


In **Signaling** field select R2, **Framing** and **Coding** should use default value. Below in the R2 Signaling Settings section set the R2 parameters.

- **Variant**: Protocol variant setting depends on country and carries.
- **Max ANI**: The maximum expected number of ANI digits.
- **Max DNIS**: The expected number of dialed digits.
- **Get ANI First**: The usual behavior for incoming calls is to get the calling party category and the ANI as soon as possible, and to get the DNIS afterwards. This doesn't work on all systems, so the option to reverse this behavior is provided.
- **Advanced Protocol File**: Additional configurations for R2 signaling.
- **Category**: Send calling party's category. Usually National Subscriber works just fine, you can set other options if needed in real application.

## SS7 Signaling

Signaling System No.7 (SS7) is a set of telephony protocols that can be delivered via E1 and T1. In the E1 settings section and Signaling field by selecting SS7, you are able to configure E1 SS7 signaling.



From the **Signaling** dropdown list, you should select SS7., **Framing** and **Coding** should be used as default value. Below in the **SS7 Settings** section set the detailed SS7 parameters.

**SS7 Settings:**

| | |
|---|---|
| Variant | ITU |
| Point Code | 20 |
| Point Code of Node Adjacent | 20 |
| ss7 dchan | 16 |
| Signaling Link Code | 0 |
| Default Destination Point Code | 20 |
| Network Indicator | International |
| Called Nai | National |
| Calling Nai | National |
| International Prefix | |
| National Prefix | |
| Subscriber Prefix | |
| Unknown Prefix | |

Please configure these parameters according to the instructions of the service provider or ask for advice from our support team. Otherwise, please do not change these settings without professional guidance.

## BRI Settings

BRI module can only be installed on Slot 2 of PLANET IPX-2200 and IPX-2500 IPPBX systems. The BRI modules you have received the jumpers on the module should be set on V2 position by default, please refer to following pictures to check the jumpers before installing.

The jumpers shown in the picture below are for crossover cables and straight cables to connect a BRI connection.



The jumpers for Port 1, 2 and Port 3, 4 are on different position. Attach the BRI lines to these ports and then configure BRI parameters.



**Type of Port** could be set to **TE_PTP**, **TE_PTMP**, **NT_PTP** and **NT_PTMP**.

- **TE_PTP**: BRI PTP Point to Point signaling (CPE side)
- **TE_PTMP**: BRI PTMP Point to Multi-Point signaling (CPE side)
- **NT_PTP**: BRI PTP Point to Point signaling (Network side)
- **NT_PTMP**: BRI PTMP Point to Multi-Point signaling (Network side)

For other parameters please configure them according to the instructions of the service provider or ask for advice from our support team. Otherwise please do not change these settings without professional guidance.
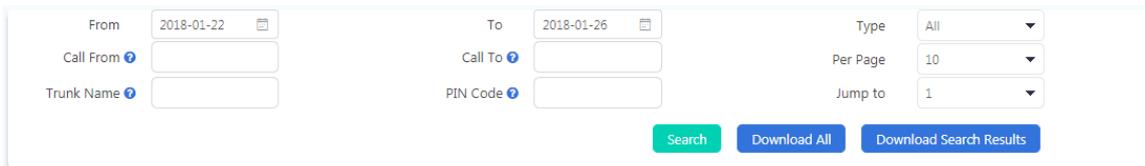
# Reports

## Call Logs

Path: **Reports -> Call Log**

Call logs are also known as CDR (Call Detailed Records), on the call logs page you can check records for any call that went through the IPPBX system.

To query call logs, you need to first specify the searching criteria.

- In the **Type** dropdown list choose Outbound, Inbound or Internal to search outbound calls, inbound call or internal calls only.
- In **From** and **To** fields set the start and end date to search call logs within this period of time.
- By specifying the name of a trunk in the **Trunk Name** field to search the inbound or outbound calls came in or sent out through this specific trunk only.
- In the **PIN Code** field specify a PIN code of a PIN Set to search outbound calls made by using this PIN code.

The searching results might look like below.

| Start Time | Call From | Call To | Final Callee | Duration | Trunk Name | PIN Code | Type | Status | Call Recordings |
|---|---|---|---|---|---|---|---|---|---|
| 2018-02-11 15:17:46 | John Doe["John Doe" <10... | 965302387 | 965302387 | 00:00:30 | FXO-1 | 9031 | Outbound | Answered | Yes |
| 2018-02-11 15:15:14 | 65302387 | IVR(biz-en) | 100 | 00:00:47 | FXO-1 | | Inbound | Answered | Yes |

- The time when this call took place will be listed in the **Start Time** column.
- In the **Call From** column lists the original caller of the calls.
- In the **Call To** column lists the first callee but might not be the last.
- The **Final Callee** column lists the extension/destination where the call finally ends.
- In the **Duration** column lists the call duration of each phone calls, this might not be the exact talk time, as when calling though the FXO ports, IPPBX system will auto answer the inbound calls so IVR works, and it will auto answer the outbound calls, so the IPPBX could send the numbers out through the PSTN lines.
- In the **Trunk Name** column lists the trunks used by those phone calls. Internal call will not take any trunk so this blank will be blank for internal calls.
- In the **PIN Code** column, only those outbound calls made out with a PIN code will list the PIN code used here. This is a good idea to tell which user/users made the call, as the PIN codes are not shared by every extension user. Every extension may have a PIN different than others or several extension users share a PIN code that is different than others.
- In the **Type** column it indicates the type of each phone call, inbound, outbound or internal.
- In the **Status** column you could tell if the calls are successfully made or failed for any reason.
- The **Call Recording** column indicates which calls are recorded and which ones are not.

You may download the search results as a CSV file by clicking on the **Download Search Results** button, or you may download all the call logs by clicking on **Download All**.

# Call Recordings

## Call Recordings

Path: **Reports -> Call Recordings -> Call Recordings**

Call recordings to be checked here are for those extensions which had enabled call recording from the extension edit page.

Search criteria can be used to search call recordings are as follows.

| | | | | | |
|---|---|---|---|---|---|
| From | 2018-01-11 📅 | To | 2018-02-11 📅 | Type | All ▼ |
| Call From ❓ | | Call To ❓ | | Per Page | 10 ▼ |
| Trunk Name ❓ | | PIN Code ❓ | | | Search |

● **Type** could be used to search the call recording according to outbound, inbound and internal calls.

● **Call From** could be used to search according to a specific caller's number (optional).

● **Call To** could be used to search according to a specific callee's number (optional).

● **Trunk Name** could be used to search according to the inbound/outbound trunk's name (optional).

● **PIN Code** could be used only for those calls which are dialed out with PIN codes define in PIN Set (optional).

The searched recordings will be displayed in a list with some detailed information.

| Start Time | Call From | Call To | Final Callee | Duration(Call/Record) | Trunk Name | PIN Code | Type | Options |
|---|---|---|---|---|---|---|---|---|
| 2018-02-09 16:33:08 | 8005 | 8000 | 8000 | 00:25:19/00:25:15 | | | Internal | ▶ ⬇ 🗑 |
| 2018-02-09 16:09:18 | 8005 | 8000 | 8000 | 00:12:32/00:12:30 | | | Internal | ▶ ⬇ 🗑 |

You may playback the recording by built-in web player by clicking on the ▶ button.

User Name: **admin**

▶   1x   00:00 ▬▬▬━━━━━━━━━━━━━━━━━  00:00   25:15   🔊 ━━━━━━━━━━━●━━  ⛶
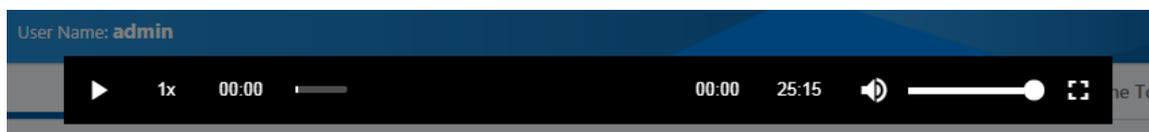
Or you may click on the ⬇ button to download or click 🗑 to delete.

Call recordings can be managed by both the admin user from admin web UI or by operator from the operator web UI.

## Conference Recordings

Path: **Reports -> Call Recordings -> Conference Recordings**

If the conferences had call recording enabled, the conference held will be recorded and conference recordings could be found for review here.

| | | | | |
|---|---|---|---|---|
| From | 2018-02-01 📅 | To | 2018-02-11 📅 | |
| Number | 90 ▼ | Per Page | 10 ▼ | Search |

From the **Number** dropdown list, you may specify the conference number and search for recordings of the specific conference number only.

