

User's Manual



Internet Telephony PBX System

▶ **IPX-330**



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This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications.

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1. Reorient or relocate the receiving antenna.
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3. Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
4. Consult the dealer or an experienced radio technician for help.

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R&TTE Compliance Statement

This equipment complies with all the requirements of DIRECTIVE 1999/5/EC OF THE EUROPEAN PARLIAMENT AND THE COUNCIL OF 9 March 1999 on radio equipment and telecommunication terminal Equipment and the mutual recognition of their conformity (R&TTE) The R&TTE Directive repeals and replaces in the directive 98/13/EEC (Telecommunications Terminal Equipment and Satellite Earth Station Equipment) as of April 8, 2000.

WEEE Caution



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.

Safety

This equipment is designed with the utmost care for the safety of those who install and use it. However, special attention must be paid to the dangers of electric shock and static electricity when working with electrical equipment. All guidelines of this and of the computer manufacture must therefore be allowed at all times to ensure the safe use of the equipment.

Customer Service

For information on customer service and support for the Gigabit SSL VPN Security Router, please refer to the PLANET website (URL: <http://www.planet.com.tw>).

Before contacting customer service, please take a moment to gather the following information:

- Internet Telephony PBX System serial number and MAC address
- Any error messages that displayed when the problem occurred
- Any software running when the problem occurred
- Steps you took to resolve the problem on your own

Revision

User's Manual for PLANET Internet Telephony PBX System

Model: IPX-330

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Chapter 1 Introduction

Intuitive, Ease-of-Use IP PBX Machine Management

PLANET IPX-330 IP PBX telephony system is SIP based and optimized for the small and medium business in daily communications. The IPX-330 is able to accept **30 user registrations**, and easy to manage a full voice over IP system with the convenience and cost advantages.

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

The IPX-330 is a feature-rich PBX system that supports seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints.



Replaces old PBX directly without any new wiring

Cost-effective, easy-to-install and simple-to-use, the IPX-330 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 IPX-330, home users and companies are able to save the installation cost and extend their past investments in telephones, conferences and speakerphones. The IPX-330 can be the bridge between traditional analog systems and IP network with an extremely affordable investment.



Distributed VoIP Network Infrastructure

For the new generation communication age, the IPX-330 supports IPv6 and VPN (client / server) connection to provide users with more flexible and advantageous communication products. With PLANET DDNS function, the IPX-330 also helps users to apply and remember the login information easier. Moreover, its multiple language features helps user to quickly and friendly manage the system.

Standard Compliance

Compliant with the Session Initiation Protocol 2.0 (RFC 3261), the IPX-330 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.

Compliant with standard SIP RFC 3261



Green IP Office

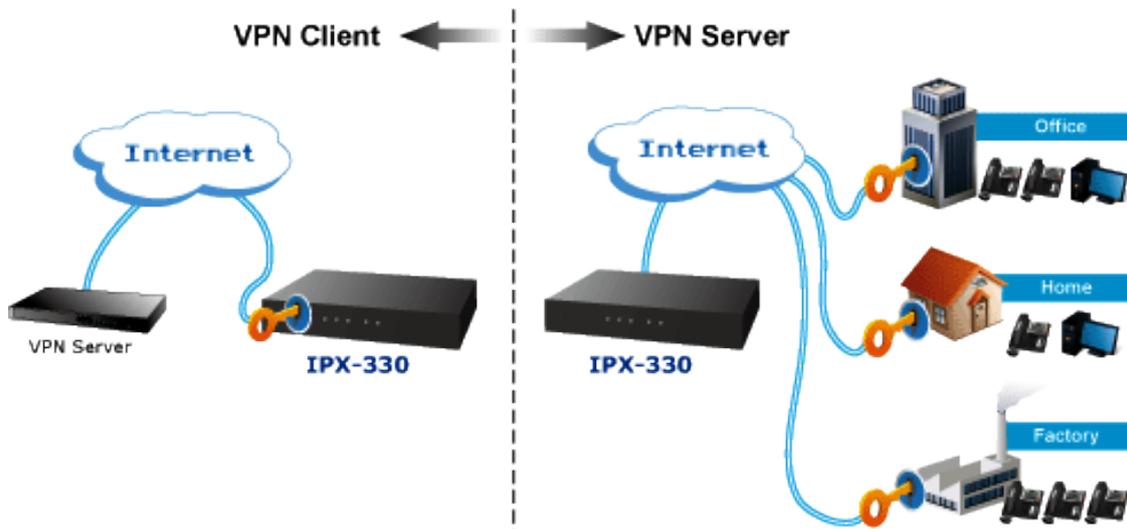
The Fax to Email / Email to Fax service by the IPX-330 allows users to transfer / receive faxes directly to / from your email inbox as file attachments. It's an easy and confidential way of receiving, storing and forwarding important fax documents, thus creating a paperless or green office.



Full Security with VPN Support

The IPX-330 VPN securely and cost-effectively connects geographically disparate offices of an organization, creating one cohesive virtual network. The IPX-330 VPN technology is also used by ordinary Internet users to connect to proxy servers for the purpose of protecting one's identity. It includes VPN server and client function that can support users full security login.

Supports VPN Client and VPN Server



1.1 Features

➤ **System Highlights**

- 10 concurrent calls and up to 30 registers
- HD voice codec G.722 for 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 Languages of GUI for international business
- Web based Control Panel for easy configuration and management of the system
- Hardware Echo Cancellation module for great and smooth communication
- Strong security features protect your system from hacking
- Supports maximum 8 ports FXO / FXS / GSM (on 2 slots)

➤ **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: RFC2833, SIP INFO, In-band

➤ **Network and Security Features**

- DDNS Client (PLANET DDNS)
- DHCP Server / SNMP v1/v2
- IEEE 802.1Q of VLAN
- IPv4 / IPv6
- Manual Configuration of Static Route Table
- Troubleshooting (Ping, Traceroute)
- VPN Client (Supports N2N / L2TP / PPTP / OpenVPN)
- VPN Server (Supports PPTP / L2TP / OpenVPN Server)
- Refuse SIP Register DoS
- Refuse Abort Invite Dos
- Refuse SSH Login DoS
- Firewall / SRTP

➤ **PBX Features**

- Black List
- BLF (Busy Lamp Field)
- CDR (Call Detailed Record)
- Conference Room (3 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 Languages of GUI
- Phone Book / PIN Set
- Record Files Download
- Ring Group / SIP Trunk
- Skype for SIP / Smart DID / System Log
- T.38 Fax (Pass-through) / Time based rule
- Virtual Fax / Voicemail & 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 Conference / Video Call

1.2 Package Contents

Thank you for purchasing PLANET Internet Telephony PBX system, IPX-330. This Quick Installation Guide will introduce how to finish the basic setting of connecting the web management interface and the Internet. Open the box of the Internet Telephony PBX system and carefully unpack it. The box should contain the following items:

- Internet Telephony PBX system unit x 1
- Quick Installation Guide x 1
- User's Manual CD x 1
- Power Adapter x 1 (12V)
- RJ-45 x 1

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

1.3 Physical Specifications

➤ Dimensions

| | |
|-------------------|----------------------------|
| Dimensions | 155(L) × 295(W) × 65(H) mm |
| Net weight | 0.5 kg (without package) |

➤ Front Panel



➤ Rear Panel

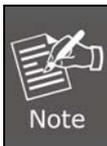


LED definitions

| Front Panel LED | State | Description |
|-----------------|----------------|------------------------------------|
| PWR | Steady Green | PBX Power ON |
| | Off | PBX Power OFF |
| SYS | Blinking Green | System is working |
| | Off | System is off |
| ETH | Blinking Green | PBX network connection established |
| | Off | Waiting for network connection |
| FXO | Steady Red | Ready / Standby |
| | Flashing | Ringing |
| | Off | Module not available |

| | | |
|----------|---------------|---|
| 1 | Reset | The reset button, when pressed, resets the IP PBX without the need to unplug the power cord. |
| 2 | 12V DC | 12V DC Power input outlet |
| 3 | ETH | The ETH port supports auto negotiating Fast Ethernet 10/100 Base-TX networks. This port allows your IP PBX to be connected to an Internet Access device, e.g. router, cable modem and ADSL modem through a CAT.5 twisted pair Ethernet cable. |
| 4 | FXO | FXO port is connects to PBX or CO line with RJ-11 (Write) analog line. FXO port was connected to the extension port of a PBX or directly connected to a PSTN line of carrier. |

| Button | Action | Description |
|--------|------------------------|--------------------------|
| Reset | Press less than 6 secs | System reboot. |
| | Press over 6 secs | Reset to Factory Default |



Please be reminded to reset to factory default. Uploaded music setting (on hold music) and backup file will not be removed.

1.4 Specifications

| | |
|-------------------------------|---|
| Product | IPX-330 Internet Telephony PBX System (30 SIP Users Registrations) |
| Hardware | |
| Ethernet | 1 x 10/100Mbps RJ-45 port |
| Analog Ports | 2 X FXO |
| Reset button | Reset to factory default |
| Protocols and Standard | |
| Standard | SIP 2.0 (RFC3261), IAX2 |
| Protocols | RFC 793 TCP RFC 826 ARP RFC 1034, 1035 DNS RFC 1631 NAT RFC 2068 HTTP RFC 2131 DHCP RFC 2516 PPPoE RFC 3261, RFC 3311, RFC 3515 RFC 3265, RFC 3892, RFC 3361 RFC 3842, RFC 3389, RFC 3489 RFC 3428, RFC 2327, RFC 2833 RFC 2976, RFC 3263 |
| Voice Codec | G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX |
| Video Codec | H.261/ H.263 / H.263+ / H.264 |
| Fax Support | T.38 Fax (Pass-through) |
| Management | HTTP Web Browser |
| Voice Processing | DTMF detection and generation In-Band and RFC 2833, SIP INFO |
| Protocols | SIP 2.0 (RFC-3261), TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE |
| Internet Sharing | |
| Network Features | DDNS Client(Planet DDNS and Easy DDNS), DHCP Server IEEE802.1Q of VLAN IP Assignment (PPPoE / DHCP / Static) IPv4 / IPv6 SNMP v1/v2 Manual Configuration of Static Route Table Troubleshooting (Ping, Traceroute) VPN Client (Supports N2N / L2TP/PPTP/OpenVPN) VPN Server(PPTP/L2TP/OpenVPN Server) |
| Security Feature | Refuse SIP Register DoS Refuse Abort Invite Dos Refuse SSH Login DoS FireWall SRTP |
| Features | |
| PBX Features | Black List BLF (Busy Lamp Field) CDR (Call Detailed Record) Conference Room(3 rooms) DID (Direct Inward Dialing Number) DISA (Direct Inward System Access) DND / Feature Codes / Flash Operation Panel |

| | |
|---|--|
| | <p>Follow Me / Auto-Provision IVR (Interactive Voice Responses) Multi-language System Prompt Multiple Languages of GUI Phone Book / PIN Set Record Files Download Ring Group / SIP Trunk Skype for SIP / Smart DID / System Log T.38 Fax (Pass-through) / Time based rule Virtual Fax / Voicemail & Voice Mail to E-Mail</p> |
| Call Features | <p>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</p> |
| System Capacity | |
| System Capacity | <p>20 Concurrent Call Legs Up to 100 IP Phone Registers/Extensions Recording(GSM/ default): 21,000 minutes; Wav: 3000 minutes Voicemail(GSM/ default): 21,000 minutes; Wav: 3000 minutes</p> |
| Network and Configuration | |
| Access Mode | Static IP, PPPoE, DHCP |
| LED Indications | <p>SYS: 1, LNK/Off ETH: 1, LNK/Off PWR: 1, LNK/Off FXO: Red</p> |
| Dimensions (W x D x H) | 343 x 154 x 35 mm |
| Operating Environment | -10~45 degrees C, 10~80% humidity |
| Power Requirements | <p>Input: 100 ~ 240 Vac Output: DC 12V / 2.0 A</p> |
| EMC/EMI | CE, FCC Class B, RoHS |
| Remarks: T.30/ T.38 support is dependent on fax machine, SIP provider and network / transport resilience. | |

Chapter 2 Installation Procedure

2.1 Web Login

Step 1. Connect a computer to an ETH port on the IPX-330. Your PC must set up to the same domain of 192.168.0.X as that of the IPX-330.

Step 2. Start a web browser. To use the user interface, you need a PC with Internet Explorer (version 6 and higher), Firefox, or Safari (for Mac).

Step 3. Enter the default IP address of the IPX-330: 192.168.0.1 in the URL address box.

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

(Default IP)

Default ETH IP: **192.168.0.1**

Default User Name: **admin**

Default Password: **admin**

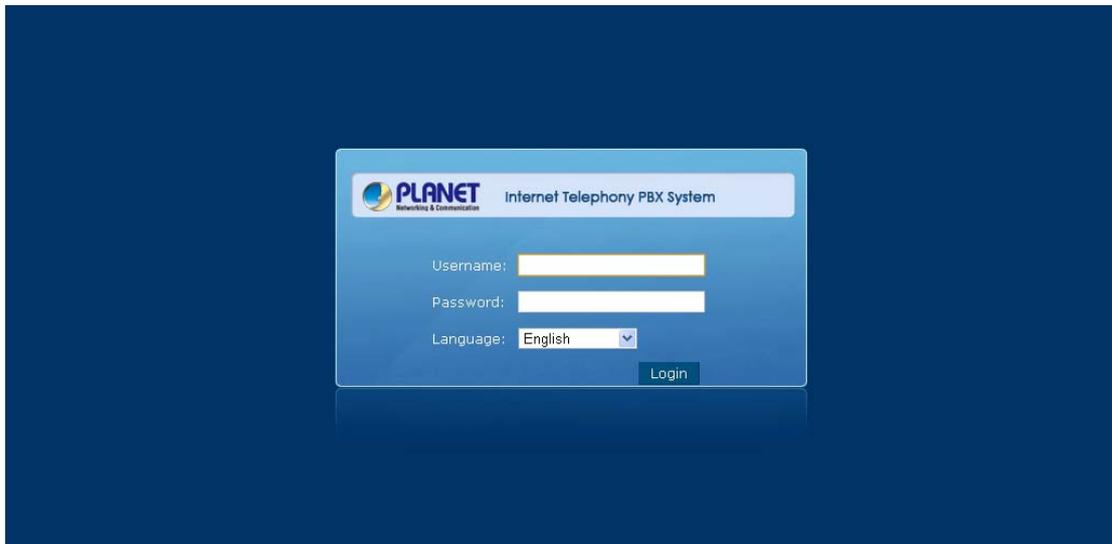


Figure 2-1. Login page of the IPX-330



Note

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

2.2 Configuring the Network Setting

Step 1. Go to Network Settings → Network

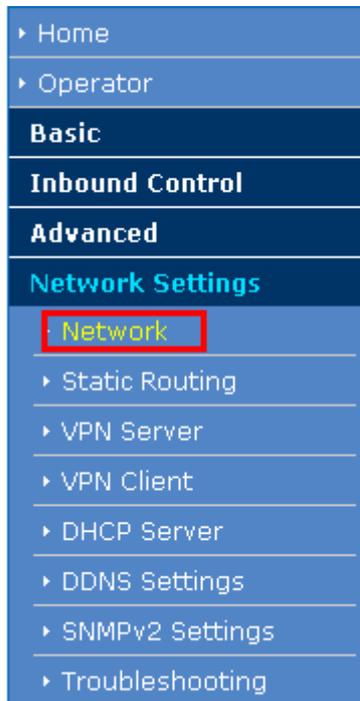


Figure 2-2. Network & Country Button

Network

| | | | | | |
|--|----------------------|----------------|----------------------|---------------|--|
| IPv4 Settings | | IPv6 Settings | | VLAN Settings | |
| Ethernet Port Setup | | | | | |
| IP Assign: | Static | | | | |
| Hostname: | IPPBX | | | | |
| IP Address: | 192.168.1.198 | | | | |
| Subnet Mask: | 255.255.255.0 | | | | |
| Gateway: | 192.168.1.254 | | | | |
| Primary DNS: | 192.168.1.254 | | | | |
| Alternate DNS: | | | | | |
| Virtual Interface | | | | | |
| <input type="checkbox"/> IP AddressV1: | <input type="text"/> | Subnet MaskV1: | <input type="text"/> | | |
| <input type="checkbox"/> IP AddressV2: | <input type="text"/> | Subnet MaskV2: | <input type="text"/> | | |

Figure 2-3. Network Setting page

Step 2. Edit your ETH port IP information .

