

User's Manual



Internet Telephony PBX system

▶ IPX-1980



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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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To assure continued compliance (example-use only shielded interface cables when connecting to computer or peripheral devices). Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with Part 15 of the FCC Rules. Operation is subject to the Following two conditions: (1) This device may not cause harmful interference, and (2) this Device must accept any interference received, including interference that may cause undesired operation.

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 following 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-1980

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Chapter 1 Introduction

PLANET IPX-1980 IP PBX telephony system is SIP based and optimized for the small and medium business in daily communications. The IPX-1980 is able to accept 100 user registrations, and easy to manage a fully voice over IP system with the convenience and cost advantages.



Based on state-of-the-art embedded technology, the IPX-1980 provides a solid, uniform platform for voice as well as data network communications. It offers a seamlessly integrated solution for the up-to-date telecommunication needs. Being more flexible, the IPX-1980 integrates up to 8 FXO ports to become a feature-rich PBX system that supports smooth communications between existing PSTN calls, analog phones, IP phones and SIP-based endpoints.

The IPX-1980 integrates NAT functions to make it perfect for small business usage. Besides traditional PBX system functions, it provides many advanced functions including voice mail to email, web management and etc. Designed to run on a variety of VoIP applications, the IP PBX provide IP-based communications, voice conferencing, support paging/intercom, call monitoring and BLF (Busy Lamp Field) functions. It also supports call detailed record (CDR), centralized Auto-Attendant (AA), and Interactive Voice Responses (IVR). The IPX-1980 utilizes standard PSTN lines via the interfaces of gateway to support seamless communications among local calls, SIP-based endpoints including low cost long distance call service, telephone number portability and one network for both voice and data.

With the IPX-1980, standard SIP phones can be easily integrated in your office. Users may build up the VoIP network in minutes by applying the IPX-1980 with PLANET IP Phone series, ATA (Analog Telephone Adapter) series and Gateway series. Allowing distributed IP technology to meet traditional voice services with proactive managed interface, the IP PBX IPX-1980 supports daily business processes more efficient and productive.

1.1 Features

■ PBX Features

- 30 Concurrent calls / Up to 100 registers
- BLF (Busy Lamp Field)
- DID (Direct Inward Dialing Number)
- Conference Room
- Automated Attendant (AA)
- Interactive Voice Responses (IVR)
- Built-in voice mail server
- DISA (Direct Inward System Access)
- User Management via Web Browsers
- Display 100 Registered User's Status: Unregistered / Registered / On-Call
- Multiple Service Providers Lines / SIP Accounts (10)
- Simultaneous Trunk Links: 10 concurrent trunk calls
- Analog/GSM, VoIP Trunk, Peer Trunk Management
- Two-stage / One-stage call to Trunk by Trunk Group Configuration
- Built in 8 port FXO
- By adding external FXO analog gateway to use Terminal trunk Line
- By adding external GSM VoIP gateway to use GSM trunk line
- Built-in SIP Proxy Server Following RFC-3261

■ Call Features

- Call Paging and Intercom
- Call Forward Immediate
- Call Forward on Busy
- Call Forward on No Answer
- Call Pickup / Call Park
- Call / Pickup Group
- Caller ID / T.38 (Pass Through)
- Music on Hold / Music on Transfer
- Call Transfer / Call Hold / Call Waiting
- Call Queue
- 3-Way conference with feature phones
- Call Monitoring

■ Other Features

- Supports Skype for SIP
- DDNS Client (Supports Planet DDNS / DynDNS.org / Zoneedit.com)
- Trouble Shooting (Ping, Traceroute)
- Auto Provision
- VPN Client (Supports N2N / L2TP)
- Black List / Phone Book

1.2 Package Contents

Thank you for purchasing PLANET Internet Telephony PBX system, IPX-1980. This Quick Installation Guide will introduce how to finish the basic setting to connect 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:

- IPX-1980 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 above items are damaged or missing, please contact your dealer immediately.