The searched recordings will be listed with detailed information of when the conference calls were started, the conference number and the call/record durations. There are also the same options to play back, download and delete the recording files.

| Start Time | Conference Number | Duration(Call/Record) | Options |
|---|---|---|---|
| 2018-02-11 14:24:14 | CONFERNCE(90) | 00:06:35/00:06:33 | ▶ ⬇ 🗑 |
| 1 Total | | | |

## One Touch Recordings

Path: ***Reports -> Call Recordings -> One Touch Recordings***

One touch recording is for those extensions that are not enabled, when the user wants to record the call; by pressing 1, it will start recording.

The recordings of once touch recording could be found here. Search criteria and recording list options are the same as the "normal" call recordings, except one touch recording could not be found on the **Call Recording** page.

# System Logging

## Fax Logs

Path: ***Reports -> System Logging -> Fax Logs***

Fax logs stores all your inbound faxes and the outbound faxes sent from the extension user portal.

The fax logs could be managed by both the Admin user and the Operator user.

| From | To | Time | Type | Info | File |
|------|-----|------|------|------|------|
| 816 | | 2018-01-15 11:08:00 | Received | Success | fax000000004.tif |
| 816 | | 2018-01-15 11:01:00 | Received | Success | fax000000003.tif |
| 816 | | 2018-01-15 10:37:00 | Received | Success | fax000000002.tif |
| 816 | | 2018-01-15 09:50:00 | Received | Success | fax000000001.tif |

Start Date: 2018-01-10  End Date: 2018-02-11  Search

4 Total

By downloading the .tif file to your operating system you may view the fax details.

## Web Access Logging

Path: ***Reports -> System Logging -> Web Access Logging***

On the Web Access Logging page you may check all the logs of the web access records, including admin user, operator user and extension users.

From 2018-02-08   To 2018-02-11   Per Page 10
User ❷ All   IP ❷

Search   Download All   Download Search Results

In the **From** and **To** fields set the start and end date. From the User dropdown list, select the user role if you want to search per the type of users, optionally if you want to search according to the user's IP address you may also specify the IP address in the **IP** field and then finally click on the Search button.

The search results are shown below.

| Log Time | By User | IP Address | Options |
|----------|---------|-----------|---------|
| 2018-02-09 17:21:55 | admin | 192.168.17.225 | tcpdump network port:wan |
| 2018-02-09 17:17:32 | admin | 192.168.17.133 | change ui language to:zh_CN |
| 2018-02-09 17:17:32 | admin | 192.168.17.133 | login |
| 2018-02-09 17:17:15 | admin | 192.168.17.133 | logout |
| 2018-02-09 17:17:05 | operator | 192.168.17.133 | login |
| 2018-02-09 17:16:52 | operator | 192.168.17.133 | logout |
| 2018-02-09 17:16:43 | operator | 192.168.17.133 | login |
| 2018-02-09 17:15:43 | admin | 192.168.17.133 | change ui language to:en_US |

The time of when the login action took place, by which user, the source IP address and the actions taken will all be listed.

You may click on the Download Search Results button to download the search results in CSV format, or click on the

 to download all web UI access logs in CSV format.

## Advanced Logging

Path: ***Reports -> System Logging -> Advanced Logging***
Advanced logging can be used for higher level of the IPPBX system troubleshooting.



- **SSH Access Logging** can be used to trace the SSH login records.
- **PBX Logging** can be used to analyze the phone services related issues.
- **The OS Logging** can be used to analyze the IPPBX system OS level issues.

Enable the desired type of logging if you are qualified to analyze such kind of logs or if our support team asked for these kinds of logs for troubleshooting, otherwise please keep them disabled.

# System

## Storage

Data storage allows you to upload your recording files, log files and voicemail messages to an FTP server through the Ethernet. Or you may attach an external USB drive to the IPPBX USB interface for saving the above-mentioned files.

### USB Storage

Path: ***System -> Storage -> USB Storage***
On IPX-330 and IPX-2100 there's only 1 USB interface, on IPX-2200 and IPX-2500 there are 2 on the back panel. USB drives could be attached to the USB interface for data backup, only 1 USB drive supported on the IPX-2200 and IPX-2500.
Supported USB file system formats are: FAT16, FAT32, exFAT, NTFS, EXT3 and EXT4. If it's a portable USB hard drive, please make sure it uses external power supply. And please make sure the USB drive only has a single partition, otherwise it won't be detected by the IPPBX system.
Before attaching the USB drive and configuring data storage settings, please make sure no one else is signed in the IPPBX web UI and there's no phone call going on in the system. Because during the configuration process of USB data storage, the recordings, logs and voicemails generated would be lost.
Once a USB drive is detected, you'll see the **USB Mount Status** changed to **Connected**. Refresh the status by clicking on the ⟳ button if nothing happens when you have attached the USB drive.



Now you may enable USB storage and configure which kind of files to be saved in the USB drive.
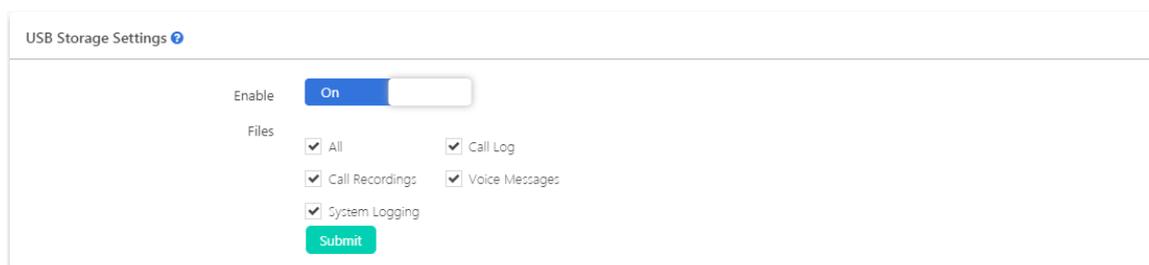


Once done click on the [Submit] button and it will start to create folders on the USB drive and upload the existing files to the USB drive too.

The whole process will take a few seconds to few minutes depending on the amount and size of the files. You may click on the ⟳ button to check the backup status. When done, the **USB Backup Status** will change to **Backup End**.



From now on the type of files you've selected will all be written directly to the USB drive. And can be browsed on the web GUI just the same as they are saved in the internal system storage.

The volume info of the USB drive could be checked every time when you first sign in to the IPPBX system web GUI (Admin and Operator user).



Please do remember that, DO NOT remove the USB drive from the IPPBX system unless you have disabled USB storage and unmounted the USB drive by clicking on the Unmount button from System -> Storage -> USB Storage page.

| | |
|---|---|
| Note | If your USB drive could not be detected by the IPPBX system, please use USB disk format tool to delete all partitions on the USB drive and create a single new partition and try again. Before doing this, please back up the data in your USB drive. It will erase all data on the drive by doing it. |

# FTP Storage

Path: *System -> Storage -> FTP Storage*

By utilizing your existing FTP server, you can configure the IPPBX system to upload 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 to. You must however ensure that your PC is always turned on or at least available at the times when your IPPBX is going to upload files.

FTP storage should not be configured to work at the same time with USB data storage. Otherwise, the data on the USB will all be migrated to your FTP server.

To configure FTP storage, enable it and configure the FTP server credentials and the file uploading options.



- In the **FTP Server Path** field you may specify the directory of where to store the uploading files.
- From the **Frequency** dropdown list select the number of days of each uploading.
- In the **Upload Time** field specify the exact time of the uploading.

Once configurations are done, click on the [Submit] button to connect the IPPBX system with the FTP server. Once connected, you'll see the **FTP Connect Status** has been changed to **Connected**.



Each time after uploading, the call recordings, voicemails and system logs will be removed from the IPPBX internal storage; call logs will be kept on the IPPBX system and also will make a duplicate on the FTP server.

# Region and Time

Path: *System -> Region and Time*

System time is very important for the IPPBX system, especially when the IPPBX system handles inbound phone calls according to time conditions, then only if the system time is correct will calls be handled properly. Also, call logs and call recordings files are named with system time. If time's not correct on the system, the phone system will not work properly.