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

Network

IPv4 Settings
IPv6 Settings
VLAN Settings

Ethernet Port Setup

| | |
|----------------|---------------|
| IP Assign: | Static |
| Hostname: | Static |
| IP Address: | DHCP 198 |
| Subnet Mask: | 255.255.255.0 |
| Gateway: | 192.168.1.254 |
| Primary DNS: | 192.168.1.254 |
| Alternate DNS: | |

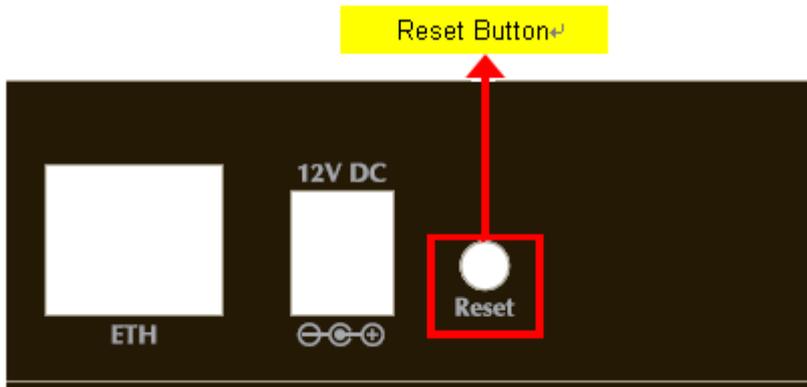
Virtual Interface

| | | | |
|--|--|----------------|--|
| <input type="checkbox"/> IP AddressV1: | | Subnet MaskV1: | |
| <input type="checkbox"/> IP AddressV2: | | Subnet MaskV2: | |

Figure 2-4. Selection of IP Connection Type

2.3 Changing IP Address or Forgotten Admin Password

To reset the IP address to the default IP address “192.168.0.1”(ETH) or reset the login password to default value, press the reset button on the front panel for **more than 6 seconds**. After the device is rebooted, you can login the management WEB interface within the same subnet of 192.168.0.xx.



Note After pressing the “Reset” button, all the system data will be reset to default; if possible, back up the config file before resetting.

Chapter 3 Basic Configuration

3.1 Preparation Before Operation

What kind of IP phone can be used with the IPX-330 IP PBX?

- Our IPX-330 is based on SIP 2.0 (RFC 3261); any IP phone model based on the same protocol can work with the IPX-330.

3.2 Before Making a Call

3.2.1 System Information

Default ETH IP: **192.168.0.1**

Default Name: **admin**

Default Password: **admin**

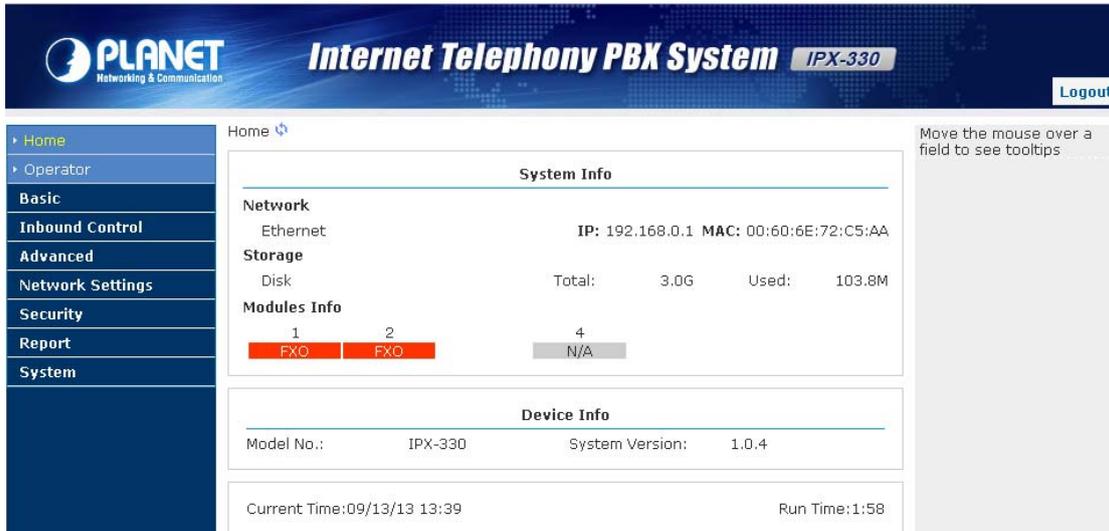


Note

1. To login to the IPX-330, your PC must use the same domain as the eth0 IP address of the IPX-330.
2. For security reason, please modify the user name and password after you login. You can modify it on this page: "System"---"Management"

[Warning]: Every Time after saving the change, please press the "Activate Changes" to make modification effective.

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



The screenshot shows the main dashboard of the IPX-330 system. It includes a navigation menu on the left with options like Home, Operator, Basic, Inbound Control, Advanced, Network Settings, Security, Report, and System. The main content area displays 'System Info' with sections for Network (Ethernet IP: 192.168.0.1, MAC: 00:60:6E:72:C5:AA), Storage (Total: 3.0G, Used: 103.8M), and Modules Info (Slots 1 and 2 are FXO, Slot 4 is N/A). Below this is 'Device Info' showing Model No.: IPX-330 and System Version: 1.0.4. At the bottom, it shows the current time and run time.

| | | |
|---|--------------------|--|
| 1 | Network | ETH0 IP and MAC will be displayed |
| 2 | Storage | Total storage and used storage will be displayed |
| 3 | Slots Info | Channel information will be based on the product model |
| 4 | Device Info | Product Model and System Version will be displayed |

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

Commonly Used Button

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

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

System Menu

System Menu includes the following sub menu:

| | | |
|---|-----------------|------------------------------------|
| 1 | Home | Display device information |
| 2 | Operator | Extension / Trunk / Channel Status |

3.2.3 Basic Configuration

Configure Extensions

Planet IP PBX supports SIP/IAX2 and analog extension; configure extension on this page:

【Basic】 ---- **【Extensions】**

- ▶ Home
- ▶ Operator
- Basic
- ▶ Extensions
- ▶ Trunks
- ▶ Outbound Routes
- Inbound Control
- Advanced
- Network Settings
- Security
- Report
- System

Extensions

Extensions
Upload/Download Extensions

Extension: Search Show All

New User
Batch Add Users
Delete Selected Users

| | Name | Extension | Port | Protocol | DialPlan | Outbound CID | Options |
|--------------------------|--------|-----------|------|----------|-----------|--------------|----------------------|
| <input type="checkbox"/> | 1 800 | 800 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 2 801 | 801 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 3 802 | 802 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 4 803 | 803 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 5 804 | 804 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 6 805 | 805 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 7 806 | 806 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 8 807 | 807 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 9 808 | 808 | -- | SIP | DialPlan1 | | Edit |
| <input type="checkbox"/> | 10 809 | 809 | -- | SIP | DialPlan1 | | Edit |

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

New X

General

SIP: IAX2:
 Name: Extension:
 Password: Outbound CID:
 DialPlan: Analog Phone:

Voicemail

Voicemail: VM Password:
 Delete VMail: Email(Fax/Voicemail):

Other Options

Web Manager: Agent: Call Waiting:
 Allow Being Spied: Pickup Group:
 Mobility Extension: Mobility Extension Number:

VoIP Settings

NAT: Transport: SRTP:
 DTMF Mode: Permit IP:

Video Options

Video Call:
 H.261 H.263 H.263+ H.264

Audio Codecs

alaw ulaw G.722 G.729 G.726 GSM Speex

Extension Settings

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

| | |
|-----------------------|--|
| SIP/IAX2 | Choose extension protocol. |
| Name | Extension Name (English Character Only), e.g. Tom. |
| Extension | Extension Number connected to the phone, e.g. 888. |
| Password | Same password as voicemail. (4-16 digits, e.g.123456) |
| Outbound CID | Override the caller ID when dialing out with a trunk. |
| Dial Plan | Please choose the Dial Plan which is defined in the menu "Outbound Routes". |
| Analog Phone | Please select the related FXS port for your analog phone. |
| Voicemail | Select this option to open the voicemail account |
| VM Password | Set password for Voicemail, e.g. "1234" |
| Delete VMail | Check this option to delete voicemail from system after it's sent to mail box. |
| Email (Fax/Voicemail) | Extension user's mail box, which is used for receiving fax or voicemail (you need to open the function to fax to email/voicemail), e.g. Tom@gmail.com |
| Web Manager | It's allowed to login Extension Management Panel to manage extension like voicemail, call recording, call transfer, etc when you select this option. |
| Agent | Check this option to set this extension user as agent. |
| Call Waiting | Enable call waiting |
| Allowing Being Spied | Check this option to allow being spied. |
| NAT | Check this option if extension user or the phone is located after the NAT (Network Address Translation) available gateway. |
| Pickup Group | Select the Pickup Group which the extension user belongs to. |
| Mobility Extension | After checking this option, you must set mobility extension number. User can make calls to the IP PBX server with this mobility number, and have all rights of this extension, e.g. Outbound Call, Internal Call, Listen to the voicemail. |
| Transport | Select the Transport Protocol: UDP, TCP, TLS |
| SRTP | Enable SRTP |
| DTMF Mode | Default DTMF is rfc2833. It can be changed if necessary. |
| Video Call | Check to enable video call for this extension. And select the audio codecs you need to use. |
| Permit IP | Set computer permitted IP to visit this IP PBX, e.g.192.168.1.77or 192.168.10.0/255.255.255.0. Computer with other IPs is not allowed to visit this IP PBX. |

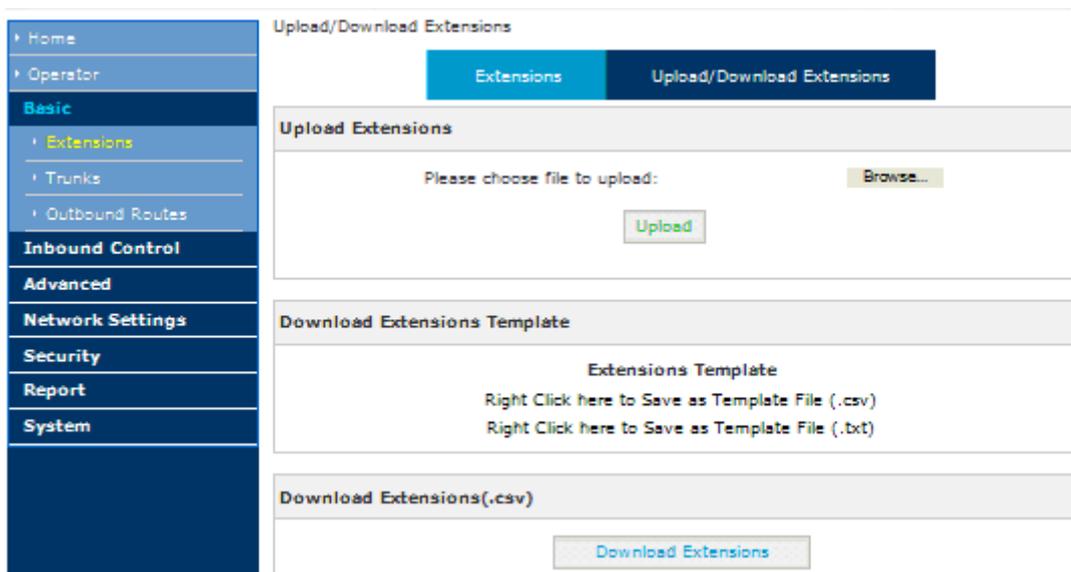
| | |
|-------------|--|
| Audio Codec | Select what audio codec you need to use. |
|-------------|--|


Note

1. There are few default extensions which number started with "8XX", you can add or delete extension by your requirement
2. Maximum extensions: 100.
3. For security reason the default password is random character or number e.g. BB%ChH64rl, and every time when you reset to default system, it will randomly have a new password again

Upload/Download Extensions

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



Download the extension template from the **【Download Extensions Templet】** , add extension information based on the template format and save.

Select the extension file to upload from **【Upload Extensions】**

Download current extension information from **【Download Extensions (.csv)】**

3.2.4 Time-based Rules

Please set time rule for working time and after-working time, and deal with inbound calls based on this time rule.

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

X
Edit

Rule Name:

Time & Date Conditions

Start Time: : End Time: :

Start Day: End Day:

Start Date: End Date:

Start Month: End Month:

Destination

if time matches:

if time unmatches:

New Time Rule:

| Item | Explanation |
|------------------------|--|
| Rule Name | Define the name for this Time Rule. |
| Time & Date Conditions | Set time segment for Day/ Date/ Month. |
| Destination | How to deal with the inbound call in different time segments. For example, inbound call can be directed to operator in working time. |

3.3 Outbound Call

3.3.1 Trunks

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

- ▶ Home
- ▶ Operator
- Basic
- ▶ Extensions
- ▶ Trunks
- ▶ Outbound Routes
- Inbound Control
- Advanced
- Network Settings
- Security
- Report
- System

VoIP Trunks

VoIP Trunks

FXO/GSM Trunks

List of Trunks
New VoIP Trunk

| Provider Name | Type | Hostname/IP | Username | Options |
|---|------|-------------|----------|---------|
| No VoIP Trunk defined Please click on 'New VoIP Trunk' button to add a Trunk | | | | |

Planet IP PBX supports 2 kinds of trunks: VoIP Trunks and FXO/ FXS Trunks.