1.3 Physical Specification

Dimensions

240 x 368 x 82 mm (W x D x H)

Weight

1800g

Front Panel



Rear Panel




LED definitions

Front Panel LED	State	Descriptions
Front Panel LED	State	Descriptions
PWR	On Off	PBX Power ON PBX Power OFF
SYS	On Flashing Off	Enabling system System is working System is off
LAN	Flashing Off	LAN is connected successfully Ethernet not connection
WAN	Flashing Off	PBX network connection established Waiting for network connection

1	12V DC	12V DC Power input outlet
2	Reset	The reset button, when pressed, resets the IP PBX without the need to unplug the power cord.
3	WAN	The WAN 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, ADSL modem, through a CAT.5 twisted pair Ethernet cable.
4	LAN	The LAN port allows your PC or Switch/Hub to be connected to the IP PBX through a CAT.5 twisted pair Ethernet cable.
5	FXO Port	Connect 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 for 6 Secs	System reboot.
	Press Over 6 Secs	Reset to Factory Default

 Note	Please be remind, reset to factory default, Upload music setting (on hold music) and backup file will not remove.
--	---

1.4 Specification

Product	Internet Telephony PBX System
Model	IPX-1980
Hardware	
LAN	1 x 10/100Mbps RJ-45 port
WAN	1 x 10/100Mbps RJ-45 port
FXO	8 x RJ-11 connection
Protocols and Standard	
Protocols and Standards	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
Management	HTTP Web Browser
Internet Connection Type	Fixed IP, PPPoE, DHCP client
Call control	SIP 2.0 (RFC3261) , SDP (RFC 2327), Symmetric RTP
Registration	Max. 100 nodes / SIP IP phones / ATA / Voice Gateways / Video phone
Calls	Max. 30 concurrent calls
Voice Codec Support	G.711-Ulaw, G.711-Alaw, G.726, G.729GSM, SPEEX
Video Codec	H.261, H.263, H.263+, H.264
Voice Processing	DTMF detection and generation In-Band and RFC 2833, SIP INFO
PBX features	30 Concurrent calls / Up to 100 registers BLF (Busy Lamp Field) DDNS Client (Supports Planet DDNS / Dyn dns.org / Zoneedit.com / No-ip.com) VPN Client (Supports N2N /L2TP) DID (Direct Inward Dialing Number) Conference Room Automated Attendant (AA) Interactive Voice Responses (IVR) Built-in voice mail server DISA (Direct Inward System Access) User Management via Web Browsers Display 100 Registered User's Status: Unregistered / Registered / On-Call Multiple Service Providers Lines / SIP Accounts (10) Simultaneous Trunk Links: 10 concurrent trunk calls Analog/GSM, VoIP Trunk, Peer Trunk Management Two-stage / One-stage call to Trunk by Trunk Group Configuration Built in 8 FXO PSTN trunk By adding external FXO analog gateway to use Terminal trunk Line Built-in SIP Proxy Server Following RFC-3261

	Fax Support using T.30 or T.38 Pass-Through ** Multiple Language: Chinese, English, Portugal
Call features	Call Paging and Intercom Caller ID Call Group Call Hold Call Waiting Call Transfer Call Forward (Always, Busy, No Answer) Call Pickup Call Park Call Resume Call Queue Music on Hold 3-Way conference with feature phones VIP-256 series, VIP-255PT, VIP-361PE, VIP-362WT, and the ATA (Analog Telephone Adapter) VIP-156, VIP-157
Internet Sharing	
Protocol	TCP/IP, UDP / RTP / RTCP, HTTP, ARP, NAT, DHCP, PPPoE, DNS
Other Function	DDNS, Ping test, Auto Provision
Connection Type	Static IP, PPPoE, DHCP
Management	HTTP Web Browser
LED Indications	System: 1, PWR, 1, SYS, 8 FXO Ports WAN:1, LNK/Off LAN:1, LNK/Off
Environment	
Dimension (W x D x H)	240 x 368 x 82 mm
Operating Temperature	-10 ~ 45 Degree C, 10 ~ 80% humidity
Power Requirement	12V DC
EMC/EMI	CE, FCC Class B, RoHS
Remark: ** T.30/ T.38 support depends on fax machine, SIP provider and network / transport resilience.	