At the initial setup while you going through the quick setup wizard your location would be set. If you have skipped the quick setup wizard or you want to change the time zone, you may do it here.
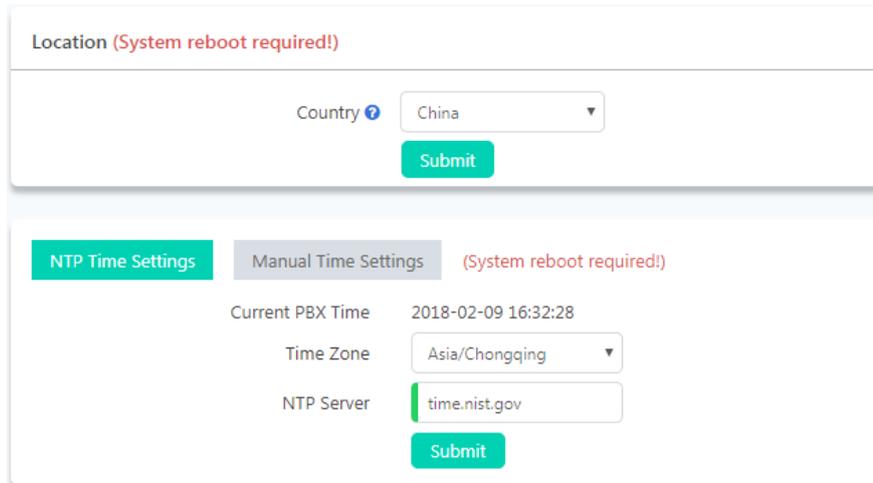
Location and time may be configured separately. Both modifying location and time settings require rebooting the IPPBX system.

The location settings will determine the type of tone (Dial tone, Busy, Congestion tone, etc.) you heard on the phones, the time zone and also the opermode on the **Analog Settings** page. So you may not change the location settings here but adjust the time settings only.

You may set the Time Zone and NTP Server to let the IPPBX system synchronize time from the NTP server. This is by default how the system time works.

Or you may manually configure the system time.
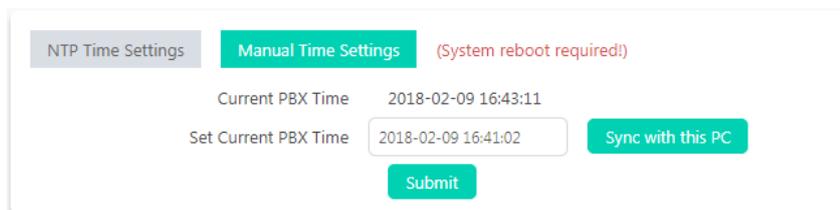
In the Set Current PBX Time blank, you may manually input the date and time to set it as the current PBX time or you may click on Sync with this PC to synchronize the current time of your operating system. Then click on the Submit button to save the manually set time to the IPPBX hardware.

# Network Settings

## Network Profiles

Path: **System -> Network Settings -> Network Profiles**
Network profiles could be configured through the quick setup wizard at the initial setup of the IPPBX system. When modification of the network profiles is required, it could be done here.



The WAN network interface of IPPBXs could be configured to work in Static IP or DHCP mode. In most cases assign a static IP would be the best practice. As all the IP phones will communicate with the IPPBX through this IP address.

On WAN port, gateway and DNS could be configured so the IPPBX could have Internet access. As a result, SIP trunking and remote extensions could work.

As for LAN, it's only used when you don't want the IPPBX system to have Internet access.



Default IP on LAN port is 192.168.0.1 -- you may change this IP but LAN IP should NOT be in the same network segment as WAN port.

# VPN

Path: *System -> Network Settings -> VPN*

VPN (Virtual Private Network) is mainly used for setting up long-distance and/or secured network connections. When used on the IPPBX system, all phone calls made and received are encrypted so it secures your remote offices/extensions' phone services. Built-in VPN Server on the IPPBX system is an easy way to set up a secured connection between other 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 firewall issues when configuring remote VoIP client such as SIP protocol which is notoriously difficult to pass through a firewall due to its random port numbers to establish connection.

IPPBX supports 2 VPN varieties, PPTP VPN and OpenVPN. On a IPPBX system you can only configure one VPN variety to work in one role. In other words, on a IPPBX system you cannot configure OpenVPN and PPTP VPN to work at the same time or to configure VPN server and client at the same time.

## OpenVPN Server

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 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 multi-client 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.

To configure OpenVPN server, please click on the OpenVPN Server button to show the configurations.
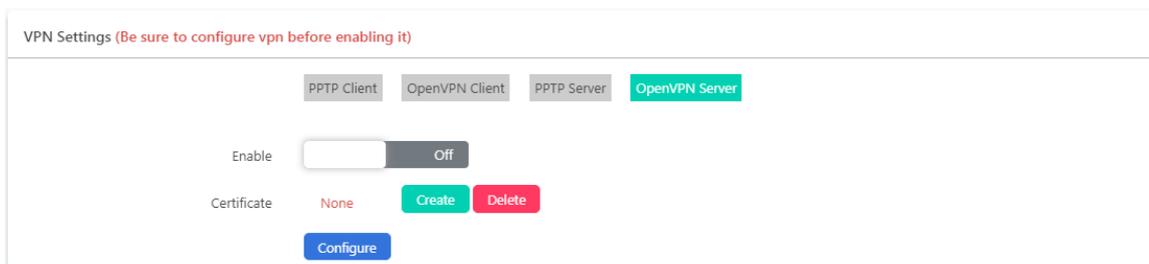


Configure the VPN server before turning it on.

In the **Certificate** field click on the Create button to create the OpenVPN certificate.

Create OpenVPN Certificate

| | |
|---|---|
| Country/Region | TW |
| Province | enm |
| City | taipei |
| Organization | planet |
| Email | planet@planet.com.tw |
| CA/KEY Expire | 3650 |

Cancel    Submit

Specify your customized information and click on the `Submit` button to continue.

Click on the `Configure` button to set up the OpenVPN server.



OpenVPN Server Settings

| | |
|---|---|
| Port ❓ | 1194 |
| Protocol ❓ | UDP |
| Device Node ❓ | TUN |
| Cipher ❓ | Default |
| Compress LZO ❓ | On |
| TLS Server ❓ | On |
| Remote Network ❓ | 172.16.0.1 / 255.255.255.0 |
| Route ❓ | 172.16.0.1 / 255.255.255.0 |
| Client to Client ❓ | On |

Cancel    Submit

● **Port**: OpenVPN service port, the default port is 1194. You will need to forward this port on your router for the clients being able to connect to the server.

● **Protocol**: You can choose either UDP or TCP. But the port forwarding (1194) on your router should be using the same protocol.

● **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 authentication and key exchange mechanism of OpenVPN.

● **Remote Network**: The OpenVPN client network, VPN server uses the first available IP of the client 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 intercommunication between clients.

Once configurations are done, click on the [Submit] button to save the configurations and you may create certificates for the OpenVPN clients now. Each VPN client needs a certificate to be able to connect to the server. OpenVPN server on IPPBX system can connect up to 20 clients.
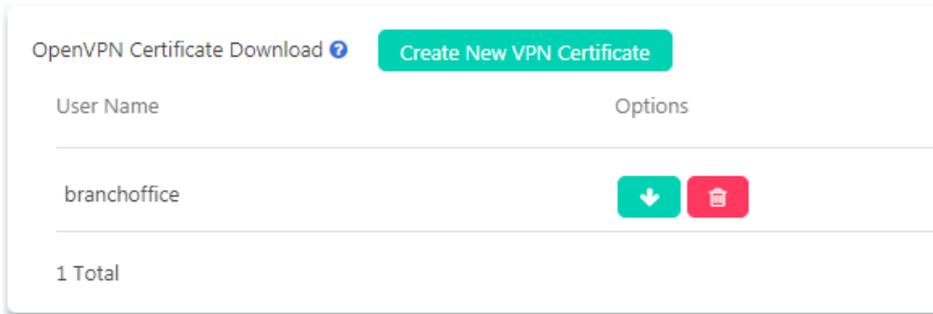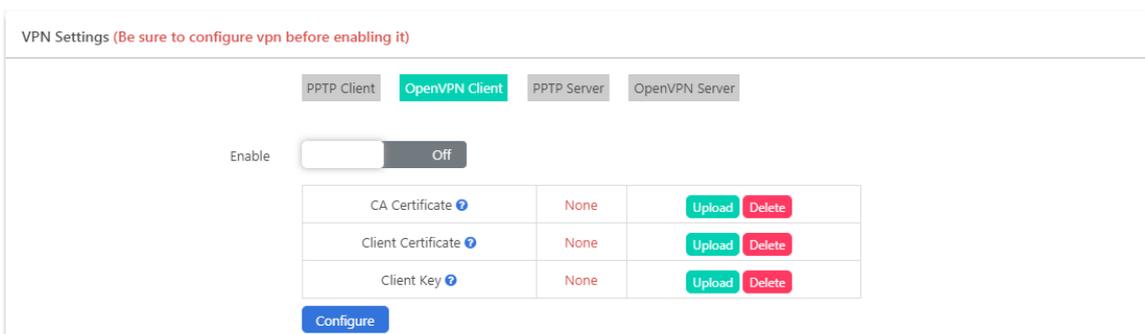


Each certificate entry created here is for an OpenVPN client. Download the certificate and extract files inside the package, 3 files you'll get and they should be uploaded on a client to be able to connect to this server.
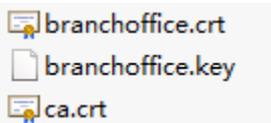


Finally turn on the enable switch to enable OpenVPN server.