VoIP Trunks

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

X
New VoIP Trunk

Description: _____

Protocol: SIP ▾

Host: _____ : 5060

Maximum Channels*: 0

Prefix: _____

Caller ID: _____

Without Authentication

Username: _____

Authuser: _____

Password: _____

Advanced Options

Domain: _____ Insecure: port,invite

From User: _____ Qualify(sec): 2

DID Number: _____ Transport: UDP ▾

DTMF Mode: RFC2833 ▾ NAT: SRTP:

Auto Fax Detection:

Context: Default ▾ Language: Default ▾

Audio Codecs

alaw ulaw G.722 G.729 G.726 GSM Speex

Video Codes

H.261 H.263 H.263+ H.264

Save
Cancel

| Item | Explanation |
|------------------------|---|
| Description | Define the VoIP (figure or character). |
| Protocol | Select protocol for outbound route, SIP or IAX2. |
| Host | Set host address (provided by VoIP Provider). |
| Maximum Channels | Set maximum channels for simultaneous call. (Only for outbound call; "0" = no limitation). |
| Prefix | The prefix will be added in front of your dialed number automatically when the trunk is in use. |
| Caller ID | This Caller ID will be displayed when user make outbound call. Note: This function must be supported by local provider. |
| Without Authentication | If you don't need the Authentication when connecting the IP PBX, please check this option. |
| User Name | User Name provided by VoIP Provider. |

| | |
|------------------|--|
| Password | Password provided by VoIP Provider. |
| Advanced Options | Advanced options for this trunk, e.g. codec, dial plan, etc. |

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

2) FXO/GSM Trunk

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

New FXO/GSM Trunk X

Description: _____

Lines: **FXO:** 3 4
GSM: _____

Prefix: _____

Advanced Options

Call Method: ▼

Busy Detection: ▼ Busy Count:

Input Volume: ▼ Output Volume: ▼

Call Progress: ▼ Progress Zone: ▼

Busy Pattern: _____ Language: ▼

Answer on Polarity Switch: ▼

Hangup on Polarity Switch: ▼

Auto Fax Detection:

| Item | Explanation |
|------------------|--|
| Description | Define the description for this trunk (figure or character). |
| Lines | Available line |
| Prefix | The prefix will be added to the dialed number automatically when this trunk is in use. |
| Advanced Options | Advanced Options for this trunk, e.g. Call Method, Busy Detection, etc. |

Set the available analog line for this device. The same analog line can't be used in several FXO/GSM trunks. If you don't have available analog line, you can't set up FXO/GSM trunk.

3.3.2 Outbound Routes

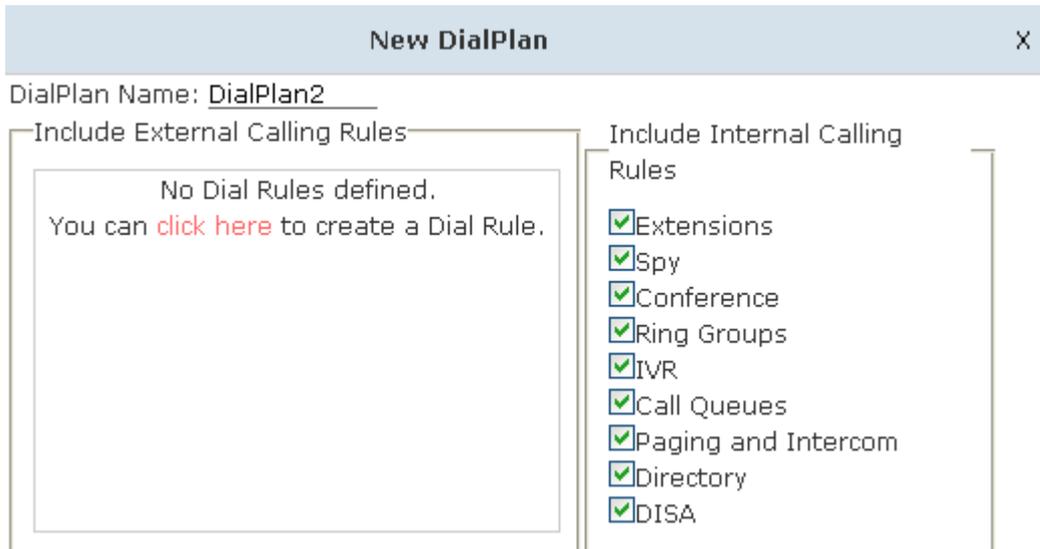
Outbound Routes is to define what trunk is used for outbound call by extension user. If user don't allow extension user to call out, please ignore this part.

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



| Default | DialPlan Name | Rules | Options |
|-------------------------------------|---------------|--|---|
| <input checked="" type="checkbox"/> | 1 DialPlan1 | Extensions, Spy, Conference, Ring Groups, IVR, Call Queues, Paging and Intercom, Directory, DISA | Edit Delete |

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



New DialPlan [X]

DialPlan Name: DialPlan2

Include External Calling Rules

No Dial Rules defined.
You can [click here](#) to create a Dial Rule.

Include Internal Calling Rules

- Extensions
- Spy
- Conference
- Ring Groups
- IVR
- Call Queues
- Paging and Intercom
- Directory
- DISA

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

New DialRule X

Rule Name: _____

PIN Set:

Place this call through:

^

v

>>>
 ↓
 ↑
 <<<

^

v

Available Trunks
Selected Trunks

Custom Pattern: _____

Z Any digit from 1 to 9

N Any digit from 2 to 9

X Any digit from 0 to 9

. Any number of additional digits

Delete ___ digits prefix from the front and auto-add digit _____ before dialing

| Item | Explanation |
|-------------------------|---|
| Rule Name | Define the name for the dial rule. |
| Pin Set | Input this Pin when you use this dial rule. |
| Place this call through | Select a trunk for this dial rule |
| Custom Pattern | N any figure from 2 to 9 Z any figure from 1 to 9 X any figure from 0 to 9 . One figure or multi-digit figures |
| Delete[]digits prefix | If one digit prefix be deleted, when dial 12345, 2345 will be sent. |
| Auto-add digit[] | If figure "1" is added,123451 will be sent when dialing 12345 |

3.4 Inbound Call

3.4.1 Inbound Routes

When a call is made from outside, you want to forward this call to an extension or IVR. This Chapter will introduce you how to deal with the inbound calls.

Please configure it on this page: **【Inbound Routes】**

- Home
- Operator
- Basic**
- Inbound Control**
- Inbound Routes
- IVR
- IVR Prompts
- Call Queues
- Ring Groups
- Black List
- Time Based Rules
- Advanced**
- Network Settings**
- Security**
- Report**
- System**

General

General

Port DIDs

Number DIDs

DOD Settings

From Analog Channels

Distinctive Ring Tone: _____

Destination: Goto IVR working time

From VoIP Channels

Distinctive Ring Tone: _____

Destination: Goto IVR working time

Save
Cancel

General

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

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

Port DIDs

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

Click **【Port DIDs】** -> **【New Port DIDs】** :

New Port DID ✕

Port: Label: _____

Destination: Goto Extension 800(800)

Save
Cancel

| Item | Explanation |
|------|------------------------------------|
| Port | Select the port for outbound line. |

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

Number DIDs

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

Click **【 Number DID 】** -> **【 New Number DID 】** :

X

DID Number:

Destination: Goto Extension ▼ 800(800) ▼

Save
Cancel

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

DOD Settings

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

X

DOD Number:

Destination: Goto Extension ▼ 800(800) ▼

Save
Cancel

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

3.4.2 IVR

IVR will improve office efficiency based on your requirement.

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

- ▶ Home
- ▶ Operator
- Basic**
- Inbound Control**
- ▶ Inbound Routes
- ▶ **IVR**
- ▶ IVR Prompts
- ▶ Call Queues
- ▶ Ring Groups
- ▶ Black List
- ▶ Time Based Rules
- Advanced**
- Network Settings**
- Security**
- Report**
- System**

IVR

| List of IVRs | | | | New IVR | |
|--------------|-----------|--------------|-----------------------|-------------------------|------------------------|
| | Extension | Name | Dial other Extensions | Options | |
| 1 | 610 | working time | Yes | Edit | Delete |
| 2 | 611 | closed time | No | Edit | Delete |

Click **【New IVR】** to create a new IVR:

X
New IVR

IVR Settings

Name: _____ Extension: 612

Welcome Message

Please Select: Test [Custom Prompts](#)

Repeat Loops: None

Dial other Extensions

Keypress Events

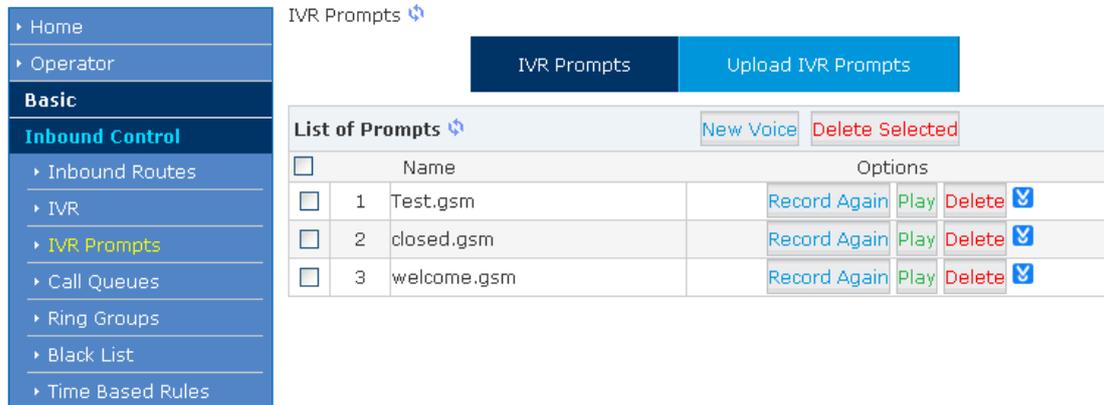
| Key | Action |
|-----|---|
| 0 | Disabled ▼ |
| 1 | Disabled ▼ |
| 2 | Disabled ▼ |
| 3 | Disabled ▼ |
| 4 | Disabled ▼ |
| 5 | Disabled ▼ |
| 6 | Disabled ▼ |
| 7 | Disabled ▼ |
| 8 | Disabled ▼ |
| 9 | Disabled ▼ |
| * | Disabled ▼ |
| # | Disabled ▼ |
| t | Disabled ▼ |

Save
Cancel

| Item | Explanation |
|-----------------------|--|
| Name | Set a name for the IVR |
| Extension | If you want to listen to the IVR by dialing extension, please input an extension Number. |
| Please Select | Select IVR audio file, please configure in this page: 【IVR Prompts】 |
| Repeat Loops | Loop times to repeat playing the IVR prompt. |
| Dial Other Extensions | Allow caller to dial other extensions besides the ones listed below. |
| Key Press Events | Each digit will be related to the actions defined in the blank. |

3.4.3 IVR Prompts

Record or play IVR music from extension. Please configure on this page: **【IVR Prompts】**



| Name | | Options |
|--------------------------|---------------|--------------------------|
| <input type="checkbox"/> | 1 Test.gsm | Record Again Play Delete |
| <input type="checkbox"/> | 2 closed.gsm | Record Again Play Delete |
| <input type="checkbox"/> | 3 welcome.gsm | Record Again Play Delete |

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

New Voice X

File Name:

Format: GSM ▼

Extension used for recording: 800 ▼

Record
Cancel

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

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

Play record voice
X

Extension used for playing:

Play
Cancel

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

Upload IVR prompt

- ▶ Home
- ▶ Operator
- Basic**
- Inbound Control**
- ▶ Inbound Routes
- ▶ IVR
- ▶ **IVR Prompts**
- ▶ Call Queues
- ▶ Ring Groups
- ▶ Black List
- ▶ Time Based Rules
- Advanced**
- Network Settings**

Upload IVR Prompts

IVR Prompts
Upload IVR Prompts

Upload IVR Prompts

Note: The sound file must be wav(16bit/8000Hz/Single), gsm, ulaw or alaw!
The size is limited in 15MB!

Please choose file to upload:

Upload



Note

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

3.4.4 Ring Groups

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

There isn't any data in the factory default **【Ring Groups】** , please configure it here.

Click **【Inbound Control】** -> **【Ring Groups】** -> **【New Ring Group】** :

New Ring Group X

Name: _____ Strategy: RingAll v

Ring Group Members

<<<

←

→

>>>

800(SIP) 800

801(SIP) 801

802(SIP) 802

803(SIP) 803

804(SIP) 804

805(SIP) 805

806(SIP) 806

807(SIP) 807

Available Channels

Label: _____

Extension for this ring group: 640

Ring (each/all) for lasting time(sec): 20

If not answered

Goto Extension

Goto Voicemail

Goto Ring Group

Goto IVR

Hangup

Save
Cancel

| Item | Explanation |
|--------------------|---|
| Name | Define a name for the Ring Group. |
| Strategy | Select "Ring All" or "Ring in order". |
| Ring Group Members | Select the Ring Group Member from "the Available Channels", click to add. |
| If not answered | You can choose to forward the call to extension, voicemail, ring group, IVR or hang up if not answered. |

3.5 Black List

If some numbers need to be blocked, you can use this functionality, please configure it here:

Click **【Inbound Control】** -> **【Blacklist】** -> **【New Blacklist】**

New Blacklist X

Blacklist Number: _____

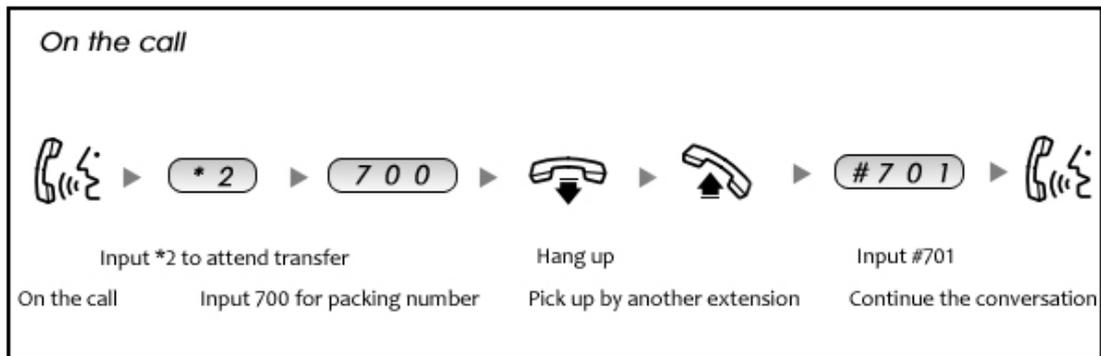
Save
Cancel

Input caller's number in the blank, then this caller's number will be blocked when the call

3.6 On The Call

3.6.1 Call Parking

If you pick up a call at your seat, but it's not convenient to talk in public, you need go to the conference room to talk secretly. At this time, you can input 700 to park this call. The system will tell you a parking number 701 which you can input for continuing conversation when you go to the conference room. Please check the following diagram to learn more:

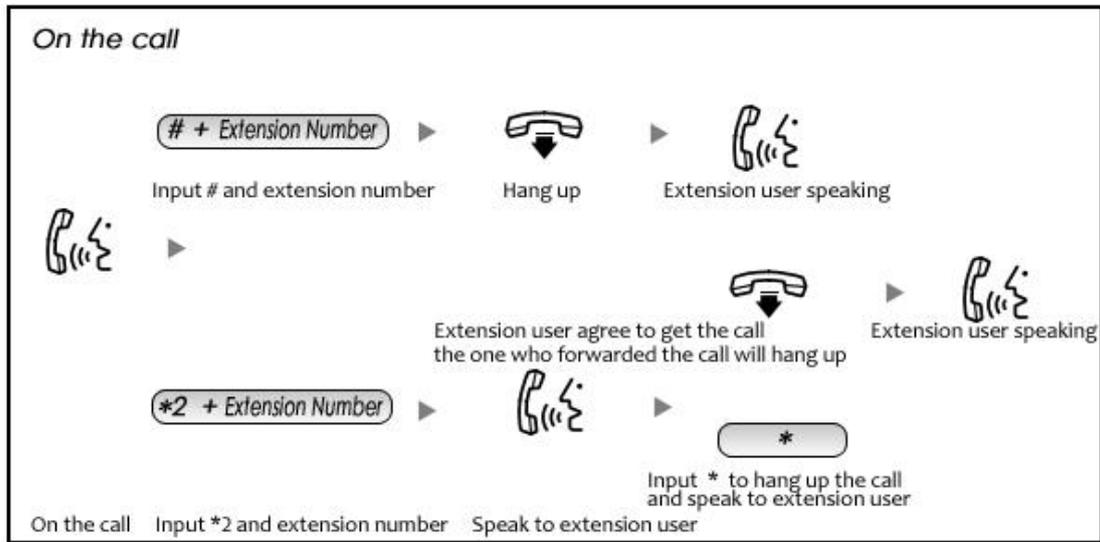


Reference Parameters and Explanation of Call Park:

| Item | Explanation |
|---|---|
| Extension to Dial for Parking Calls | Default Number: 700, Define in 【Feature Codes】 |
| What Extension to park calls on | Default Number: 701 - 720. Define in 【Feature Codes】 |
| How many seconds a call can be parked for | Default is 45 seconds. Define in 【Feature Codes】 . |