Chapter 2 Installation Procedure

Basic System Configuration

2.1 Web Login

- Step 1.** Connect a computer to a LAN port on the IPX-1980. Your PC (DHCP client mode) will obtain an IP address automatically. (It is usually in the 192.168.0.x range.)
- 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-1980: 192.168.0.1 into the URL address box.
- Step 4.** Enter the default user name **admin** and the default password **admin** then click Login to enter Web-based user interface.

(Default IP)

Default WAN IP: **172.16.0.1**

Default LAN IP: **192.168.0.1**

Default Name: **admin**

Default Password: **admin**

PLANET
Networking & Communication

Username:

Password:

Language:

Login

Please login...

Figure 2-1. Login page of the IPX-1980

2.2 Configuring the WAN

Step 1. Go to System → System Connection to find **Network & Country**.

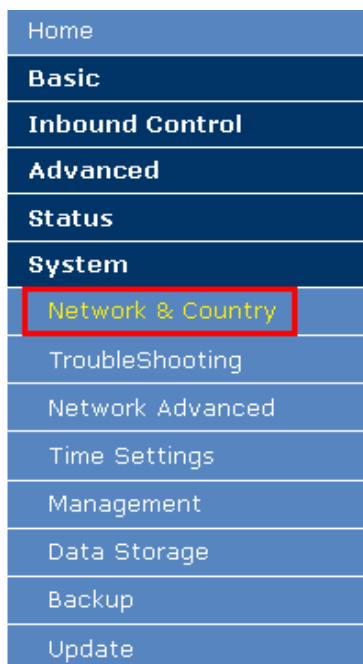


Figure 2-2. Network & Country button

Network & Country

WAN Port Setup	
IP Assign:	Static
Hostname:	IPPBX
IP Address:	192.168.1.100
Subnet Mask:	255.255.255.0
Gateway:	192.168.1.1
Primary DNS:	192.168.1.1
Alternate DNS:	
HTTP Port:	80
Remote Administration:	<input checked="" type="checkbox"/>

LAN Port Setup	
IP Address:	192.168.0.1
Subnet Mask:	255.255.255.0

Tone Zone Setting	
Country:	TW - Taiwan

Figure 2-3. Network setting page

Step 2. Edit your WAN information.

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

WAN Port Setup	
IP Assign:	Static
Hostname:	Static
IP Address:	DHCP 1.100
Subnet Mask:	PPPoE 255.255.255.0
Gateway:	192.168.1.1
Primary DNS:	192.168.1.1
Alternate DNS:	
HTTP Port:	80
Remote Administration:	<input checked="" type="checkbox"/>

LAN Port Setup	
IP Address:	192.168.0.1
Subnet Mask:	255.255.255.0

Figure 2-4. WAN connection type selection item

2.3 Remote Management

The function can enable users to manage the Internet Telephony PBX system at remote sites and to allow technical person to assist you in solving network problem.

Step 1. Go to **Network & Country** → **HTTP port and Remote Administration**.

Network & Country

The screenshot shows a configuration interface for 'WAN Port Setup' and 'LAN Port Setup'. In the 'WAN Port Setup' section, the 'IP Assign' is set to 'Static'. The 'Hostname' is 'IPPBX'. The 'IP Address' is '192.168.1.100', 'Subnet Mask' is '255.255.255.0', 'Gateway' is '192.168.1.1', and 'Primary DNS' is '192.168.1.1'. The 'HTTP Port' is set to '80' and 'Remote Administration' is checked. In the 'LAN Port Setup' section, the 'IP Address' is '192.168.0.1' and the 'Subnet Mask' is '255.255.255.0'.