### OpenVPN Client

To configure OpenVPN client, please click on the [OpenVPN Client] button to show the configurations.



The certificate files downloaded from the OpenVPN server should be uploaded here.
In the **CA Certificate** field upload the ca.crt file.
In the **Client Certificate** field upload the xxxx.crt file.
In the **Client Key** field upload the xxxx.key file.
When done, click on the [Configure] button to configure the OpenVPN client to connect to the server.

- In the **Server Address** field you should specify the OpenVPN server address, which can be a public IP or a domain name.
- The **Port** number should be exactly the same as on the OpenVPN server. By default it's 1194.
- The transport **Protocol** should be exactly the same as on the OpenVPN server. By default UDP is used.
- **Device Node** could be set to TUN or TAP, a TAP device is a virtual Ethernet adapter, while a TUN device is a virtual point-to-point IP link.
- **Cipher** (or Cypher) is an algorithm for performing encryption or decryption.
- Either to enable **Compress LZO** or not, depends on if you have enabled it on the server.
- If **Default Gateway** enabled, it will use VPN connection as default gateway, data which should be sent to the default gateway will now be sent though VPN connection.

Once done, click on submit to save the configurations. Finally click on Enable switch to switch on the VPN client connection.



And you may check the VPN connection status in the **VPN Client Status** section.



In the VPN client status section the VPN client IP, the VPN type and the connection status will be displayed.

## PPTP VPN Server

PPTP (The Point-to-Point Tunneling Protocol) 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.
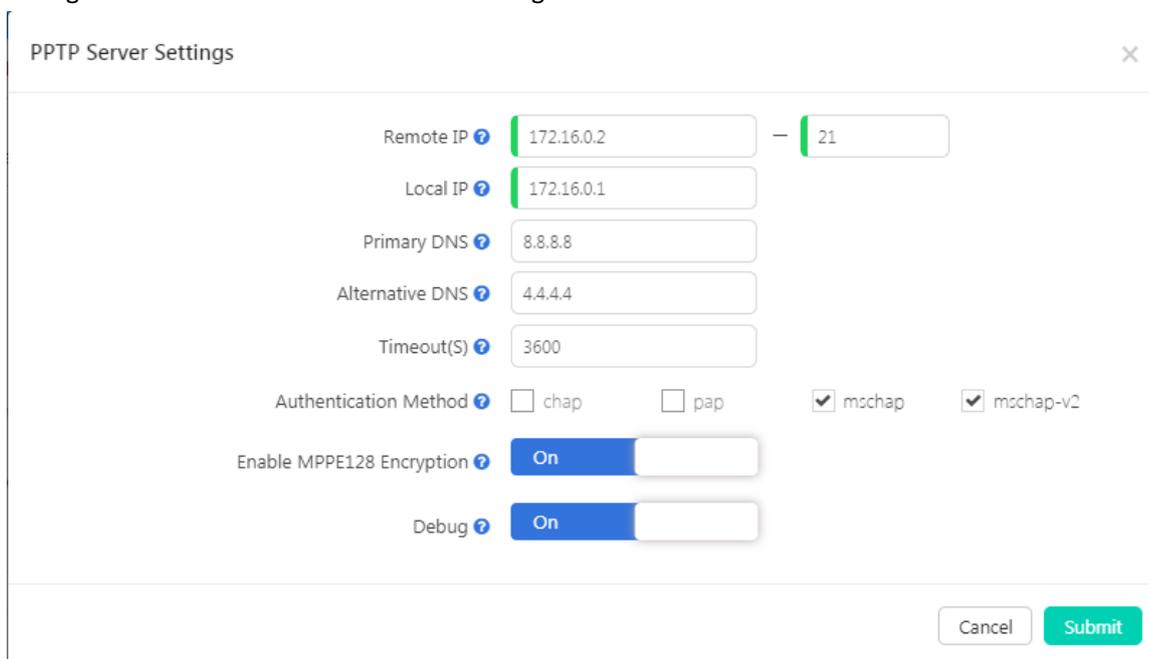
Click on the  PPTP Server  button to show the configurations.
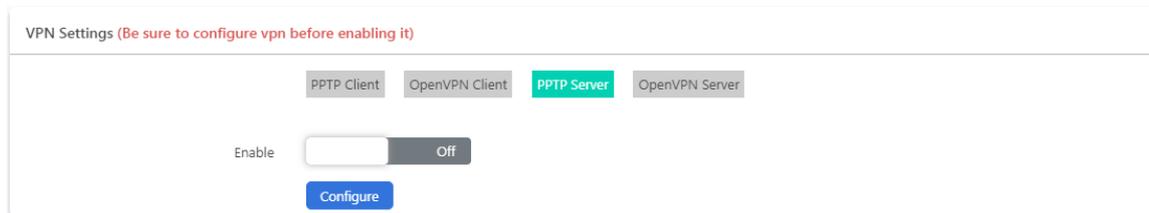


Configure the PPTP VPN server before enabling it.



- **Remote IP**: PPTP VPN remote network IP range, there must be 10 or less available IP addresses between start IP and end IP.
- **Local IP**: PPTP VPN local server IP address.
- **Primary DNS**: Primary DNS for VPN connection.
- **Alternative DNS**: Secondary DNS for VPN connection.
- **Timeout(S)**: Session timeout for PPTP tunnels.
- **Authentication Method**: Choose method/methods for the authentication of the VPN clients.
    - **chap**: Challenge Handshake Authentication Protocol, CHAP takes a more sophisticated and secure approach to authentication by creating a unique challenge phrase (a randomly generated string) for each authentication.
    - **pap**: Password Authenticate Protocol PAP works like a standard login procedure; it uses static user name and password to authenticate the remote system.
    - **mschap**: MS-CHAP is the Microsoft version of the Challenge-Handshake Authentication Protocol.
    - **mschap-v2**: Microsoft Challenge Handshake Authentication Protocol version 2 (MS-CHAP v2), this provides stronger security for remote access connections.

- ● **Enable MPPE128 Encryption**: 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**: Enable debug for PPTP VPN connection, debug information will be written into system logs.

Once server configurations are done, you may create PPTP client users, each user created is for a VPN client to connect. PPTP VPN server on IPPBX can connect up to 20 PPTP VPN clients.
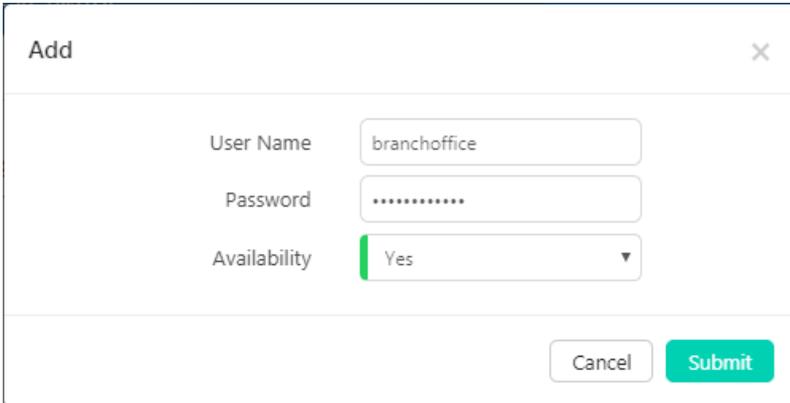