3.6.2 Call Transfer

If an incoming call is for your colleague, you can transfer the call directly to your colleague or transfer the call after agreeing by your colleague. Please check the diagram below to learn more:



Reference Parameters and Explanation of Transfer:

| Item | Explanation |
|---|---|
| Blind Transfer | Default is #t. Define in 【Feature Codes】 |
| Attended Transfer | Default is *2. Define in 【Feature Codes】 |
| Disconnect Call | Default is *, it can be used when you use *2. Define in 【Feature Code】 |
| Timeout for answer on attended transfer | Default is 15 seconds. Define in 【Feature Codes】 |

3.6.3 Conference

If you want to create a conference room for some extension users or with external lines, you can input conference room number 900, input conference room password 1234 (Admin's password is 2345), then enter the conference room. This IPX-330 supports 3 conference rooms. Please configure it on this page **【Conference】** :

- ▶ Home
- ▶ Operator
- Basic**
- Inbound Control**
- Advanced**
- ▶ Options
- ▶ Voicemail
- ▶ SMTP Settings
- ▶ Email to Fax
- ▶ **Conference**
- ▶ Music Settings
- ▶ DISA
- ▶ Follow Me
- ▶ Paging and Intercom
- ▶ PIN Sets
- ▶ Call Recording
- ▶ Speed Dial
- ▶ Smart DID
- ▶ Callback
- ▶ Phone Book

Conference(Default)

Conference(Default)

Conference 2

Conference 3

Conference Number

Room Extension:

Conference Password

Guest Password:

Administrator Password:

Conference Options

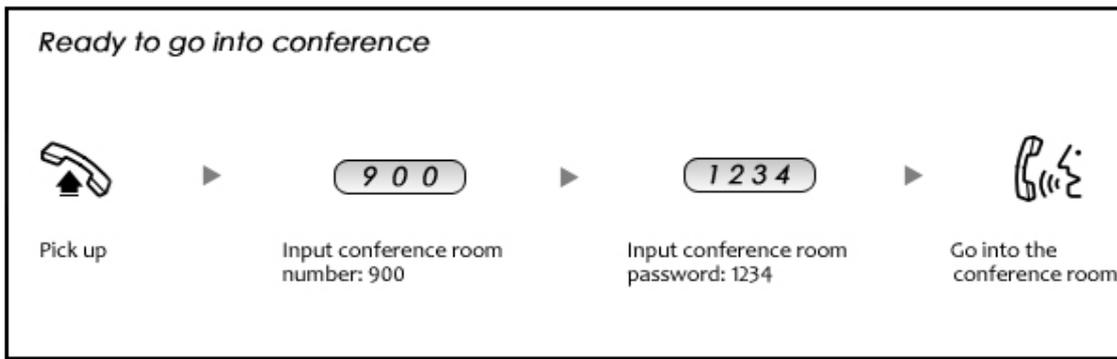
Conference DialPlan:

- Play hold music for first caller
- Enable caller menu
- Announce callers
- Record conference
- Quiet Mode
- Leader Wait

| Item | Explanation |
|---|--|
| Conference Number | The number that users call in order to access the conference room; the default number is "900". |
| Conference Password | Password for users to access the conference, e.g."1234". |
| Administrator Password | Password for administrator to access the conference. |
| Conference DialPlan | Use this dial plan to invite other participants. |
| Play hold music for the first participant | Check this option to play the hold music for the first participant in the conference until another participant enters this conference. |
| Enable caller menu | Check this option to allow the participant to access the Conference Bridge menu by pressing "*" on the dialpad. |
| Announce callers | Check this option to announce to all Bridge participants that a new participant is joining the conference. |
| Record conference | Recorded conference format is WAV. |
| Quiet Mode | If this option is checked, all the participants in the conference can hear only, but it is not allowed to speak. |
| Leader Wait | Wait until the conference leader (administrator) enters the conference before starting the conference. |

Please check the following diagram to learn:

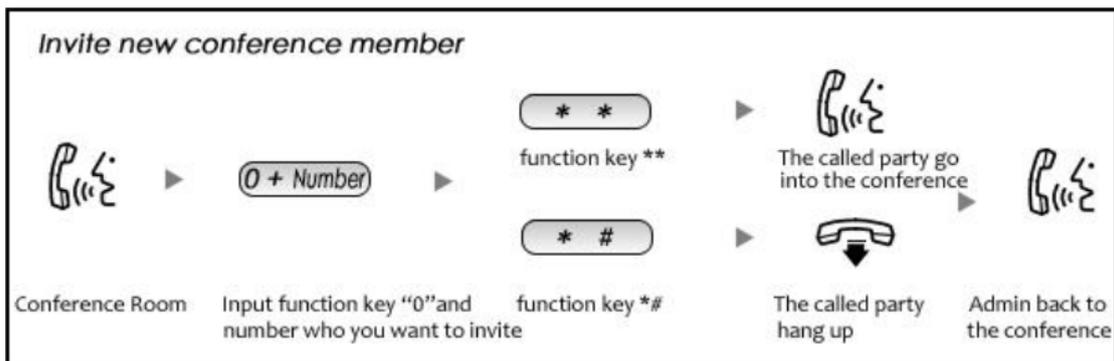
Go to conference:



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

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

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



3.7 Settings before leaving office

3.7.1 Follow me

If you don't want to miss any call, please configure this function as shown below:

Click **Basic** -> **Extension** -> **Edit** the extension you want to configure.

Edit
X

General

SIP: IAX2:
 Name: Extension:
 Password: Outbound CID:
 Dial Plan: Analog Phone:

Voicemail

Voicemail: VM Password:
 Delete VMail: Email(Fax/Voicemail):

Other Options

Web Manager: Agent: Call Waiting:
 Allow Being Spied: Pickup Group:
 Mobility Extension: Mobility Extension Number:

VoIP Settings

NAT: Transport: SRTP:
 DTMF Mode: Permit IP:

Video Options

Video Call:
 H.261 H.263 H.263+ H.264

Audio Codecs

alaw ulaw G.722 G.729 G.726 GSM Speex

Check **【Web Manager】** and **【Save】**

Then login to the Extension Web Panel:


Internet Telephony PBX System **IPX-330**
Username:800
Logout

- ▶ Record List
- ▶ Voicemail List
- ▶ Call Forward
- ▶ Follow Me
- ▶ Settings
- ▶ Send Fax

Call Recording

Call Recording
One Touch Recording

Start Date: End Date:

List of Recording Files

| Caller ID | Destination ID | Date | Options |
|-----------|----------------|------|---------|
| | | | |

Move the mouse over a field to see tooltips



Note

Extension Web Panel default Login user name = extension account
password = Voice mail password (Default is 1234)

Click **【Call Forward】** :

| Forward Settings | |
|---|-----------------|
| <input type="checkbox"/> | Always _____ |
| <input type="checkbox"/> | Busy _____ |
| <input type="checkbox"/> | No Answer _____ |
| <input type="button" value="Save"/> <input type="button" value="Cancel"/> | |

Reference

| Item | Explanation |
|-------------|--|
| Always | All incoming calls will be forwarded. |
| Busy | Forward when extension is busy. |
| No Answer | Forward when no answer from extension. |

Select an extension, set the ring duration, and add the numbers in the Follow Me List; **【Save】** and **【Activate】** .

List Format: Extension Number, Ring Duration

E.g.: 806,30

808,20

806 rings, after 30 seconds, the call is going to 808

【Follow Me Option】

| Follow Me Options | |
|-------------------------------------|--|
| <input type="checkbox"/> | Playback the incoming status message prior to starting the follow-me step(sec). |
| <input type="checkbox"/> | Record the caller's name so it can be announced to the callee on each step. |
| <input type="checkbox"/> | Playback the unreachable status message if we've run out of all steps or the callee was set not to be reachable. |
| <input type="button" value="Save"/> | |

3.7.2 Voice Mail

If you don't want to configure "Follow Me", you can record the message of incoming call, and email the message to your defined mailbox.

Click **【Extension】** --- **【Extension Settings】**

Edit
X

| | | | |
|--|--|---------------|--|
| Name: | <input type="text"/> | Extension: | <input type="text" value="804"/> |
| Password: | <input type="text" value="804"/> | Outbound CID: | <input type="text"/> |
| VM Password: | <input type="text" value="804"/> | E-mail: | <input type="text"/> |
| Dial Plan: | <input type="text" value="DialPlan1"/> ▼ | | |
| Analog Phone: <i>No Analog lines detected.</i> | | | |
| VoiceMail: | <input checked="" type="checkbox"/> | Can Reinvite: | <input type="checkbox"/> |
| SIP: | <input checked="" type="checkbox"/> | IAX2: | <input type="checkbox"/> |
| T.38 Fax: | <input type="checkbox"/> | Agent: | <input type="checkbox"/> |
| NAT: | <input checked="" type="checkbox"/> | Pickup Group: | <input type="text" value="0"/> ▼ |
| Delete VMail: | <input type="checkbox"/> | DTMF Mode: | <input type="text" value="RFC2833"/> ▼ |
| Video Call: | <input type="checkbox"/> | Permit IP: | <input type="text"/> |

Auto Provision

Manufacturer: ▼ Mac

Audio Codecs Configure

alaw
 ulaw
 G.729
 G.726
 GSM
 Speex

Video Codecs Configure

H.261
 H.263
 H.263+
 H.264

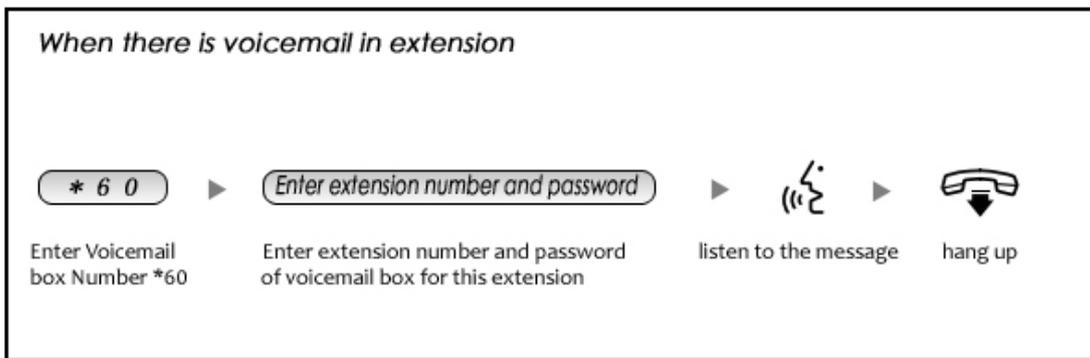
Save
Cancel

Please enable **【Voice mail】** before configuration, and configure **【VM Password】** and **【Email】**. If there is no answer for the incoming call and when the default ring time is over, the system will play: "Please leave your message and press the "#" key. Then voicemail will be sent to the specified mailbox by email.

Leave a message:



Listen to the message



1. If you would like to use this function, you must write the correct email address in "extension settings".
2. You need to configure SMTP and Email model in **【Voice Mail】** . Please check the details in the following chapter **【Voice Mail】**

3.8 Call Center(Call Queues)

3.8.1 Create Agent

Click **【Basic】** -> **【Extension】** -> **【Edit】** the extension you want to configure:

Edit X

General

SIP: IAX2:
 Name: Extension:
 Password: Outbound CID:
 Dial Plan: Analog Phone:

Voicemail

Voicemail: VM Password:
 Delete VMail: Email(Fax/Voicemail):

Other Options

Web Manager: Agent: Call Waiting:
 Allow Being Spied: Pickup Group:
 Mobility Extension: Mobility Extension Number:

VoIP Settings

NAT: Transport: SRTP:
 DTMF Mode: Permit IP:

Video Options

Video Call:
 H.261 H.263 H.263+ H.264

Audio Codecs

alaw ulaw G.722 G.729 G.726 GSM Speex

Step1: Check **Agent** and **Save**

Step2: Click **Inbound Control** -> **Call Queues**

- Home
- Operator
- Basic**
- Inbound Control
- Inbound Routes
- IVR
- IVR Prompts
- Call Queues
- Ring Groups
- Black List
- Time Based Rules
- Advanced**
- Network Settings
- Security
- Report
- System

Call Queues 1

Call Queues 1
Call Queues 2
Call Queues 3

Call Queue Reference:

Queue Number: Label:

Ring Strategy:

Agents:

You do not have any users defined as agents!
[click here](#) to manage users.

| Queue Options: | Announcements: |
|--|---|
| Agent TimeOut(sec): <input type="text" value="15"/> <input type="checkbox"/> Auto Pause Wrap-Up-Time(sec): <input type="text" value="10"/> Max Wait Time(sec): <input type="text"/> Max Callers: <input type="text" value="8"/> <input type="checkbox"/> Join Empty <input type="checkbox"/> Leave When Empty <input type="checkbox"/> Auto Fill <input type="checkbox"/> Report Hold Time | Caller Position Announcements Frequency(sec): <input type="text" value="30"/> Announce Hold Time: <input type="text" value="yes"/> Periodic Announcements Repeat Frequency(sec): <input type="text" value="0"/> Announcements Prompt: <input type="text"/> If not answered Destination: <input type="text" value="Hangup"/> |

| Item | Explanation |
|---------------|---|
| Queue Number | Define an extension number for the queue. |
| Label | Define the label for the queue. |
| Ring Strategy | RingAll -- Ring all available agents until one answers (default) RoundRobin -- Every available agent will take turns to ring. LeastRecent -- Agent with the least calls rings FewestCalls -- Agent with the fewest completed calls rings. Random -- Agent rings randomly. RRmemory -- RoundRobin with Memory, and remember where it's left off in the last ring. |
| Agent | Every extension defined as Agent will be listed here. Selected agent will be a member of the current Queue. |

| Queue Options: | Announcements: |
|--|---|
| Agent TimeOut(sec): <u>15</u> <input type="checkbox"/> Auto Pause Wrap-Up-Time(sec): <u>10</u> Max Wait Time(sec): <input type="text"/> Max Callers: <u>8</u> <input type="checkbox"/> Join Empty <input type="checkbox"/> Leave When Empty <input type="checkbox"/> Auto Fill <input type="checkbox"/> Report Hold Time | Caller Position Announcements Frequency(sec): <u>30</u> Announce Hold Time: <u>yes</u> ▾ Periodic Announcements Repeat Frequency(sec): <u>0</u> Announcements <input type="text"/> ▾ Prompt: <input type="text"/> If not answered Destination: <u>Hangup</u> ▾ |

| Item | Explanation |
|---------------------|--|
| Agent TimeOut (sec) | The next Agent will ring after this time. |
| Auto Pause | Pause the Agent when it fails to answer the first call. |
| Wrap-Up-Time (sec) | Wrap-up time between the first answer and second answer. (Default is 0, which means no wrap-up time.) |
| Max Wait Time (sec) | Maximum wait time for callers in the queue. |
| Max Callers | Maximum number of callers who are allowed to wait in the queue. (Default is 0, which means no limitation.) |
| Join Empty | Allow callers to enter the Queue when no Agents are available. If this option is not defined, callers will not be able to enter Queues with no available agents. |
| Leave When Empty | All callers in the Queue will be moved out when new caller cannot enter the Queue. This option cannot be used with Join Empty simultaneously. |
| Auto Fill | Callers will be distributed to Agent automatically. |

| | |
|-----------------------|---|
| Report Hold Time | Report the hold time of the next caller for Agent when the Agent is answering the call. |
| Frequency(sec) | Repeat frequency to announce the hold time for callers in the Queue. ("0" means no announcement). |
| Announce Hold Time | Announce the hold time. Announce (yes), not announce(no) or announce once(once), it will not be announced when the hold time is less than 1 minute. |
| Repeat Frequency(sec) | Interval time to play the voice menu for callers. ("0" mean not to play). |
| Announcement Prompt | Select a prompt as the Announcements Prompt from the IVR Prompts. |

Chapter 4 Advanced

4.1 Options

Options include local extension settings and new extension default settings **【General】** , caller ID setting **【Global Analog Setting】** , and NAT FAX setting **【Global SIP Setting】** .