Figure 2-5. HTTP port and Remote Administration

Step 2. Mark **Remote Administration** to active it. And then enter the control port you want to use. The default value is **80**.

Step 3. Click **Apply** to save the configuration. And you can type the **http://WAN IP address:** to access the IPX-1980 from the remote side.

Step 4. To verify the IP address of your computer and the Internet Telephony PBX system. Click **Start** form Windows → **Run**. Type the **cmd** to open the command window, then type **ipconfig** for getting default gateway address. In the below case, the default gateway is 192.168.0.1 and the user's PC is 192.168.0.100.

```
Ethernet adapter Local Area Connection:

Connection-specific DNS Suffix . : smb.com
IP Address. . . . . : 192.168.0.100
Subnet Mask . . . . . : 255.255.255.0
IP Address. . . . . : fe80::222:19ff:fe06:b981%9
Default Gateway . . . . . : 192.168.0.1
```

Figure 2-6. Check the IP address for the PC and IPX-1980



Every Time after save the change please press the “Activate Changes” to make modification effect.

Chapter 3 Basic Configuration

3.1 Preparation Before Operation

What kind of IP Phone can be used with IPX-1980 IP PBX?

- Our IPX-1980 is base on SIP 2.0 (RFC 3261) any IP phone model base on same protocol can be work with IPX-1980.

3.2 Before Making a Call

3.2.1 System Information

Default WAN IP: **172.16.0.1**

Default LAN IP: **192.168.0.1**

Default Name: **admin**

Default Password: **admin**



Username:

Password:

Language:

Login

Please login...

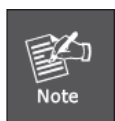


1. To login IPX-1980 must segment PC to same domain as IPX-1980 WAN or LAN IP address.
2. For security reason, please modify the username and password after you login. You can modify in this page: "System"---"Management"
3. **Every Time after save the change please press the "Activate Changes" to make modification effect.**

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

The screenshot displays the PLANET web management interface for an IPX-1980 device. The interface includes a navigation menu on the left with options like Home, Basic, Inbound Control, Advanced, Status, and System. The main content area is titled 'System Info' and is divided into several sections: Network (showing WAN IP 210.66.155.90 and LAN IP 192.168.0.98), Storage (Flash: Total 1015.0M, Used 33.2M), Channels (8 FXO ports), and Device Info (Model No.: IPX-1980, System Version: V4.0). At the bottom, there are buttons for 'Refresh', 'Reboot', and 'Factory Defaults'. A 'Run Time' indicator shows 2 days 2:39. The top right corner has 'Activate Changes' and 'Logout' buttons. A tooltip提示 'Move the mouse over to a field to see tooltips' is visible on the right side.

- **Network** WAN/ LAN Port IP will be displayed
- **Storage** Total storage and used storage will be displayed
- **Channels** Channel information will be based on the product model
- **Device Info** Product Model and System Version will be displayed



1. If FXO without connection, the color will be Orange.
2. If FXO do connected, the color will be Green, also the front panel LED will be lighting.

System Info			
Network			
WAN IP			210.66.155.90
LAN IP			192.168.0.98
Storage			
Flash		Total: 1015.0M	Used: 35.4M
Channels			
1	2	3	4
5	6	7	8
FXO	FXO	FXO	FXO
FXO	FXO	FXO	FXO

Common Button

Besides of the device info in the home page, the following common buttons are displayed as well:

- [Log out](#) Log out GUI
- [Reboot](#) Reboot the IP PBX system
- [Factory Defaults](#) Restore all settings to factory default
- [Activate Changes](#) Activate the changes for your current configuration

System Menu

System Menu includes the following sub menu:

- [Home Page](#) Display device info
- [Basic](#) Basic configuration on extension, trunks, etc
- [Inbound Control](#) Configure Inbound Route, IVR and Black List, etc
- [Advanced](#) Configure extension's default info, conference, etc.
- [Status](#) Check record list, call logs, register status, etc here.
- [System](#) Configure network, time, etc; manage call logs, back up files, etc

3.2.2 Basic Configuration

Configure Extensions

Planet IP PBX support SIP/IAX2 and analog extension, configure extension from this page: **【Basic】** ---- **【Extensions】**

The screenshot displays the PLANET web interface for extension configuration. The sidebar on the left contains navigation links: Home, Basic, Extensions, Trunks, Outbound Routes, Inbound Control, Advanced, Status, and System. The main content area is titled 'Extension Configuration' and includes an 'Extension Number' search field with 'Search' and 'Show All' buttons. Below this are buttons for 'Create New User', 'Batch Add Users', and 'Delete Selected Users'. A table titled 'Extensions' lists 10 entries with columns for S.No, Name, Extension, Port, Protocol, Dial Plan, Outbound CID, and Options. Each row has an 'Edit' link. A tooltip on the right side says 'Move the mouse over to a field to see tooltips'.

S.No	Name	Extension	Port	Protocol	Dial Plan	Outbound CID	Options
<input type="checkbox"/>	1	801		SIP	DialPlan1		Edit
<input type="checkbox"/>	2	802		SIP	DialPlan1		Edit
<input type="checkbox"/>	3	803		SIP	DialPlan1		Edit
<input type="checkbox"/>	4	804		SIP	DialPlan1		Edit
<input type="checkbox"/>	5	805		SIP	DialPlan1		Edit
<input type="checkbox"/>	6	806		SIP	DialPlan1		Edit
<input type="checkbox"/>	7	807		SIP	DialPlan1		Edit
<input type="checkbox"/>	8	808		SIP	DialPlan1		Edit
<input type="checkbox"/>	9	809		SIP	DialPlan1		Edit
<input type="checkbox"/>	10	810		SIP	DialPlan1		Edit

Extension Settings

Item	Explanation
Search	Search extension precisely or fuzzily
Show all	Show all extensions
Extension	Be connected to the phone eg: "888"
Name	Extension name (English letter is supported only) eg: "Tom"
Password	Password of SIP/IAX2 extension eg: "12u3b6"
Caller ID	Caller's ID eg: "801"
Outbound CID	Overrides the caller id when dialing out with a trunk.
VM Password	Voicemail Password for this user, eg: "1234".
E-mail	The e-mail address for this user, eg. "Tom@gmail.com"
Analog Phone	If this user is attached to an analog port on the system, please choose the port number here.
Dial Plan	Please choose the Dial Plan for this user, Dial Plan is defined under the "Outbound Routes".
Voicemail	This user will have a voicemail account after choosing this option.
Can reinvoke	Set up calls directly between caller and receiver, after being connected by IP PBX system. This method is known to cause problems with certain hardware, such as the common Cisco ATA 186.
SIP	Check this option if the User or Phone is using SIP or is a SIP device.
IAX2	Check this option if the User or Phone is using IAX2 or is an IAX2 device.
T.38 Fax	Enables T.38 fax (UDPTL) pass through on SIP to SIP calls

Agent	Check this option if this User or Phone is a Call Agent.
NAT	Check this option if the User or Phone is located behind a NAT (Network Address Translation) enabled gateway.
Pickup Group	Select your pickup group.
Delete VMail	Voicemail will not be checkable by phone if you choose this option. Messages will be sent by email only. Note: You must configure SMTP server for this functionality.
DTMF Mode	The Dual-Tone Multi-Frequency mode to be used is specified here and can be changed if necessary. The default is rfc2833.
Video Call	Enable/Disable Video call for this extension
Permit IP	IP address and network restriction. eg: "192.168.1.77" or "192.168.10.0/255.255.255.0"
Auto Provision	Please select the phone manufacture and input MAC address of the IP Phone. For more details, Please check in Part 3.10
Codecs Configure	The allowed and disallowed codecs can be selected by clicking this link. Default codecs are alaw, ulaw and G.729.