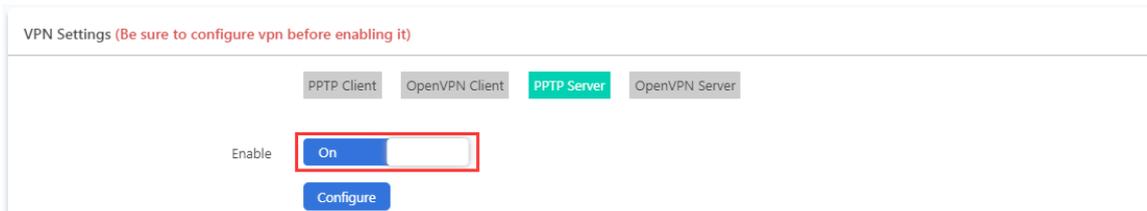Remember to set the Availability to Yes, when you don't want this user to connect, just set Availability to No or you may remove the user from the VPN user list.
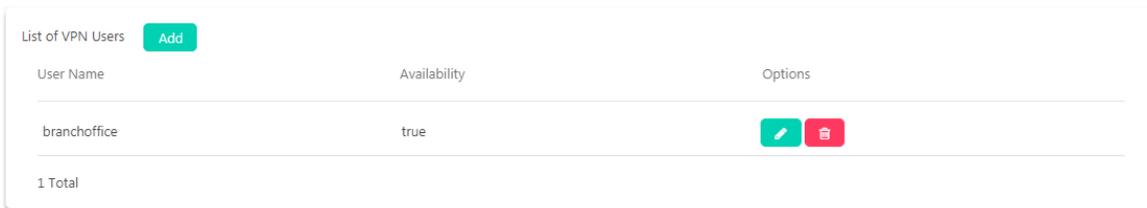


Finally click on the Enable switch to turn the PPTP VPN server on.



## PPTP VPN Client

To configure PPTP VPN client, please click on the PPTP Client button to show the configurations.
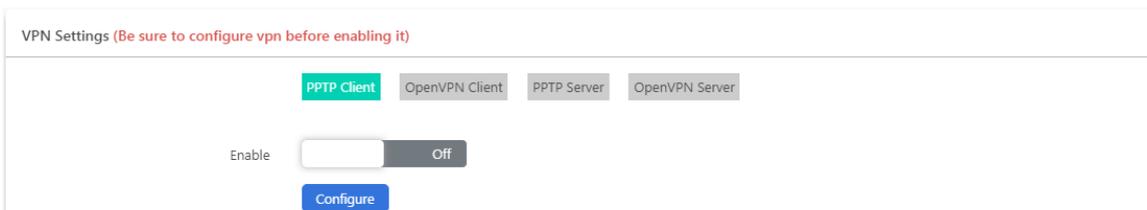


Configure PPTP VPN client settings before enabling it.

- **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**: If enabled, all network traffic will go through the PPTP VPN connection.

Once done, click on the [Submit] button to continue, and now you may click on Enable switch to turn on PPTP VPN client.



Later it should be connected to the PPTP VPN server, and the connection status will be displayed in the **VPN Client Status** section.



In the VPN client status section the VPN client IP, the VPN type and the connection status will be displayed.

## Static Routing

Path: **System -> Network Settings -> Static Routing**

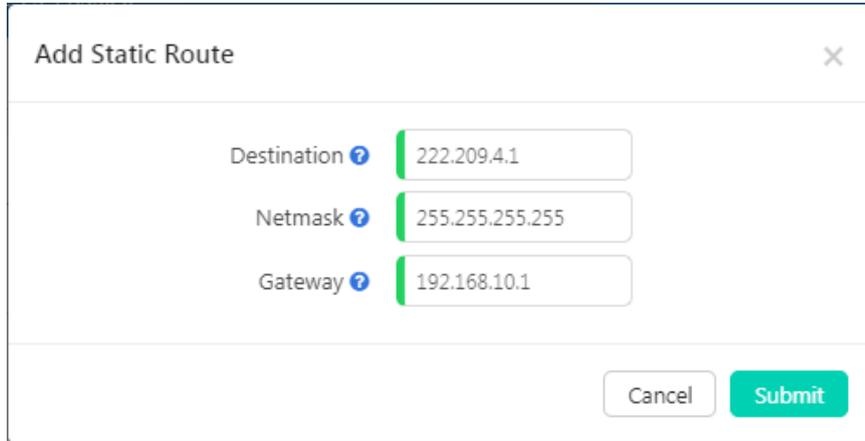Static Routing is a form of routing that occurs when a router uses a manually-configured routing entry, rather than information from a dynamic routing protocol to forward traffic.

When needed you may click on the [Add] button to add a manual static route.



- **Destination** is the IP address of the destination host or network address.
- If the packets are to be sent to the **Destination** specified above then send them to the **Gateway** address.

After the new record has been manually created you will see it listed in the route table.

## DHCP Server

Path: **System -> Network Settings -> 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 IPPBXs WAN/LAN port which saves administrators a lot of time when compared with having to configure these settings manually.

Before activating DHCP services, please ensure there's no other DHCP server running in your LAN, otherwise there will be collision between servers.

Set the DHCP server network parameters and turn it on.

The DHCP clients which obtained IP addresses from the IPPBX system DHCP server will be listed on the right side of the page, in the **DHCP Clients** section.

If you want some host or client to always get the same IP address, **IP Address Reservation** will help.



Just simply specify the MAC address of the client device and associate an IP address with it, and this IP will always be reserved for this specific client device.

## DDNS

Path: *System -> Network Settings -> DDNS*

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 IPPBX system only requires you to sign up with a Dynamic DNS provider, then with the domain name they provide which maps your IP address on the Internet, you can access IPPBX and also other
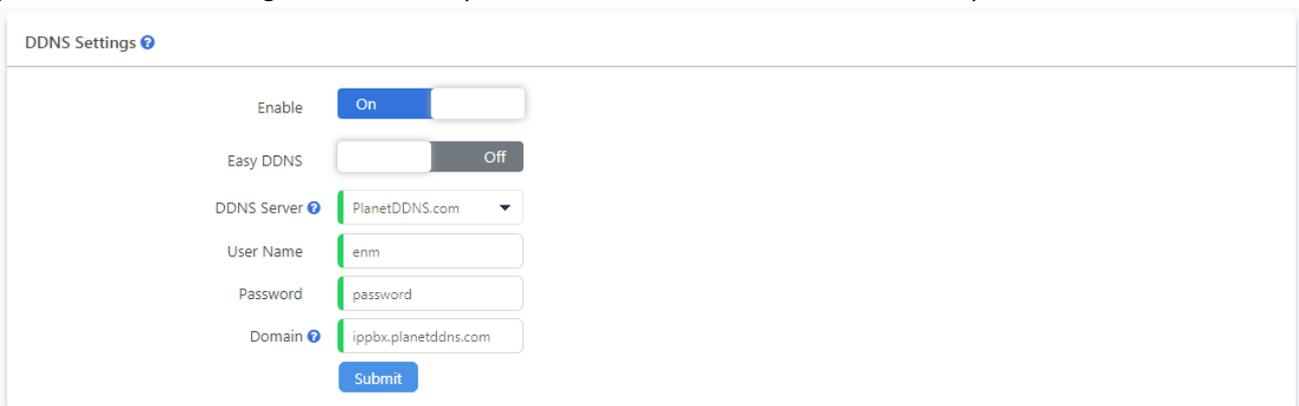
services within your LAN via the domain name without needing to know your Dynamic public IP Address.

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

IPPBX system supports the following DDNS service providers:

- Easy DDNS
- PlanetDDNS.com/
- http://dyndns.org/
- http://www.noip.com/
- http://www.zoneedit.com/
- http://www.oray.com/
- http://www.3322.net
- http://freedns.afraid.org/

Sign up to one of these DDNS service providers' website and subscribe a dynamic domain name. Once you have your account details, register it here and you DDNS domain will work with the IPPBX system.



After completing the above, please configure port forwarding on your router/firewall, then you'll be able to remote access IPPBX services from the internet using this dynamic domain. For example, you can port forward port number 443 and then you can access the IPPBX web interface by using the URL:

http://www.planetddns.com/

# Security Center

IPPBX system has been preconfigured with a built-in firewall which prevents your IP phone system from unauthorized access, malicious users and some other attackers.

You may not need to specifically configure the firewall settings but for security precautions please always keep it on.

## Firewall

Path: **System -> Security -> Firewall**

IPPBX system uses Fail2Ban to perform intrusion detection and uses iptables to block any attack attempts.