4.1.1 General

Click **【General】** to display the dialog as shown below:

| | | |
|---------|------------------------|---------------------|
| General | Global Analog Settings | Global SIP Settings |
|---------|------------------------|---------------------|

Local Extension Settings

Operator Extension: ▼

Global RingTime Set(sec):

Enable Transfer:

Enable Music On Ringback:

Record Format: ▼

Default Settings for New User

SIP: IAX2: Web Manager: Call Waiting:

Agent: Voicemail: Delete VMail: VM Password:

NAT: Transport: ▼ SRTP:

Audio Codecs

alaw ulaw G.722 G.729 G.726 GSM Speex

Extension Preferences

User Extensions to

Conference Extensions to

IVR Extensions to

Queue Extensions to

RingGroup Extensions to

PagingGroup Extensions to

Reset

| Item | Explanation |
|---------------------------|-------------------------------------|
| Operator Extension | Set extension number for Operator. |
| Global Ring Time Set | Set Ring time for every extension. |
| Enable Transfer | Check to enable Transfer. |
| Enable Music On Ring back | Check to enable Music On Ring back. |

| | |
|-------------------------------|--|
| Record Format | Set the format for recording files. (GSM/WAV only) |
| Default Settings for New User | Check to enable the default settings. |
| Extension Preferences | Set the rule for extensions. |

4.1.2 Global Analog Settings

Click **【Advance】** -> **【Options】** -> **【Global Analog Settings】**:

| | | |
|---------|-------------------------------|---------------------|
| General | Global Analog Settings | Global SIP Settings |
|---------|-------------------------------|---------------------|

| | |
|-------------------------|-------------------------------------|
| Caller ID Detect | |
| Caller ID Detection: | <input checked="" type="checkbox"/> |
| Caller ID Signalling: | Bell-US |
| Caller ID Start: | Ring |
| CID Buffer Length: | 2500 |

| | |
|-----------------------|-------------------------------------|
| General | |
| Opermode: | FCC |
| ToneZone: | China |
| Relax DTMF: | <input type="checkbox"/> |
| Send Caller ID After: | 1 |
| Echo Cancel: | <input checked="" type="checkbox"/> |
| Echo Training: | 800 (yes/no/number) |
| Busy Detection: | <input checked="" type="checkbox"/> |
| Busy Count: | 3 |

| Item | Explanation |
|---------------------|--|
| Caller ID Detection | Enable/Disable Caller ID Detection |
| Caller ID Signaling | Select the mode of Caller ID Signaling. |
| Caller ID Start | Ring--Caller ID start before ring. Polarity--Caller ID start when polarity reversal starts. |
| CID Buffer Length | Default CID Buffer Length |
| Opermode | Set the Opermode for FXO/GSM Ports. |
| ToneZone | Select the ToneZone in your country. |
| Relax DTMF | Enable/Disable Relax DTMF inspection. |
| Echo Cancel | Enable/Disable Echo Cancel |
| Echo Training | Set Echo Training (default unit: ms) |

| | |
|----------------|---|
| Busy Detection | Enable/Disable Busy Detection. |
| Busy Count | Count the Busy Detection. It will be active when enabling Busy Detection. |

4.1.3 Global SIP Settings

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

General
Global Analog Settings
Global SIP Settings

General

Enable

 Enable

UDP Port: 5060
 TCP Port: 5060
 TLS Port: 5061 [Download CA](#)
 Start RTP Port: 10000
 End RTP Port: 20000
 DTMF Mode: Auto ▼
 Max Registration/Subscription Time(sec): 3600
 Min Registration/Subscription Time(sec): 60
 Default Incoming/Outgoing Registration Time(sec): 60

| Item | Explanation |
|---|--|
| UDP Port to bind to | SIP standard port is 5060 |
| TCP Port | Default TCP port is 5060 |
| TLS Port | Default TLS port is 5061 |
| Start RTP Port | RTP port range |
| End RTP Port | RTP port range |
| DTMF Mode | Set default DTMF mode for sending DTMF, support auto, RFC2833, inband, info. Default: RFC 2833 |
| Max Registration/Subscription Time | Maximum duration (in seconds) of incoming registrations/subscriptions is 3600 seconds by default |
| Min Registration/Subscription Time | Minimum duration (in seconds) of registrations/subscriptions is 60 seconds by default |
| Default Incoming/Outgoing Registration Time | Default duration (in seconds) of incoming/outgoing registration |

| NAT Support |
|--|
| External IP: _____ External Host: _____ External Refresh(sec): _____ Local Network Address: _____ |

| Item | Explanation |
|-----------------------|--|
| External IP | Address that we're going to put in outbound SIP messages if we're behind a NAT |
| External Host | Alternatively, you can specify an external host, and Asterisk will perform DNS queries periodically. Not recommended for production environments! Use external IP instead |
| External Refresh | How often to refresh external host if used. You may specify a local network in the field below |
| Local Network Address | 192.168.0.0/255.255.0.0' : All RFC 1918 addresses are local networks, '10.0.0.0/255.0.0.0' : Also RFC1918, '172.16.0.0/12' : Another RFC1918 with CIDR notation, '169.254.0.0/255.255.0.0' : Zero conf local network |

| T.38 Fax Passthrough Support |
|--|
| T.38 Fax (UDPTL) Passthrough: <input type="checkbox"/> |

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

| Type of Service |
|--|
| <div style="text-align: right; margin-bottom: 5px;">TOS for Signalling packets: <input type="text" value=""/> <input type="button" value="v"/></div> <div style="text-align: right; margin-bottom: 5px;">TOS for RTP audio packets: <input type="text" value="ef"/> <input type="button" value="v"/></div> <div style="text-align: right; margin-bottom: 5px;">TOS for RTP video packets: <input type="text" value=""/> <input type="button" value="v"/></div> <div style="text-align: right; margin-bottom: 5px;">Enable Relaxed DTMF: <input checked="" type="checkbox"/></div> <div style="text-align: right; margin-bottom: 5px;">RTP TimeOut: <input type="text" value=""/></div> <div style="text-align: right; margin-bottom: 5px;">RTP HoldTimeOut: <input type="text" value=""/></div> <div style="text-align: right; margin-bottom: 5px;">Trust Remote Party ID: <input type="checkbox"/></div> <div style="text-align: right; margin-bottom: 5px;">Send Remote Party ID: <input type="checkbox"/></div> <div style="text-align: right; margin-bottom: 5px;">Generate In-Band Ringing: <input type="text" value=""/> <input type="button" value="v"/></div> <div style="text-align: right; margin-bottom: 5px;">Add 'user=phone' to URI: <input type="checkbox"/></div> <div style="text-align: right;">Send Compact SIP Headers: <input type="checkbox"/></div> |

| Item | Explanation |
|---------------------------|--|
| TOS for Signaling packets | Sets Type of Service for SIP packets |
| TOS for RTP audio packets | Sets Type of Service for RTP audio packets |
| TOS for RTP video packets | Sets Type of Service for RTP video packets |
| Enable Relaxed DTMF | Relax DTMF handling |
| RTP Time Out | Terminate call if 60 seconds of no RTP activity when we're not on hold |
| RTP Hold Time Out | Terminate call if 300 seconds of no RTP activity when we're on hold (must be > RTP time out) |
| Trust Remote Party ID | If Remote-Party-ID should be trusted |
| Send Remote Party ID | If Remote-Party-ID should be sent |
| Generate In-Band Ringing | If we should generate in-band ringing always, use 'never' to never use in-band signaling, even in cases where some buggy devices might not render it. Default: never |
| Add 'user=phone' to URI | If checked, 'user=phone' is added to URI that contains a valid phone number |
| Send Compact SIP Headers | Send compact sip headers |

| Outbound SIP Registrations |
|---|
| Register TimeOut: <input style="width: 50px;" type="text"/> Register Attempts: <input style="width: 50px;" type="text"/> |

| Codecs |
|---|
| Disallowed Codecs: <input style="width: 80px;" type="text" value="all"/> Allowed Codecs: <input style="width: 80px;" type="text" value="alaw,ulaw"/> Edit |

| Item | Explanation |
|-------------------|---|
| Register Time Out | Retry registration calls at every 'x' seconds (default 20) |
| Register Attempts | Number of registration attempts before we give up; 0 = continue forever |
| Disallowed Codecs | Default is disallowed = all |
| Allowed Codecs | Choose the codec that system allows |

| | |
|--|---|
|  Note | <p>In the extension “Audio Codecs Configure” the priority is higher than “Allowed Codec” items, “Allowed Codec” items are the default codec setting, if user marks the extension “Audio Codecs Configure”, then system will use it first, if not system will let the “Allowed Codecs” define what codec can be used in extension.</p> |
|--|---|

4.2 VoiceMail

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

General

General

Email Settings

VoiceMail Reference

Max Greeting Time(sec):
 Dial "0" for Operator:

Voice Message Options

Message Format:
 Maximum Messages:
 Max Message Time(min):
 Min Message Time(sec):

Playback Options

- Say Message CallerID
- Say Message Duration
- Play Envelope
- Allow Users to Review

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

Email Settings

Email Settings

General

Email Settings

Template for Voicemail Emails

Attach voicemail to email

Sender Name test

From pbx@zycoo.com

Subject New Voicemail from \${VM_CALLERID}

Message Hello \${VM_NAME}, you received a message lasting
\${VM_DUR} at \${VM_DATE} from,
(\${VM_CALLERID}).

Save

Cancel

Template Variables:

\${VM_NAME} : Recipient's first name and last name
 \${VM_DUR} : The duration of the voicemail message
 \${VM_MAILBOX} : The recipient's extension
 \${VM_CALLERID} : The Caller ID of the person who left the message
 \${VM_MSGNUM} : The message number in your mailbox
 \${VM_DATE} : The date and time the message was left

| Item | Explanation |
|---------------------------|---|
| Attach voicemail to Email | The voicemail will be sent as attachment to the user's Email. |
| Sender Name | The sender's name will be displayed when you receive the Email. |
| From | Mailbox to send email |
| Subject | Subject of the Email. |
| Message | Input the Email template. |

4.3 SMTP Setting

SMTP Settings

SMTP Settings:

SMTP Server: _____
 Port: 25
 SSL/TLS:
 Enable SMTP Authentication
 Username: _____
 Password: _____

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

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

Send Test X

Email Address: _____

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

4.4 Email to Fax

Users can send fax by Email. Please configure as shown below.

Click **【Advanced】** -> **【Email to Fax】**

Email to Fax

Enable:

Username: _____

Password: _____

IMAP Server: _____

SSL/TLS:

Access Code: _____

Dial Plan: ▼

Check “Enable”, input user name, password and IMAP Server(server format: imap.XX.com), select the Dial Plan and then “Save” and “Activate”.

Practical Case:

Send a fax to telephone number 85337096: In Dial Plan 1, there is prefix “9” before the telephone number; you need to input the **【Access Code】** : 985337096 and take it as the subject when sending Email. Then the fax will be sent by Email as attachment.

If you need to dial the extension when sending fax, e.g. fax number: 85337096 ext.800, you need to use the **【Access Code】** : 985337096-800 as subject.

4.5 Music Settings

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

Click **【Music Settings】** to display the dialog as below:

Music Settings:

Music Settings

Music Settings
Music Management

Music On Hold Reference

Music:

Music On Ringback Reference

Music:

Music On Queue Reference

Music:

Please define different music files for different music folders.

Music Management:

Music Management

Music Settings
Music Management

Music Management

Select Music Directory:

Files:

Upload Music File

Select Music Directory:

Note: The sound file must be wav(16bit/8000Hz/Single), gsm, ulaw or alaw!
The size is limited in 15MB!.

Please choose file to upload:

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

| | |
|--|-------|
| | 15MB. |
|--|-------|



The sound file must be **wav** (16bit/8000Hz/Single), **gsm, ulaw or alaw** !! The size is limited in **15MB**

4.6 DISA

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

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

New DISA X

Name:

PIN: Without PIN

Response Timeout(s):

Digit Timeout(s):

Extension for this DISA(Optional):

Allow Outbound Route

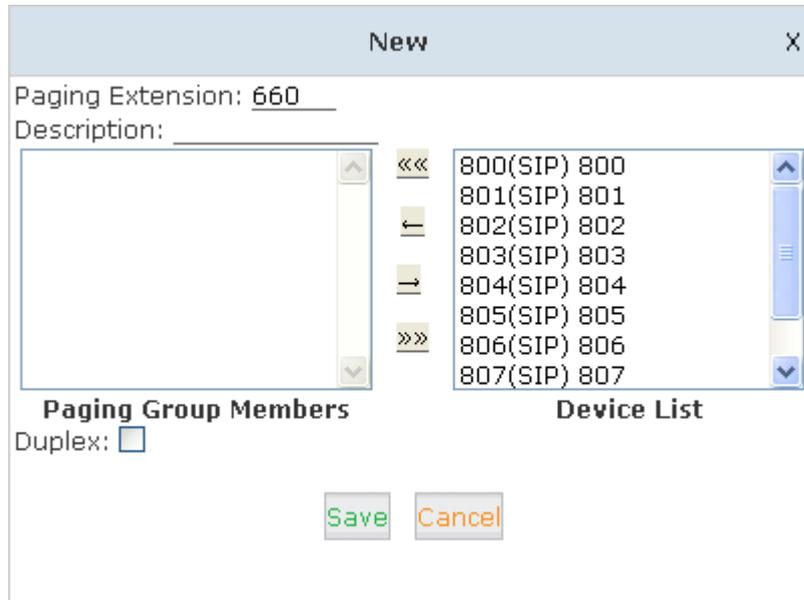
Select DialPlan

| Item | Explanation |
|-----------------------------------|---|
| Name | Define a name for DISA. |
| PIN Set | User will be prompted to input this number when PIN Authentication is needed. |
| Record in CDR | Check to record. |
| Response Timeout(sec) | The maximum time for waiting before hanging up if the dialed number is incomplete or invalid. Default is 10 seconds |
| Digit Timeout(sec) | The maximum interval time between digits when typing extension number is 5 seconds by default. |
| Extension for this DISA(Optional) | If you want to access DISA by dialing an extension, you can define an extension number for this DISA. |
| Select Dial Plan | Select the Dial Plan for this DISA. |

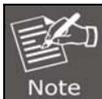
4.7 Paging And Intercom

Paging and Intercom is used for calling a paging extension; all terminals which support this function will be picked up automatically and listen; meanwhile, it supports duplex.