1. There are few default extensions which number started with "8", you can add or delete extension by your requirement
2. As our professional suggestion, extensions don't exceed 100 accounts. If extensions were over 100, it will cause the system crashed or other problems.
3. For security reason the default password is random character or number ex: BB%ChH64rl, and every time when you reset to default system will random a new password again.

3.2.3 Time Based Rules

You can set working time rule and after-working time rule, and deal with your inbound call based on this time rule. Please set from this page: **【Time Based Rule】** --- **【New Time Rule】** :

New Time Rule
X

Rule Name: (Ex: July4)

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 did not match:

New Time Rule:

Item	Explanation
Rule Name	Define the time rule name.
Time & Date Conditions	Set time segment of Month/Date/Week.
Destination	How to deal with the inbound call in different time segment eg: Inbound call will be forward to IVR 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】**



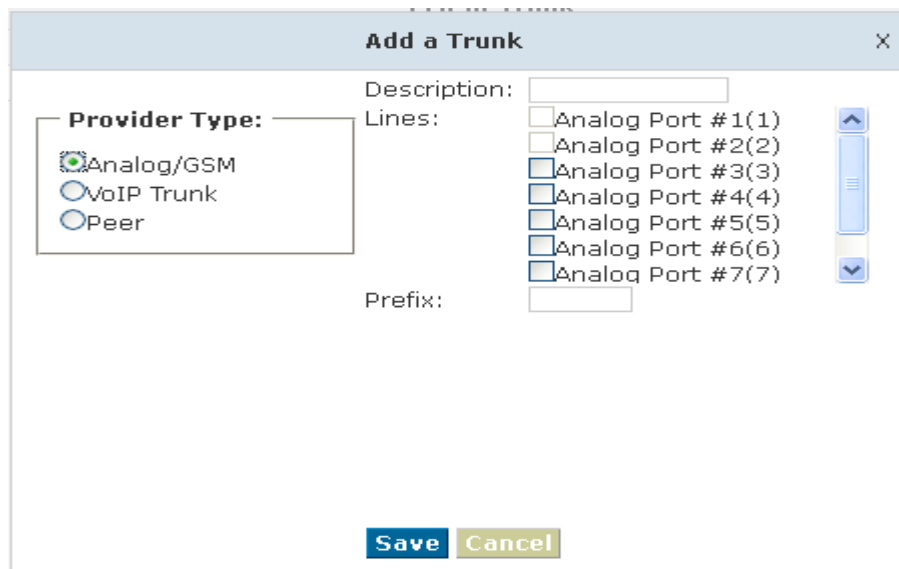
S.No	Trunk	Type	Options
1	Ports 1,2	Analog/GSM	Options ▾

Planet IP PBX supports 3 kinds of trunks: Analog/GSM line, Custom VoIP, Peer.

How to add each trunk:

1) Analog/ GSM Line

Click **【Add a Dial Rule】** -> **【Analog/GSM】**



Add a Trunk [X]

Provider Type:

- Analog/GSM
- VoIP Trunk
- Peer

Description:

Lines:

- Analog Port #1(1)
- Analog Port #2(2)
- Analog Port #3(3)
- Analog Port #4(4)
- Analog Port #5(5)
- Analog Port #6(6)
- Analog Port #7(7)

Prefix:

Save **Cancel**

Item	Explanation
Description	Define description for the trunk.
Lines	Individual lines of the PBX eg: Analog Port #3: The third analog port of the PBX.