- First of all make sure the **Firewall** option is enabled. Only consider disabling your firewall if your PLNAET IPPBX is behind a router/firewall without any port forwarding from the Internet.
- **Drop Ping** will cause the system to ignore ping request. If enabled, you cannot ping the IPPBX system.
- **Drop All** will cause all packets sent to the IPPBX system being dropped, this will cause IPPBX system to block all communication with the outside world. Except web UI still works in local network, other services will all be terminated.
- **Geo IP** is a security policy which can be used to grant access of IP addresses from certain countries/regions, all IP addresses from other countries/regions which are not specified will all be blocked. By default, web UI will still be accessible. Enabling **Geo IP** requires **Drop All** to be enabled too.

   To implement Geo IP please follow the steps below.

   **Step 1**: Enable Geo IP and Drop All
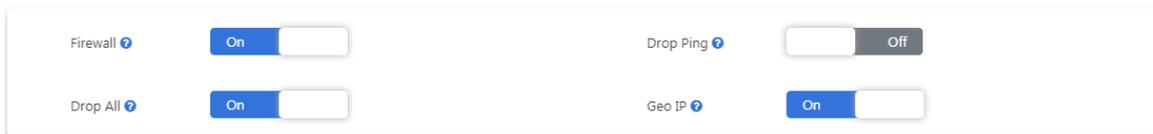


   **Step 2**: Select trusted countries/regions



   Besides selecting the trusted IP addresses from certain countries/regions, you'll still need to add a common rule in the **Common Rules** section to grant access or the local network hosts/devices.

   **Step 3**: Add a common rule to grant access of your local LAN.
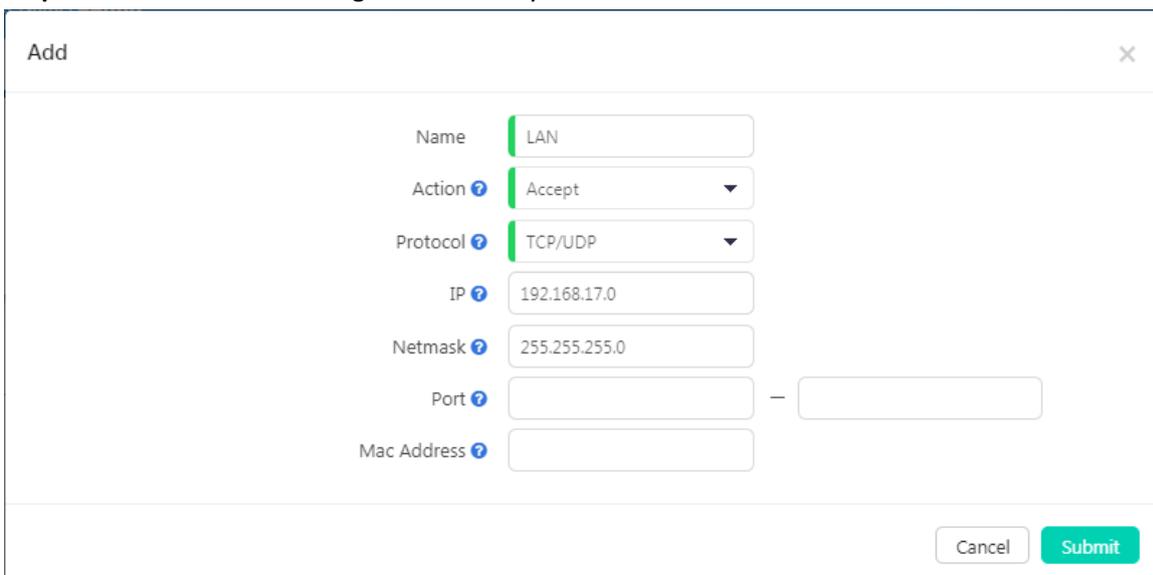


- The **Action** of this rule needs to be set as **Accept**.
- **Protocol** should be set as **TCP/UDP**.

- **IP** should be the local network address instead of a single IP address.
- **Netmask** should be the subnet mask of the network address.
- The **Port** range determines which kind of services to be granted. In this case you may leave it blank to grant local network all access to the IPPBX system.
- **Mac Address** determines the action to be taken according to the Mac address of a device instead of its IP address, it only works with devices within the same local network because Mac addresses are not routable. In this case you are going to grant access of all the local network hosts/devices, so you may leave it blank.

By now, Geo IP security policy should work. The private IP addresses from your local network and the public IP addresses from the countries/regions you've selected should be able to access your IPPBX system. Other IP addresses will all be blocked.

**Common Rules** can be used to 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. The priority from high to low of the firewall rules is from the top of the list to the bottom.

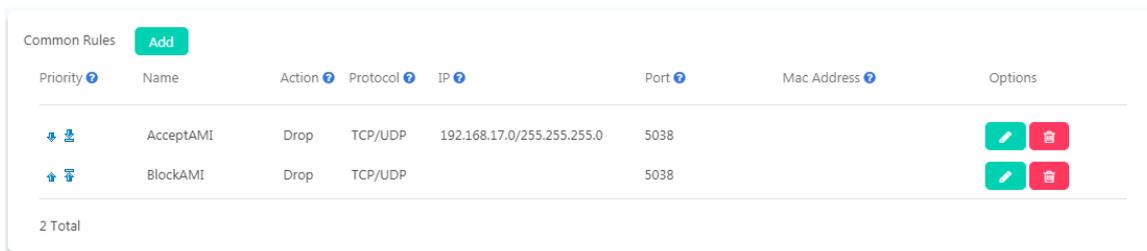If you are going to grant access of some kind of services to specific IP address or network, add the grant rule/rules first then add the deny rules. If the order of the rules is not correct you may use the arrows in the **Priority** column to adjust the order of the rules.

| Common Rules | Add | | | | | | |
|---|---|---|---|---|---|---|---|
| Priority ❓ | Name | Action ❓ | Protocol ❓ | IP ❓ | Port ❓ | Mac Address ❓ | Options |
| ⬇ ⬓ | AcceptAMI | Drop | TCP/UDP | 192.168.17.0/255.255.255.0 | 5038 | | ✏ 🗑 |
| ⬆ ⬓ | BlockAMI | Drop | TCP/UDP | | 5038 | | ✏ 🗑 |
| 2 Total | | | | | | | |

In the above given example, the 2 rules "AcceptAMI" and "BlockAMI" limited that only the IP addresses from network 192.168.17.0 can have AMI access. Except IP from this network others will all be denied to access. In this case, if the "AcceptAMI" rule is moved beneath the "BlockAMI" rule, then the AMI port will be totally lockdown, no one can access it.

| Note | If you are going to add rules to block some IP addresses from accessing some kind of services on the IPPBX system, be sure you add the correct IP/network address (if not defined, the firewall will consider as ALL), and the correct service port number (if not defined, the firewall will consider as ALL), otherwise misconfiguration of a deny rule might cause the IPPBX system total lockdown, only way would be using Console (IPX-330 and IPX-2100) or HDMI (IPX-2200 and IPX-2500) to unlock the IPPBX from command lines. |
|---|---|

**Auto Defense** will help with the prevention of DDOS attacks.

You may specify the service port number and the maximum packets to be accepted on this port number in a certain time interval. Except the specified number of packets, more packets sent within the time interval will be dropped by the IPPBX system.

## Intrusion Detection and Prevention

Path: ***System -> Security Center -> IP Blacklist***

IPPBX system uses Fail2Ban to perform intrusion detection. 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.

There are 4 default intrusion detection and prevention rules to secure SIP, IAX2, Web and SSH services on your IPPBX system. And by default all of them are activated to keep your IPPBX system safe.



Each of the intrusion detection and prevention rule is configured with a maximum **Illegal Attempts** and the **Observation** time duration, once the **Illegal Attempts** reached the given value in the given **Observation** time duration, the source IP address of where the illegal attempts coming from will be banned by the firewall for the given time duration specified in Ban for field. Banned IP will be listed on the **IP Blacklist** page.

Besides the 4 default rules, if you want to add more rules you can do it on the **Firewall** page **Auto Defense** section.

## IP Blacklist

Path: *System -> Security Center -> IP Blacklist*

IP Blacklist will list all suspected intruders/attackers' IP addresses. The list is automatically generated by the system firewall if possible intrusion/attacking had been detected. And the list will show the IP address of the banned hosts, as well as what kind of service intrusion was detected.

| Type | IP | Options |
|---|---|---|
| HTTPS | 117.136.70.48 | 🗑 |
| 1 Total | | |

If an IP address appears incorrectly in the list of rejected IP, you can click on the 🗑 button to remove it from the IP blacklist.