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



| Item | Explanation |
|----------------------|--|
| Paging Extension | The number users will dial to page this group. |
| Description | Provide a descriptive title for this Page Group. |
| Paging Group Members | Selected device(s) on this page |
| Device List | Select Device(s) to page. |
| Duplex | Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it like an "instant conference". |



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

4.8 PIN Set

Monitor is used for recording the defined extensions.

Click **【Monitor】** --- **【New Monitor】** to display the dialog below:

X
New Monitor

Extension:

Monitoring Time

Always Monitor:

Start Time: : : End Time: : :

Start Day: End Day:

Monitor Settings

Inbound Record: Outbound Record:

| Item | Explanation |
|--------------|-----------------------------------|
| PIN Set Name | Define the name for this PIN Set. |
| PIN List | Define PIN codes in this list. |

4.9 Call Recording

Call Recording is used for recording extension. Please configure it as shown below:

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

X
New Call Recording

Extension:

Call Recording Time

Always Recording:

Start Time: : : End Time: : :

Start Day: End Day:

Call Recording Settings

Inbound Record: Outbound Record:

Reference:

| Item | Explanation |
|---------------------|------------------------------------|
| Extension | Define an extension for recording. |
| Call Recording Time | Set the time to record. |
| Inbound Record | Check to record inbound calls. |

| | |
|-----------------|---------------------------------|
| Outbound Record | Check to record outbound calls. |
|-----------------|---------------------------------|

4.10 Speed Dial

Please configure as shown below:

Click **【Advanced】** -> **【Speed Dial】** -> **【New Speed Dial】** :

New Speed Dial
X

Notice: Don't forget to add the outbound dial prefix if you would like to dial an outside number

Source Number: _____

Destination Number: _____

Save
Cancel

E.g. prefix is *99, speed number is 00, destination telephone number is 85337096.

When dialing *9900, the call is going to 85337096 automatically.

4.11 Smart DID

Smart DID: After extension user makes an outbound call, the call is ringing back to Planet IP PBX, and directed to the one who made the last call. Please configure it as shown below:

Click **【Advanced】** -> **【Smart DID】** :

Smart DID

Smart DID

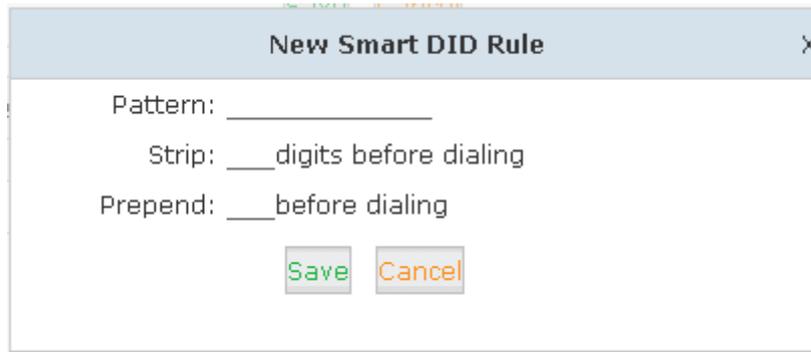
Enable:

Save
Cancel

| Smart DID Rules List | | New Smart DID Rule | |
|----------------------|-------|------------------------------------|--|
| Pattern | Strip | Prepend | Options |
| 1 | X. | | Edit Delete |

Check “Enable” and “Save” to make this function activates.

Click **【New Smart DID Rule】** to display the following diagram:



New Smart DID Rule [X]

Pattern: _____

Strip: ___ digits before dialing

Prepend: ___ before dialing

[Save] [Cancel]

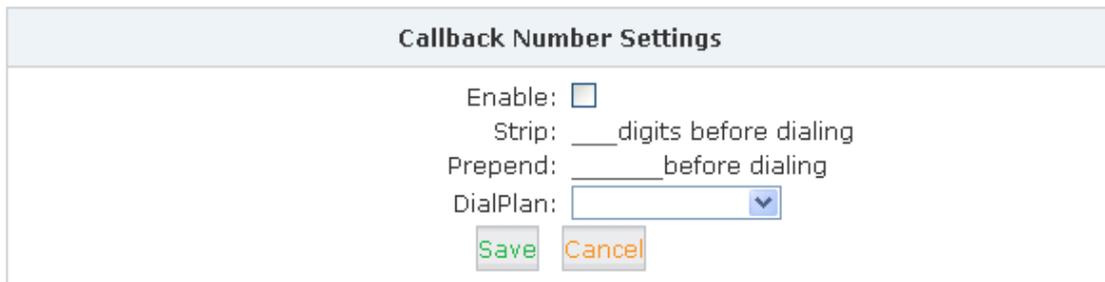
Input the pattern and define how many digits need to be striped or prepend, and then click "Save"--"Activate".

4.12 Call Back

When user makes calls by the callback number to Planet IP PBX, the call will be hung up automatically. Then the PBX will call back this number and forwarded to define destination after the call is connected. Please configure it as shown below:

Click **【Advanced】** -> **【Callback】** :

Callback Number Settings



Callback Number Settings

Enable:

Strip: ___ digits before dialing

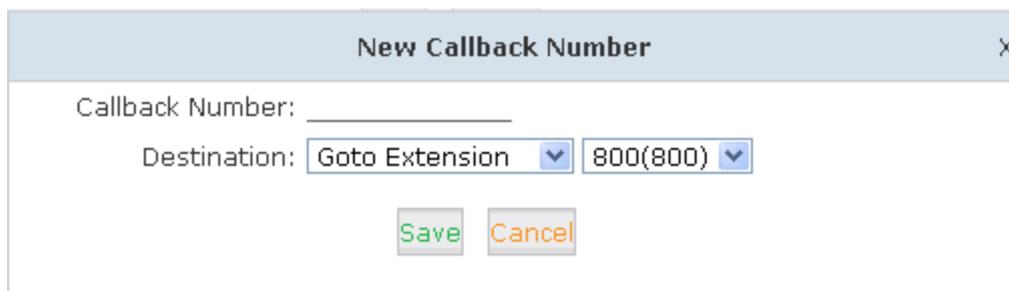
Prepend: ___ before dialing

DialPlan: _____ [v]

[Save] [Cancel]

| List of Callback Number | | | New Callback Number |
|-----------------------------|-------------|---------|---------------------|
| Callback Number | Destination | Options | |
| No Callback Number defined! | | | |

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



New Callback Number [X]

Callback Number: _____

Destination: [Goto Extension [v]] [800(800) [v]]

[Save] [Cancel]

Input callback number and define the destination.

4.13 Phone Book

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

Click **【Advanced】** -> **【Phone Book】** :

Phone Book

| Phone Book | | | | Create Contact |
|--------------------------|----------------------|---------------------------------------|---|--|
| Name: | <input type="text"/> | <input type="button" value="Search"/> | <input type="button" value="Show All"/> | <input type="button" value="Delete Selected"/> |
| <input type="checkbox"/> | Name | Phone Number | Options | |
| <input type="checkbox"/> | 1 David | 85362145 | <input type="button" value="Edit"/> | <input type="button" value="Delete"/> |

| Item | Explanation |
|----------|---|
| Search | Search by name |
| Show All | All contacts will be displayed in the following list. |

Click **【Create Contact】** to see the following diagram:

Create Contact
X

Name:

Phone Number:

| Item | Explanation |
|--------------|---|
| Name | Input contact's name. (Letter or figure only). |
| Phone Number | Input Phone Number of contact. (IDD Number is available). |

Phone book is for the incoming call to use; if the incoming caller ID matches the number in Phone book, it will display the name defined in Phone book.

For example, Name: David Number: 123456789.

When system receives the call 123456789, the extension answers this call with "David" being displayed.

4.14 Feature Codes

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

| Feature Codes Management | |
|----------------------------|--|
| Call Parking | Extension to Dial for Parking Calls: <u>700</u> Extension Range to Park Calls: <u>701-720</u> Call Parking Time(sec): <u>45</u> Parking Hints: <input type="checkbox"/> |
| Pickup Call | Pickup Extension: <u>*8</u> Pickup Specified Extension: <u>**</u> |
| Transfer | Blind Transfer: <u>#</u> Attended Transfer: <u>*2</u> Disconnect Call: <u>*</u> Timeout for answer on attended transfer(sec): <u>15</u> |
| One Touch Recording | One Touch Recording: <u>*1</u> |
| Call Forward | Enable Forward All Calls: <u>*71</u> Disable Forward All Calls: <u>*071</u> Enable Forward on Busy: <u>*72</u> Disable Forward on Busy: <u>*072</u> Enable Forward on No Answer: <u>*73</u> Disable Forward on No Answer: <u>*073</u> |

| Item | Explanation |
|-------------------------------------|--|
| Extension to Dial for Parking Calls | Define an extension for parking calls. |
| Extension Range to Park Calls | Define the extension range for parking calls. (e.g. 701-720) |
| Call Parking Time(sec) | Define the time for parking calls. Planet IP PBX will call the extension again if parking is over time. |
| Pickup Extension | Define an extension for pickup. |
| Pickup Specified Extension | Pick up the specified extension. Default: Dial**+extension number to pick up the specified extension |
| Blind Transfer | Allow unattended or blind transfers. It works like this: While on a conversation with A, you dial the blind transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number (B's number) and A is put through to B immediately. Your line is off. The caller ID displayed to B is exactly the same as the caller ID presented to you. |
| Attended Transfer | Allow attended transfer or supervised transfer. It works like this: |

| | |
|---|--|
| | While on conversation with A, you dial the Attended Transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number (B's number) and talk with B to introduce the call, then you can hang up and A will be connected with B. In case B does not want to answer the call, he/she simply hangs up and you will be back to your original conversation. |
| Disconnect Call | Disconnect the current transfer call (for Attended transfer). |
| Timeout for answer on attended transfer (sec) | Set the timeout value |
| One Touch Recording | Configure the function key for One Touch Recording |
| Call Forward | Enable/Disable Call Forward and the settings of function keys for different forward modes. |
| Do Not Disturb | Enable/Disable "Do Not Disturb" |
| Spy | Configure the function keys for spy modes. |
| Blacklist | Add/Delete blacklisted number. |
| Voicemail | Configure the function keys for entering voicemail and check extension voicemail. |
| Invite Participant | In conference, the administrator can invite people into the conference by dialing "0". After pressing "0", you will get dial tone, and you can dial to invite people. After the call is connected, please press ** to direct the people into the conference, or *# to hang up the current call and return to the conference. |
| Create Conference | During the call, you can dial *0 to forward to the conference with the callee. |
| Return to conference with participant | In conference, the administrator can dial "0" to invite people into the conference. After pressing "0", you will get dial tone, and you can dial to invite the participant; when the call is connected, dial "***" to return to the conference with invited participant. |
| Return to conference without participant | In conference, the administrator can dial "0" to invite people into the conference. After pressing "0", you will get dial tone, and you can dial to invite the participant. When the call is connected, you can dial "**#" to hang up and return the conference yourself. |
| Pause Queue Member Extension | Pause the agent, and the agent cannot receive the call. |
| Unpause Queue Member Extension | Unpause the agent, and the agent can receive the call. |
| Others | Function key for Intercom/ Paging/ Directory |

Chapter 5 Network Settings

5.1 Network

You can configure the WAN Port, and define the Virtual Interface.

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

IPv4 Settings
IPv6 Settings
VLAN Settings

Ethernet Port Setup

| | |
|----------------|--|
| IP Assign: | <input type="text" value="Static"/> |
| Hostname: | <input type="text" value="IPPBX"/> |
| IP Address: | <input type="text" value="192.168.1.198"/> |
| Subnet Mask: | <input type="text" value="255.255.255.0"/> |
| Gateway: | <input type="text" value="192.168.1.254"/> |
| Primary DNS: | <input type="text" value="192.168.1.254"/> |
| Alternate DNS: | <input type="text"/> |

Virtual Interface

| | | | |
|--|----------------------|----------------|----------------------|
| <input type="checkbox"/> IP AddressV1: | <input type="text"/> | Subnet MaskV1: | <input type="text"/> |
| <input type="checkbox"/> IP AddressV2: | <input type="text"/> | Subnet MaskV2: | <input type="text"/> |

Reference

| Item | Explanation |
|-------------------|--|
| IP Assign | Static/ DHCP/PPOE supported. |
| Virtual Interface | Define the virtual interface for WAN Port. |

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

IPv4 Settings
IPv6 Settings
VLAN Settings

| | |
|----------------|-------------------------------------|
| Enable: | <input checked="" type="checkbox"/> |
| IPv6 Address: | <input type="text"/> |
| Prefix Length: | <input type="text"/> |
| Gateway: | <input type="text"/> |
| Primary DNS: | <input type="text"/> |
| Alternate DNS: | <input type="text"/> |

IPv6 Reference:

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

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

IPv4 Settings
IPv6 Settings
VLAN Settings

VLAN 1

Enable:

VLAN ID: _____

VLAN IP Address: _____

Subnet Mask: _____

VLAN 2

Enable:

VLAN ID: _____

VLAN IP Address: _____

Subnet Mask: _____

VLAN Reference:

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

5.2 Static Routing

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

New Static Routing X

Destination Network: _____

Subnet Mask: _____

Gateway: _____

Save
Cancel

| Item | Explanation |
|-------------|---|
| Destination | Set destination network for static routing. |
| Subnet Mask | Set subnet mask of the destination network. |
| Gateway | Define the gateway accessing the destination network. |

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

Routing Table

Static Routing
Routing Table

Routing Table:

Kernel IP routing table

| Destination | Gateway | Genmask | Flags | Metric | Ref | Use | Iface |
|-------------|---------------|---------------|-------|--------|-----|-----|-------|
| 0.0.0.0 | 192.168.1.254 | 0.0.0.0 | UG | 0 | 0 | 0 | ETH |
| 192.168.1.0 | 0.0.0.0 | 255.255.255.0 | U | 0 | 0 | 0 | ETH |

5.3 VPN Server

Planet IP PBX supports three kinds of VPN servers: L2TP, PPTP and OpenVPN.

Click **【Network Settings】** -> **【VPN Server】**:

VPN Server
VPN Users Management

VPN Server

L2TP
 PPTP
 OpenVPN

Enable:

Remote Start IP: _____

Remote End IP: _____

Local IP: _____

Primary DNS: _____

Alternate DNS: _____

Authentication Method: chap pap

Debug:

Save
Cancel

Reference:

| Item | Explanation |
|-----------------|--|
| VPN Server Mode | Three kinds of VPN servers -- L2TP, PPTP and OpenVPN -- supported (Only one mode can be enabled simultaneously). |
| Enable | Enable/Disable VPN Server |

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

VPN Users Management

VPN Server
VPN Users Management

| List of VPN Users | | New VPN User | |
|-------------------|----------|------------------------------|---|
| # | Username | Availability | Options |
| 1 | test | yes | Edit Delete |

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

When the mode is OpenVPN server, click **【Network Settings】** -> **【VPN Server】** -> **【OpenVPN Certificate Download】**:

VPN Server
VPN Users Management

VPN Server

L2TP
 PPTP
 OpenVPN

Enable:

Certificate: None [Create](#) [Delete](#)

Port:

Protocol:

TLS-Server:

Remote Network: /

Route: /

Client-to-Client:

Save
Cancel

Status: L2TP (Disabled)

This page is used for management of OpenVPN certificate file.

5.4 VPN Client

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

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

VPN Client

L2TP
 PPTP
 OpenVPN
 N2N

Enable:

Enable 40/128-bit encryption for MPPE:

Server Address: 192.168.100.100

Username: admin

Password: •••••

```
Status:pptp client Connect: ppp1 <--> /dev/pts/2
pptp client sh: can't execute '/sbin/ip': No such file or directory
pptp client sh: can't execute '/sbin/ip': No such file or directory
```

Reference:

| Item | Explanation |
|------------|--|
| VPN Client | Four kinds of VPN Clients supported: L2TP, PPTP, OpenVPN and N2N (Only one mode can be enabled simultaneously) |
| Enable | Enable/Disable VPN Client |

5.5 DHCP server

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

DHCP Server Settings

Enable:

Start IP: 192.168.1.101

End IP: 192.168.1.200

Subnet Mask: 255.255.255.0

Gateway: 192.168.1.1

Primary DNS: 61.139.2.69

Lease Time(min): 1440

TFTP Server: _____

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

DHCP Server
DHCP Client List
Static MAC

DHCP Client List:

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

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

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

Click **【Network Settings】** -> **【DHCP Server】** -> **【Static MAC】** -> **【New Static MAC】** :

New Static MAC X

MAC Address:

IP Address:

5.6 DDNS Settings

After setting DDNS (Dynamic Domain Network Server), Planet IP PBX settings will be visited remotely. Click **【Network Settings】** -> **【DDNS Settings】**:

DDNS Settings

Enable:

Enable EasyDDNS:

Easy Domain:

DDNS Server:

Username:

Password:

Domain:

Planet supports DDNS provided by Planet DDNS / Dyndns.org / No-ip.com / zoneedit.com.

5.7 SNMPv2 Settings

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

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

SNMPv2 Settings

| Read Only |
|--|
| <div style="text-align: right; margin-right: 20px;">Enable: <input checked="" type="checkbox"/></div> <div style="text-align: right; margin-right: 20px;">RO Community: <input type="text" value="public"/></div> <div style="text-align: right; margin-right: 20px;">RO Network: <input type="text" value="____"/> / <input type="text" value="____"/></div> |
| Read and Write |
| <div style="text-align: right; margin-right: 20px;">Enable: <input checked="" type="checkbox"/></div> <div style="text-align: right; margin-right: 20px;">RW Community: <input type="text" value="private"/></div> <div style="text-align: right; margin-right: 20px;">RW Network: <input type="text" value="____"/> / <input type="text" value="____"/></div> |
| <input type="button" value="Save"/> <input type="button" value="Cancel"/> |

Reference

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

5.8 Troubleshooting

You can ping other network devices through Planet IP PBX and track network routing by command "Traceroute" . Click **【Network Settings】** -> **【Troubleshooting】** :

Troubleshooting

Ping
Traceroute

Ping 192.168.1.254 Packets: 4

```

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

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

Chapter 6 Security

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

6.1 Network And Country

Click **【Security】** -> **【Firewall】**

Firewall

Command: iptables Run

Result:

IP Tables List:

```
Chain INPUT (policy ACCEPT)
target      prot opt source                destination

Chain FORWARD (policy ACCEPT)
target      prot opt source                destination