## IP Whitelist

Path: *System -> Security Center -> IP Whitelist*

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



By adding a trusted IP to the IP whitelist, you may also define which kind of services it could access.
- **SIP** allows the IP to be able to register SIP extensions.
- **IAX** (IAX2) allows the IP to be able to register IAX extensions.
- **HTTPS** allows the IP to access the web UI of the IPPBX system.
- **SSH** allows the IP to access the IPPBX system command lines through SSH.

| | You'll only need to add trusted IP addresses to the IP Whitelist when you have configured Drop All or Geo IP security policies. And in the policies these IP addresses are not included as trusted IP addresses. Otherwise you don't have to add them to the IP whitelist. |
|---|---|
| Note | |

# Email Services

## Mail Server Settings

Path: ***System -> Email Services -> Mail Server Settings***

Various kinds of Emails could be sent from the IPPBX system. The Emails could be automatically sent by the IPPBX system in certain circumstances or manually sent by admin and operator users.

To configure the IPPBX system being able to send out emails, mail (SMTP) server needs to be configured at first priority. At the initial system setup stage while you were going through the quick installation wizard, mail server could be configured, if you've not done it from the wizard, it still can be configured from here.

We have built-in some popular Email service providers' SMTP configuration templates for users to quickly deploy their mail server.
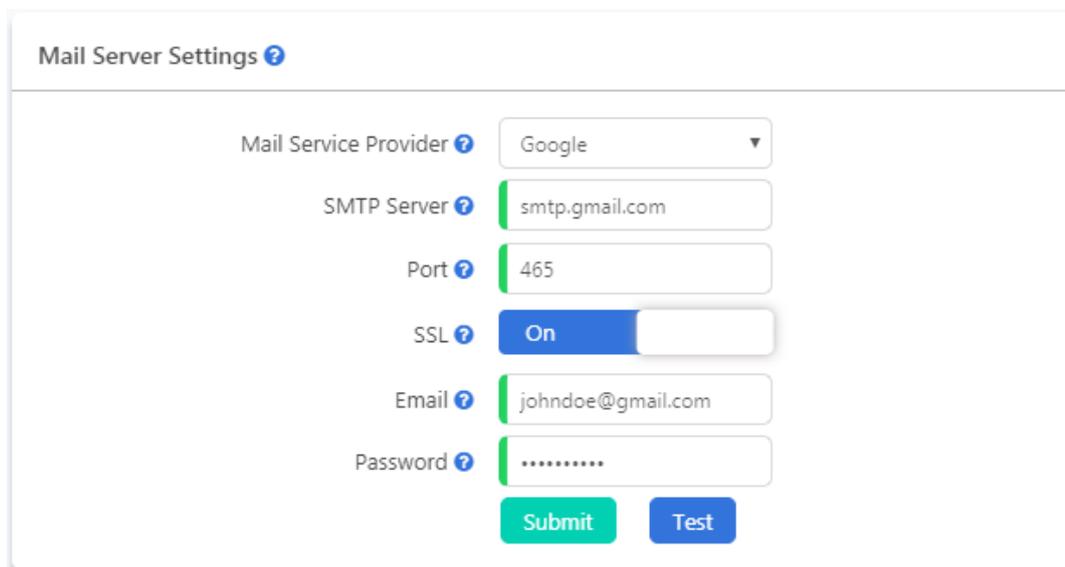


- In the **Mail Service Provider** dropdown list select your Email service provider. If it's not included here, please choose **Other**.
- Once you have selected the mail service provider the **SMTP Server** field will be auto filled. Otherwise you'll have to manually input the SMTP Server address.
- Default SMTP service **Port** is 25, but with SSL/TLS it would be 465. Otherwise you'll have to input the actual port number your mail service provider uses.
- **SSL** encrypts a communication channel between the IPPBX system and the SMTP server. Most of the mail service providers have implemented SSL support.
- In the **Email** field input the Email account to be used by the IPPBX system, all mails from the IPPBX system will be sent out by this mail account.
- In **Password** field input the password of the Email account you have specified.

Once done the above settings, click on the [Submit] button to make configurations effective. And you may click on [Test] button and input an Email address to send a test email to verify if the mail server is successfully deployed.

> **Note** You may need to activate SMTP service from your Email web portal before you can successfully configure SMTP server on the IPPBX system.

## Voicemail to Email Settings

Path: *System -> Email Services -> Voicemail to Email Settings*

Voicemail to Email is a very useful feature for the extension users, as the IPPBX system has the ability to send received new voicemail messages of their extensions to their Email box.

It could be an Email notification or administrator could set the IPPBX system to send Email with voice messages attached in the Email notifications.



- The **Mail Subject** field you can set customized Email subject which will be received by the extension users when they have new messages.
- The **Mail Body** is also customizable, you may use variables in the mail body to describe the new voice messages they got. The format of the variables must be the same as listed in the **Variables** section.
- **Variables** could be used in the mail body to indicate the extension users about their new voice message details.
- With **Deliver Message as Attachment** option enabled the voice message will be attached to the notify Email, users may playback the voice messages when they got the notify Email.

With Voicemail to Email enabled and Mail Server configured, the extension users will get Email notifications when new voice message received on their extensions, just make sure the extensions have their Email addresses specified.

## Fax to Email Settings

Path: *System -> Email Services -> Fax to Email Settings*

If you have configured the IPPBX system to send the inbound fax as Email, here on this page is where you configure the Email templates.

You may customize the fax notification mail subject and mail body. The variable ${FAX_SENDER} could be used in mail subject or mail body to describe who sent the fax.

## Email Notifications

Path: **System -> Email Services -> Email Notifications**
You may configure the IPPBX system to send Email notifications to the administrator on some system events.



- **Storage Usage Alert** can be used to notify the administrator about the system internal storage usage when reached the threshold.
- **Change Admin Password** can be used to notify the administrator about the admin password changing.
- **System Reboot** can be used to notify the administrator after a system reboot. No matter after a manual reboot or a power failure.
- **System Reset** can be used to notify the administrator before resetting the IPPBX system to factory defaults.
- **Firmware Upgrade** can be used to notify the administrator before upgrading the IPPBX system firmware.
- In the **Email** field you should specify the Email address of who's responsible to respond to these system events.

# Maintenance

## Users

There are 3 user roles could be used to manage the IPPBXs.
- Admin

Admin user is mainly used for IPPBX system administrations from the IPPBX web interface, including all system advanced configurations and management.
- Operator

Operator user is mainly used by the company receptionist, secretary or some other employee who's responsible for managing the departments, extensions, call logs, recordings, faxes, some common call features, etc.
- Root

Root user is used by highly experienced technicians for advanced managing and troubleshooting the IPPBX system from command line level.

### Admin User

Path: *Maintenance -> Users -> Admin User*
Admin user password should be changed regularly on this page. And the password should be at least 6 characters long and contain a combination of letters, digits and symbols.

### Operator User

Path: *Maintenance -> Users -> Operator User*
Admin user could change the operator password here on this page when needed. And can turn AMI (Asterisk Management Interface, port 5038) access with operator credentials on or off.
AMI access using operator credentials can be used to achieve some CTI (Computer Telephony Integration) implementations.

### Root User

Path: *Maintenance -> Users -> Root User*
Root user password could be managed by admin user as well. When admin user modifies the root password for the first time, admin needs to provide the old (default) root password which is the last 8 characters of the IPPBX WAN port MAC address.
By default, SSH access to the IPPBX system is disabled. When enabled, user may access the IPPBX system Linux command line interface via SSH on port 22.
If you are not a highly experienced Linux user with Asterisk command line troubleshooting skills, for security precautions please DO NOT enable SSH.

# Upgrade

PLANET will update the IPPBX firmware at regular intervals for new features and bug fixes. You can visit our official website www.planet.com.tw to check the updates for your IPPBX system.

The downloaded firmware package should be in .rar or .zip format, please extract the package first and upgrade with the uImage-md5.xxx file to upgrade your IPPBX system.

Path: *Maintenance -> Upgrade*

The current firmware version and the last time when the firmware was upgraded will be displayed on the firmware upgrade screen.

| Firmware Version | |
|---|---|
| Current Firmware Version | v3.0.0 |
| Last Upgrade ❷ | 2017-11-09/15:24:00 |

Before upgrading, you may refer to the information and consider if a new firmware upgrading is necessary.

To upgrade the firmware please click on the [⬆ Select File] button to locate the uImage-md5.xxx file, the file extension ".xxx" should match your IPPBX model, otherwise the upgrade will fail.