Chain OUTPUT (policy ACCEPT)
target      prot opt source                destination
```

| Iptables Command | Explanation |
|--|---|
| Check iptables list | iptables -L -n |
| Clear iptables list | iptables -F |
| Deny an IP(192.168.0.3 | iptables -A INPUT -s 192.168.0.3 -j DROP |
| Deny every IP to access 80 port | iptables -A INPUT -p tcp --dport 80 -j DROP |
| Deny IP (192.168.0.3) to access port 80 | iptables -A INPUT -s 192.168.0.3 -p tcp --dport 80-j DROP |

6.2 Service

【Service】 : Settings of SSH/FTP and HTTP Port.

Click 【Security】 -> 【Service】 :

Service Settings

Service Settings

Enable SSH: Port: 22

Enable FTP: Port: 21

HTTP Port: 80

Enable SSH to login background management system through SSH.

Enable FTP to allow uploading files to system through FTP.

Chapter 7 Report

7.1 Record List

Check recordings of specified extension or conference here, or delete the recording file.

【Record List】 :

| | | |
|---|------------|--|
| Call Recording | Conference | One Touch Recording |
| Extension: <input type="text"/> <input type="button" value="Delete"/> | | |
| Start Date: <input type="text"/> <input type="text"/> <input type="text"/> End Date: <input type="text"/> <input type="text"/> <input type="text"/> <input type="button" value="Filter"/> | | |
| List of Recording Files | | <input type="button" value="Delete Selected"/> |
| <input type="checkbox"/> | Caller ID | Destination ID |
| | | Date |
| | | Options |

【Conference】 :

| | | |
|---|-----------------|--|
| Call Recording | Conference | One Touch Recording |
| Start Date: <input type="text"/> <input type="text"/> <input type="text"/> End Date: <input type="text"/> <input type="text"/> <input type="text"/> <input type="button" value="Filter"/> | | |
| List of Conference Record Files | | <input type="button" value="Delete Selected"/> |
| | | <input type="button" value="Delete All"/> |
| <input type="checkbox"/> | Conference Room | Date |
| | | Options |

【One Touch Recording】

| | | |
|---|------------|--|
| Call Recording | Conference | One Touch Recording |
| Extension: <input type="text"/> <input type="button" value="Delete"/> | | |
| Start Date: <input type="text"/> <input type="text"/> <input type="text"/> End Date: <input type="text"/> <input type="text"/> <input type="text"/> <input type="button" value="Filter"/> | | |
| List of Recording Files | | <input type="button" value="Delete Selected"/> |
| <input type="checkbox"/> | Caller ID | Destination ID |
| | | Date |
| | | Options |

7.2 Call logs

Check call logs by caller ID or callee ID.

Click 【Report】 -> 【Call Logs】 :

Call Logs

| | | | |
|-------------|--|---|---------------------------------------|
| Start Date: | <input type="text"/> <input type="text"/> <input type="text"/> | Field: <input type="text"/> | <input type="button" value="Filter"/> |
| End Date: | <input type="text"/> <input type="text"/> <input type="text"/> | <input type="button" value="Download"/> | <input type="button" value="Delete"/> |
| Call Start | Caller ID | Destination ID | Account Code |
| | | Duration(sec) | Disposition |


 Note

Duration in the call logs is not really charged duration. If you need billing, PSTN must support polarity reversal function, and meanwhile, you must configure relevance parameters of polarity reversal in trunk configuration for Planet IP PBX.

7.3 System logs

Click **【Report】** -> **【System Logs】** , and you can download/ delete the system logs.

System Logs

Enable System Log: Enable PBX Log:
 Enable PBX Debug Log: Enable Access Log:

Save
Cancel

| List of Logs | | Download Selected | Delete Selected |
|--------------------------|-------------------|-------------------|--|
| <input type="checkbox"/> | Name | Type | Options |
| <input type="checkbox"/> | 1 login201303.log | Login Log | Delete Download |
| <input type="checkbox"/> | 2 login201304.log | Login Log | Delete Download |
| <input type="checkbox"/> | 3 pbx20130311.log | PBX Log | Delete Download |
| <input type="checkbox"/> | 4 pbx20130313.log | PBX Log | Delete Download |
| <input type="checkbox"/> | 5 pbx20130315.log | PBX Log | Delete Download |
| <input type="checkbox"/> | 6 pbx20130319.log | PBX Log | Delete Download |
| <input type="checkbox"/> | 7 pbx20130320.log | PBX Log | Delete Download |

7.4 Data Storage

When you need mass storage of recording files, voicemails, call logs, etc, you can upload these files to FTP server through FTP Data Storage based on the specified time frequency

Click **【System】** -> **【Data Storage】** :

Data Storage
Data Storage Log

FTP Data Storage

Enable:

Server Address: _____

Username: _____

Password: _____

Directory: _____

Automatically upload frequency(day):

Time of automatically upload: :

Forcibly upload when the flash storage is over:

Status: Disabled

Reference

| Item | Explanation |
|--|---|
| Enable | Enable FTP Data Storage. |
| Server Address | Set FTP server address (IP address or domain). |
| User Name | User name for login FTP. |
| Password | Password for login FTP. |
| Directory | Define a directory used for storage on FTP server. |
| Automatically upload frequency (by the day) | Define frequency (by the day) to upload the data. |
| Time of automatically upload | Define the time to upload the data. |
| Forcibly upload when the flash storage is over | Forcibly upload data when flash storage is over the percentage value. |

Check from **【Data Storage Log】** :

Data Storage
Data Storage Log

Data Storage Log

Click **【Refresh】** to refresh data storage log.

Click **【clear】** to clear data storage log.

7.5 Management

【Management】 is used to modify password of Planet system, and the settings of system voice.

Click 【System】 -> 【Management】 :

Management

Change Password

Password: _____
New Password: _____
Retype New Password: _____

Set Language

Set Voice Language: English ▼

【Set Language】 Choose the voice language you want

Set Language

Set Voice Language: English ▼

English

English

中文

Français

Español

Português

Italiano

7.6 Backup

Click 【System】 -> 【Backup】

Backup
Upload Backup File

| List of Backups | | Take a Backup | | |
|-----------------|-------------------------|-------------------------------|-------------------------|------------------------|
| Name | Date | Options | | |
| 1 | backup_2013jan09_135847 | Jan 09, 2013 | Restore | Delete |
| 2 | backup_2013jan09_135854 | Jan 09, 2013 | Restore | Delete |
| 3 | backup_2013may16_160601 | May 16, 2013 | Restore | Delete |

Reference:

| Item | Explanation |
|---------------|---|
| Take a Backup | Take a backup of the current system configuration. |
| Restore | Restore system to the specified backup configuration. |
| Delete | Delete specified backup file. |

Click the download button "" to download the specified backup file and manage locally.

Click **【Upload Backup File】** to upload the backup file here.

Backup
Upload Backup File

Upload Backup File

Note: Don't change the backup file name.

Please choose file to upload:

Click **【browse】** to select the local backup file, and click **【Upload】** to upload the backup file to system.

7.7 Reset & Reboot

If you need to reset the system to factory default or reboot, please click **【System】** -> **【Reset & Reboot】** :

Factory Defaults

Warning: Restore factory settings,will lost all configuration data on the system!

Factory Defaults

Reboot

Warning: Rebooting the system will terminate all active calls!

Reboot

Click **【Factory Default】** to reset the system to factory default.

Click **【Reboot】** to reboot the system.

7.8 Upgrade

7.8.1 WEB Upgrade

Click **【System】** -> **【Upgrade】** -> **【WEB Upgrade】** :

Upgrade System Package

WEB Upgrade TFTP Upgrade

Restore Default Set:

Please choose file to upload:

Click **【Browse】** to select the firmware file, and then click **【Upload】** to upload the selected firmware to system and finish the upgrading automatically.

If check **【Restore Default Set】** , the system will clear all the configuration and reset to factory default.

7.8.2 TFTP Upgrade

Click **【System】** -> **【Upgrade】** -> **【TFTP Upgrade】** :

Upgrade System Package

WEB Upgrade TFTP Upgrade

Restore Default Set:

Enter The Package Name:

TFTP Server IP address:

Reference:

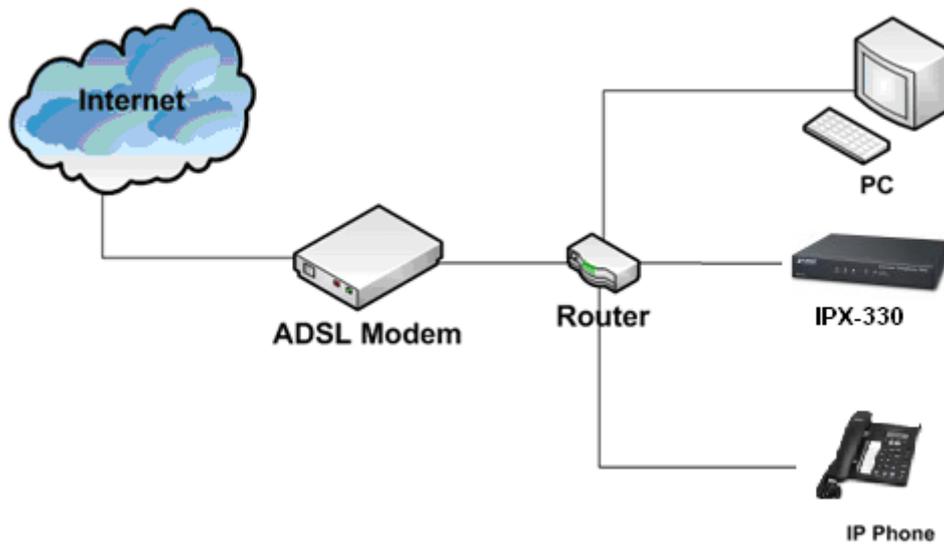
| Item | Explanation |
|------------------------|---|
| Restore Default Set | System will restore to factory defaults after checking this option. |
| Enter The Package Name | Enter the package name for upgrading. |
| TFTP Server IP address | Enter your TFTP server IP address. |

Chapter 8 Operating Instructions

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

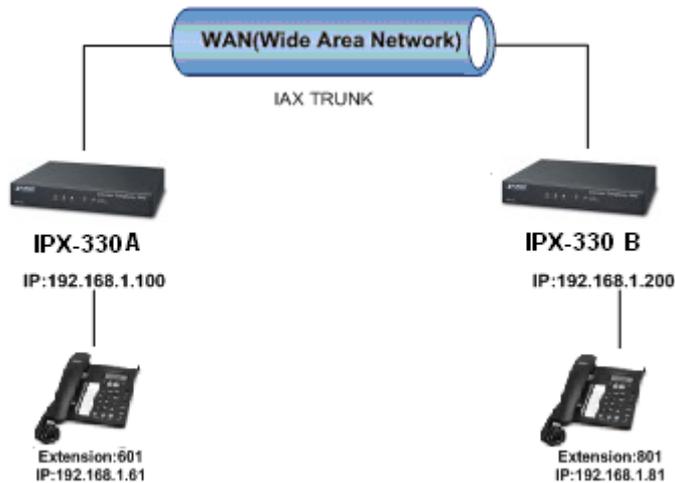
8.1 How to connect the IPX-330 IP PBX to the Internet

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



8.2 How to combine two IPX-330 IP PBX in a different network

Normally, two sets of the IPX-330 are located in different places with different IP addresses for Internet access.



For external line configuration, you must use public IP address.

Take the following instructions as an example:

Register IPX-330-B IP to a trunk of IPX-330-A with authentication.

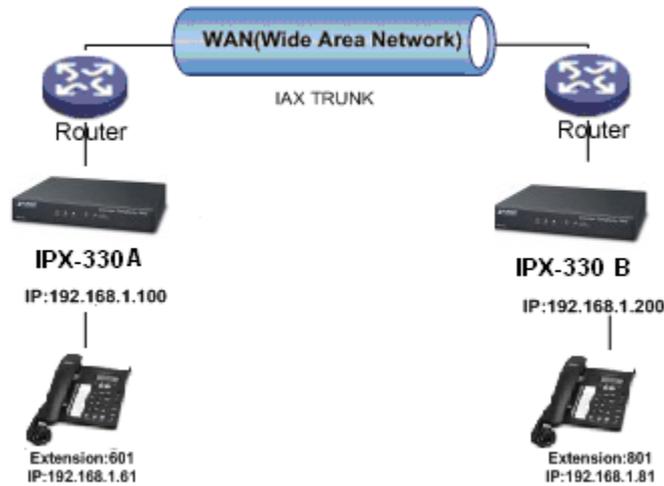
Configuration Rule:

1. IP Phone registers on IPX-330-A as extension 601.
2. Another IP Phone registers on U50-B as extension 801.
3. IPX-330-A IP: 192.168.1.100.
4. IPX-330-B IP: 192.168.1.200.
5. Extension format of IPX-330-A: 6XX.
6. Extension format of IPX-330-B: 8XX
7. Create an extension 888 with password 123456 on IPX-330-B.
8. All extensions on IPX-330-A can call extensions on IPX-330-B with format 8XX.
9. All extensions on IPX-330-B can call extensions on IPX-330-A with format 6XX.

For detailed steps, please take chapter 8.2 as reference.

Two sets of IPX-330 behind router

Sometimes the IPX-330 doesn't have a public IP address, and you have to configure port mapping for your router.



Step1: Configure the mapping rule of IPX-330-A on the router.

The IPX-330-B is connected behind the router, and registers on IPX-330-A through internet. You need to configure the port mapping of IAX2 port(4569) on the router. Then, all data received from eth0 port of router(192.168.1.100:4569) will be sent to IPX-330-A

Now, take the web management panel of AND-4100 router as an example.

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

| Advanced | PORT FORWARDING | | | | | | | | | | | | | | | | | | |
|--------------------------|--|--------------------------|-------------------------|----------------|-------------------------|-------------------|-------------------------|-------------------|---------------|-----------|--------------------------|-----|-------------|-----------|------|-----------|---------------|--------|--|
| Advanced Wireless | Port Forwarding allows you to direct incoming traffic from the WAN side (identified by protocol and external port) to the internal server with a private IP address on the LAN side. The internal port is required only if the external port needs to be converted to a different port number used by the server on the LAN side. A maximum of 80 entries can be configured. | | | | | | | | | | | | | | | | | | |
| Port Forwarding | Select the service name, and enter the server IP address and click "Apply" to forward IP packets for this service to the specified server. Note: Modifying the Internal Port Start or Internal Port End is not recommended. If the External Port Start or the External Port End changes, the Internal Port Start or Internal Port End automatically changes accordingly. | | | | | | | | | | | | | | | | | | |
| DMZ | PORT FORWARDING SETUP | | | | | | | | | | | | | | | | | | |
| Parental Control | <table border="1"> <thead> <tr> <th><input type="checkbox"/></th> <th>Server Name</th> <th>Wan Connection</th> <th>External Port Start/End</th> <th>Protocol</th> <th>Internal Port Start/End</th> <th>Server IP Address</th> <th>Schedule Rule</th> <th>Remote IP</th> </tr> </thead> <tbody> <tr> <td><input type="checkbox"/></td> <td>IAX</td> <td>mer_0_35...</td> <td>4569/4569</td> <td>both</td> <td>4569/4569</td> <td>192.168.1.100</td> <td>Always</td> <td></td> </tr> </tbody> </table> | <input type="checkbox"/> | Server Name | Wan Connection | External Port Start/End | Protocol | Internal Port Start/End | Server IP Address | Schedule Rule | Remote IP | <input type="checkbox"/> | IAX | mer_0_35... | 4569/4569 | both | 4569/4569 | 192.168.1.100 | Always | |
| <input type="checkbox"/> | Server Name | Wan Connection | External Port Start/End | Protocol | Internal Port Start/End | Server IP Address | Schedule Rule | Remote IP | | | | | | | | | | | |
| <input type="checkbox"/> | IAX | mer_0_35... | 4569/4569 | both | 4569/4569 | 192.168.1.100 | Always | | | | | | | | | | | | |
| Filtering Options | <input type="button" value="Add"/> <input type="button" value="Edit"/> <input type="button" value="Delete"/> | | | | | | | | | | | | | | | | | | |
| QoS Configuration | | | | | | | | | | | | | | | | | | | |
| Firewall Settings | | | | | | | | | | | | | | | | | | | |
| DNS | | | | | | | | | | | | | | | | | | | |
| Dynamic DNS | | | | | | | | | | | | | | | | | | | |
| Network Tools | | | | | | | | | | | | | | | | | | | |
| Routing | | | | | | | | | | | | | | | | | | | |

Step2: IPX-330 Configuration

Configure the trunk and dial plan on IPX-330-B, and register IPX-330-B IP to IPX-330-A. The configuration is the same as the above, but you have to replace the public IP address with the internal IP: 192.168.1.21.

Step3: Configure port mapping rule of IPX-330-B on the router

Configure port mapping of IPX-330-B on the router according to Step1.

Step4: Connect two sets of the IPX-330 and make the call

Create extension 601 on IPX-330-A, extension 801 on IPX-330-B, and create the correct outbound rule.



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

8.3 How to resolve the problem about hearing one side only

If the IPX-330 is behind router, to resolve the problem, please set up IP address as shown below:

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

| NAT Support |
|--|
| External IP: _____ External Host: _____ External Refresh(sec): _____ Local Network Address: _____ |

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

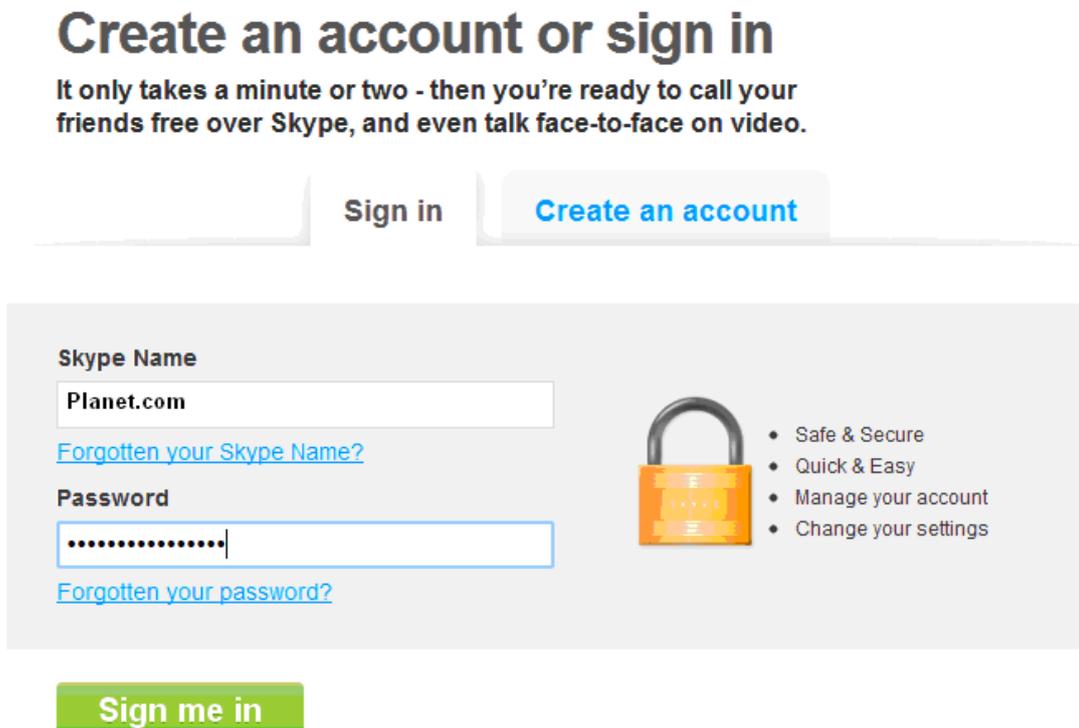
8.4 How to use Skype account in IPX-330

[Answer] :

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

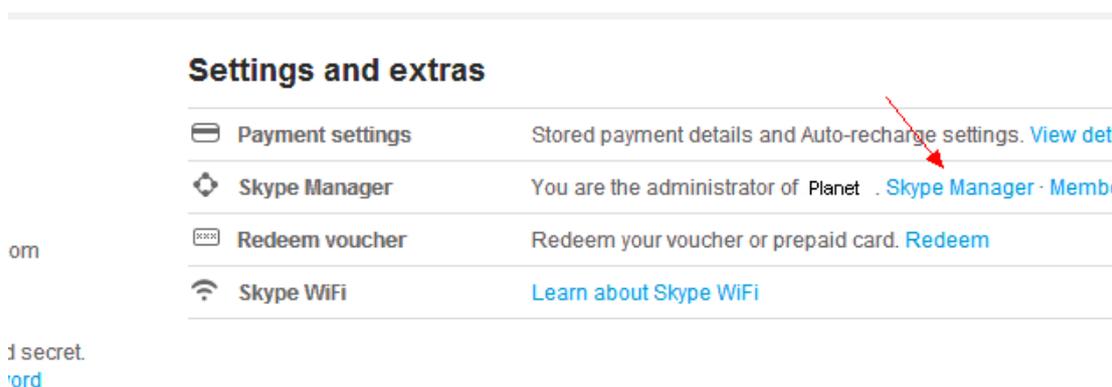
1 <https://login.skype.com>

Sign in with the business account.



The screenshot shows the Skype login interface. At the top, it says "Create an account or sign in" with a subtext: "It only takes a minute or two - then you're ready to call your friends free over Skype, and even talk face-to-face on video." Below this are two buttons: "Sign in" and "Create an account". The "Sign in" button is highlighted. The login form contains a "Skype Name" field with "Planet.com" entered, a "Password" field with masked characters, and a "Sign me in" button. To the right of the form is a padlock icon and a list of benefits: "Safe & Secure", "Quick & Easy", "Manage your account", and "Change your settings".

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



The screenshot shows the "Settings and extras" section of the Skype interface. It lists several options: "Payment settings" (Stored payment details and Auto-recharge settings. View details), "Skype Manager" (You are the administrator of Planet . Skype Manager · Member), "Redeem voucher" (Redeem your voucher or prepaid card. Redeem), and "Skype WiFi" (Learn about Skype WiFi). A red arrow points to the "Skype Manager" link.

1 secret.
ord

David Yao

 Your Skype Name
Planet.com
[Profile details](#)
 Your email
[Email settings](#)
 Your password 
 Keep your password secret.
[Change your password](#)

Settings and extras

| | | |
|---|------------------|---|
|  | Payment settings | Stored payment details and Auto-recharge settings. View details |
|  | Currency | Your currency is set to EUR (Euros). Change |
|  | Skype Manager | You are the administrator of Planet Skype Manager · Member page |
|  | Redeem voucher | Redeem your voucher or prepaid card. Redeem |

3 Please click the Skype connect

Your features

Some features have been suspended

-  Allocate [Skype Credit](#) to your members
-  Set up [Subscriptions](#) for your members
-  Set up [Group video calling](#) for your members
-  Set up [Online Numbers](#) for your members
-  Set up [Call forwarding](#) for your members
-  Set up [Voicemail](#) for your members
-  7 profiles set up for [Skype Connect](#) 

Your members

Your Skype Manager has **2 members**

[Add members](#)

Since you last signed in

No changes since you last logged in.

Still unresolved

[One unresolved invite](#)

-  **Subscriptions**
0 members
-  **Group video calling**
0 members
-  **Voicemail**
0 members
-  **Online Numbers**
0 members
-  **Call forwarding**
0 members
-  **Skype Connect** 
3 profiles

Connect your existing SIP-enabled PBX to Skype with Skype Connect. [Learn more](#)

 Some of your SIP Profiles have been suspended because your Skype Manager has insufficient credit available to pay for the channel subscription. [Buy more credit](#) and the profiles will be reactivated.

Your SIP Profiles

[Set up a SIP Profile](#)

档案2 [View profile](#)

4 Create a SIP profile

Create a SIP profile

- 1 Choose name 2 Set up subscription 3 Authentication

Creating a SIP profile is as easy as three steps. Simply choose a name for your profile, purchase a channel subscription, and get your authentication details.

Choose a profile name

aaa

For example, "New York office". You can edit this name later.

[Next](#) [Cancel](#)

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



aaa

Profile settings

Authentication details

Reports

[← Back to SIP Profile list](#)

Profile settings

Profile name: aaa

Calling channels: [Buy a channel subscription to activate this profile](#)

Outgoing calls: [Set up outgoing calls](#)

To make outgoing calls from this SIP Profile you need to add Sk...
You can also set up Auto-recharge so you never run out of credi...
call. Outbound calls to landlines and mobiles in the US* are ch...
cents/min. For all other destinations see [Skype's standard per t...
rates.](#)

[Add credit](#) [Auto-recharge settings](#)

 € 0.30 [Add credit](#)

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



aaa

Profile settings

Authentication details

Reports

[← Back to SIP Profile list](#)

Authentication details

Please choose the method of authentication needed for your PBX.

✔ **Registration**
(Username/password)

or, IP Authentication ?

| | |
|-----------------------|--|
| SIP User | Skype user name |
| Password | Skype password Generate a new password |
| Skype Connect address | sip.skype.com |
| UDP Port | 5060 |

⚠ SIP user is not yet registered at sip.skype.com

5 Settings on IPPBX

5.1 Build one sip trunk with Skype for sip account

Provider Type: Custom Trunk

Host: sip.skybe.com

User name: the user name you defined in Authentication detail

Password: the password you defined in Authentication detail

New VoIP Trunk
X

Description: Skype

Protocol: SIP ▼

Host: sip.skype.com : 5060

Maximum Channels*: 0

Prefix: _____

Caller ID: _____

Without Authentication

Username: Skype user name

Authuser: Skype password

Password:

Advanced Options

Save
Cancel

5.2 Set one outbound rule

New DialRule X

Rule Name: skype

PIN Set:

Place this call through:

| | | |
|---|----------------------|--|
| <div style="border: 1px solid #ccc; height: 100px; width: 100%;"></div> <p style="text-align: center; margin-top: 5px;">Available Trunks</p> | >>> ↓ ↑ <<< | <div style="border: 1px solid #ccc; padding: 5px;">Skype(SIP)</div> <p style="text-align: center; margin-top: 5px;">Selected Trunks</p> |
|---|----------------------|--|

Custom Pattern: 0.

Z Any digit from 1 to 9
N Any digit from 2 to 9
X Any digit from 0 to 9
. Any number of additional digits

Delete 1 digits prefix from the front and auto-add digit _____ before dialing

Edit X

DialPlan Name: DialPlan1

| | |
|---|--|
| Include External Calling Rules <input checked="" type="checkbox"/> Skype | Include Internal Calling Rules <input checked="" type="checkbox"/> Extensions <input checked="" type="checkbox"/> Spy <input checked="" type="checkbox"/> Conference <input checked="" type="checkbox"/> Ring Groups <input checked="" type="checkbox"/> IVR <input checked="" type="checkbox"/> Call Queues <input checked="" type="checkbox"/> Paging and Intercom <input checked="" type="checkbox"/> Directory <input checked="" type="checkbox"/> DISA |
|---|--|

5.3 Make an outbound call

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

For example, dialing number 00(outbound prefix number)+ 001(International Code)+ 886(Country code) + 2(city Area code without 0)+ 22199518(local phone number) will enable you to contact Taiwan Planet Company

5.4 Set inbound rule

New Number DIDX

DID Number: Skype number

Destination: Goto IVR ▼ working time ▼

Save Cancel