After the file is located, click on the [Submit] button to upgrade. For IPX-330 and IPX-2100, the whole process will take 4 to 5 minutes, for IPX-2200 and IPX-2500, the whole process will take 2 to 3 minutes, when done please refresh the web page to re-login.

| | |
|---|---|
| **Note** | Firmware upgrade will cause the IPPBX system to restart and as a result all the phone calls going through the system will be terminated. So please make sure there're no phone calls going on before you upgrade. |

# Diagnostic

## PING

Path: *Maintenance -> Diagnostic -> 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.

First specify the domain or IP of the host you want to contact then click on the [Start] button and then the command begins to process. You will receive results output from the system indicating the reachability of the destination.

## Traceroute

The traceroute command is used to discover the routes that the packets actually took while traveling to their destination.

Path: *Maintenance -> Diagnostic -> Traceroute*

In the IP Address/Domain Name field specify the IP or domain name that you want to lookup and click on [Start] button to begin tracing.



During the whole process each step will output in the Results field, you can view which routes the packets have taken before reaching their final destination.

## Ethernet Capture

Ethernet capture uses TCPDUMP which is a common pocket analyzer allows users to capture TCP/IP and other packets being transmitted or received over a network to which the 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.

Path: *Maintenance -> Diagnostic -> Ethernet Capture*

To capture the network traffic, you need to select the network interface according to on which the IPPBX system is working on. Then click on the [Start] button to start capturing the network traffic.
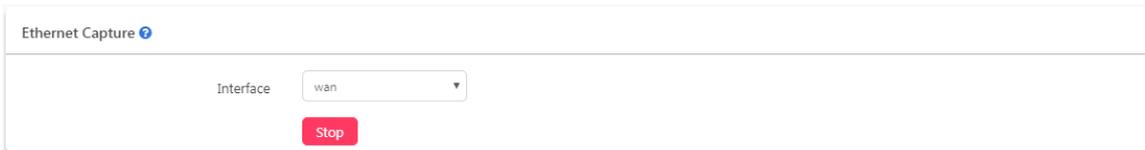
Once the process begins, the Start button will change to Stop. At this moment, you should make a call to recur the phone call problem or ensure some other problem had recurred, so the captured network traffic could content errors that are helpful for troubleshooting. Once done click on `Stop` button, and the captured network traffic will be automatically downloaded.

The downloaded file could be analyzed by Wireshark or you could send the file to PLANET support team for help.

## Channel Monitor

Channel Monitor, technically DAHDI Monitor 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 sounds editor to visual 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. If needed, you can send this file to PLANET support for help.

Path: *Maintenance -> Diagnostic -> Channel Monitor*

Before starting channel monitor, you need to select an FXO interface. Then click on `Start` button to capture signaling on the selected interface.



Once the process started the button will change to Stop. Now you should recur the problem by making a call in through the selected interface. When the extension started to ring the third time you may hangup and stop channel monitor by clicking on `Stop` button. As soon as the channel monitor stopped, the captured signaling will be automatically downloaded.

If you have knowledge of how to analyze the files you may open them with some sound editors like Wavepad, or you may send the file to PLANET support team for help.

# Backup

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

Normally the first backup should be taken when you have finished configuring the IPPBX to work for the very first time. Also, when you have applied new changes to your configuration is always a good time to take another

backup.

Path: *Maintenance -> Backup*

You may click on the [Backup] button to take a backup of your IPPBX system when necessary. A backup file will be generated.
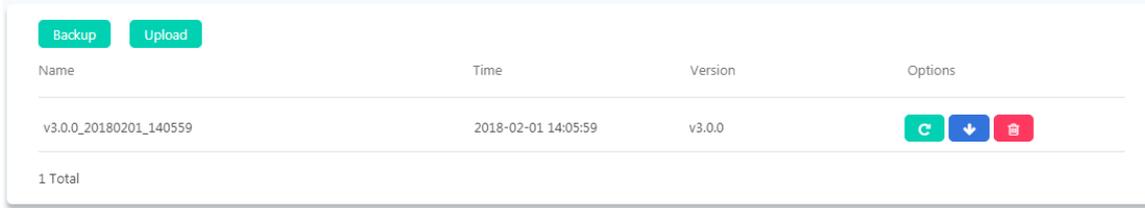
| Backup Upload | | | |
|---|---|---|---|
| Name | Time | Version | Options |
| v3.0.0_20180201_140559 | 2018-02-01 14:05:59 | v3.0.0 | C ↓ 🗑 |
| 1 Total | | | |

File name is generated according to the software version, date and exact time when the backup is performed. You may click on ↓ button to download the backup to you operating system. Or click on 🗑 button to delete it from the IPPBX system.

When you want to restore the backup, you may click on the [C] button. Restoring a backup will cause the system reboot, so please make sure there are no phone calls going on in the IPPBX system before you doing this.

If you are going to restore an offline backup (backup downloaded to your operating system) please click on the [Upload] button.

| | Backups will not be cleared after a system reset. So you may not need to download the backup to your operating system. And after a system reset, you may skip the quick setup wizard and go to the backup |
|---|---|
| Note | page to restore a backup directly to recover your previous configurations. |

# Reboot and Reset

## Reboot

Path: *Maintenance -> Reboot and Reset*

By clicking on the [Reboot] button you may restart your IPPBX from the web UI. Restart the IPPBX system will terminate all active phone calls, please make sure there're no phone calls going on before restarting the IPPBX system.

With version 3 series IPPBX, restarting each of the IPPBX model the time it takes for the services to back on is as in below chart.

| | IPX-330 | IPX-2100 | IPX-2200 | IPX-2500 |
|---|---|---|---|---|
| Phone call | 2mins | 2mins | 1.5mins | 1.5mins |
| Web UI | 4mins | 4mins | 2mins | 2mins |

## Reset

Path: *Maintenance -> Reboot and Reset*

To reset IPX-330 and IPX-2100 please refer to below 3 reset methods (**Method 1**, **Method 2** and **Method 3**).
To reset IPX-2200 and IPX-2500 there's only 1 way, please refer to **Method 1**.

**Method 1**: Reset from web UI

Resetting the IPPBX system

Click on [Reset] button and confirm with the popup window, reset process will begin. During the reset process the IPPBX system will restart and the whole process will take around 4 to 5mins for IPX-330 and IPX-2100, 2 to 3mins for IPX-2200 and IPX-2500.

Before resetting you may enable "I'd like to keep the network profiles" option, so after resetting you may still access the IPPBX system web UI from the same IP. If network profiles had been reset too, you'll need to access the IPPBX system via the default IP address.

WAN default IP: 172.16.0.1

LAN Default IP: 192.168.0.1

After resetting when you access the web UI you'll first see the quick setup wizard, if you choose to use backup file to restore the system configurations, you may skip the quick setup wizard, if you wish to configure a fresh new phone system, you may follow the wizard to complete the configurations.

| | |
|---|---|
| Note | Reset from web UI will clear all system configurations, except if you have enabled "I'd like to keep the network profiles" option which will keep the network configurations. Except configurations to be erased, user data will also be erased, including call logs, call recordings, voicemails, faxes, etc. By default backups will be kept, so after resetting from web UI you may restore backup directly from the IPPBX system. |

**Method 2**: Reset by RESET button at system running stage (IPX-330 and IPX-2100 only)

When the IPX-330/IPX-2100 IPPBX system is running, the SYS LED indicator winks once every 2 seconds. Now you may press and hold the RESET button on the back panel of the IPPBX for about 7 seconds, then the SYS LED will go off, the IPPBX system will reboot and start the reset process.

Reset IPX-330/IPX-2100 IPPBX this way is the same as resetting from the web UI. Only difference would be you cannot choose to keep the network profiles, so you will need to access the IPPBX system via the default IP.

**Method 3**: Reset by RESET button at system booting stage (IPX-330 and IPX-2100 only)

Reset the IPX-330/IPX-2100 IPPBX system by RESET button at system booting stage will erase everything on the IPPBX system, including backups will be erased as well. Resetting this way will fully recover the IPPBX system to factory defaults.

So if you wish to restore the IPPBX configurations with a previous backup, please download it to you operating system first before resetting.

To reset the IPX-330/IPX-2100 IPPBX system at system booting stage, you need to first cut off the power supply. Press and hold the RESET button then power it on. 4 to 5 seconds later when SYS LED goes off release the RESET button.

Around 5mins later access the IPPBX system via the default IP address. You'll first be directed to the quick setup wizard page, you may start configuring a fresh new phone system or you may skip and upload offline backup to restore previous configurations.