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

2) Custom VoIP

Custom VoIP allows you to create a VoIP trunk, please configure on this page:

【Add a Trunk】 -> 【VoIP Trunk】

Item	Explanation
Description	Description for VoIP Trunk, digit or letter is allowed.
Protocol	Choose protocol for this trunk, SIP or IAX2
Dial Plan	Choose a dial plan for this trunk, define it in the submenu named 【Outbound Routes】 .
Register	Check for opening register service; otherwise register service is closed
Host	Host Address provided by VoIP Provider.
Outbound proxy	Outbound proxy is provided by VoIP Provider.
Proxy Port	Proxy Port is provided by VoIP Provider.
Without	If you don't use Authentication when connecting server, Please

Authentication	check this option.
Username	Username provided by VoIP Provider.
Password	Password provided by VoIP Provider.

3) Peer

PLANET IP PBX will be taken as a Client when you use "Peer", it's used for outbound call by connecting to another IPX-1980 IP PBX.

Add a Trunk
✕

Provider Type:

Analog/GSM

VoIP Trunk

Peer

Peer Name:

Protocol:

Dial Plan:

Host:

NAT:

Prefix:

Without Authentication

Username:

Password:

Save
Cancel

Item	Explanation
Peer Name	Define the Peer Name, digit or letter is allowed.
Protocol	Choose protocol for this trunk, SIP or IAX2
Dial Plan	Choose a dial plan for this trunk, define it in the submenu named 【Outbound Routes】 .
Host	IP Address of the other IPX-1980 IP PBX
NAT	Check this option, extension user will be configured after NAT (Network Address Translation).
Without Authentication	If you don't use Authentication when connecting server, Please check this option.
Username	Username provided by the other IPX-1980 IP PBX.
Password	Password provided by the other IPX-1980 IP PBX.

Once A trunk is added, this trunk will be displayed in the "List of Trunk". You can define the codecs, configure advanced settings or delete this trunk from the drop downs of "Option"

3.3.2 Outbound Routes

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

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



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

The 'Add a Dial Rule' dialog box contains the following fields and options:

- Rule Name:
- PIN Set:
- Place this call through: (dropdown)
- Failover:
- Dialing Rules: If the number begins with and followed by (more than) digits
(Define a custom pattern)
- Delete digits prefix from the front and auto-add digit before dialing

Buttons: **Save** (blue), **Cancel** (yellow)

Item	Explanation
Rule Name	Set a name for this dial rule
PIN Set	Set PIN which you need input when you dial out by this rule.
Record in CDR	If you selected it, CDR will show which pin the call is outbound through
Place this call	Choose a trunk for this rule

through	
Failover	Choose a failover trunk for using when the above chosen trunk is not available.
Dialing Rules	Define the number match pattern for dialing.
Define a custom pattern	N digit from 2 to 9 Z digit from 1 to 9 X digit from 0 to 9 . One digit or multiple digits
Delete[]digits prefix	If deleted one digit prefix, when dial 12345, digit 2345 will be sent.
Auto-add digit[]	If added digit"1", when dial 12345, digit 123451 will be sent.

3.4 Inbound Call

3.4.1 Inbound Routes

When a call 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.

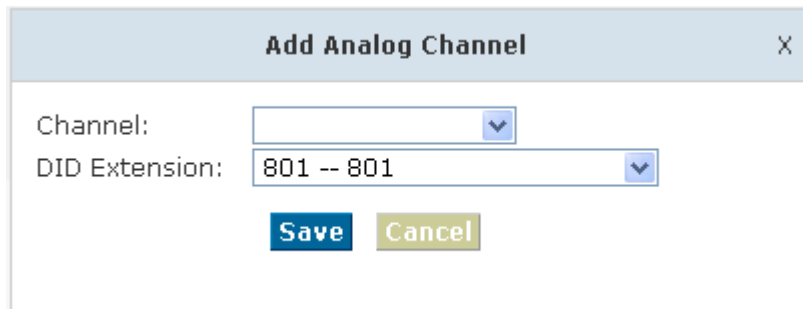
Please configure on this page: **【Inbound Routes】**

General

When a call from a trunk (Analog/ VoIP), it could be forwarded to an extension, call queue, conference or IVR. You can choose based on your requirement.

Analog Channel DID

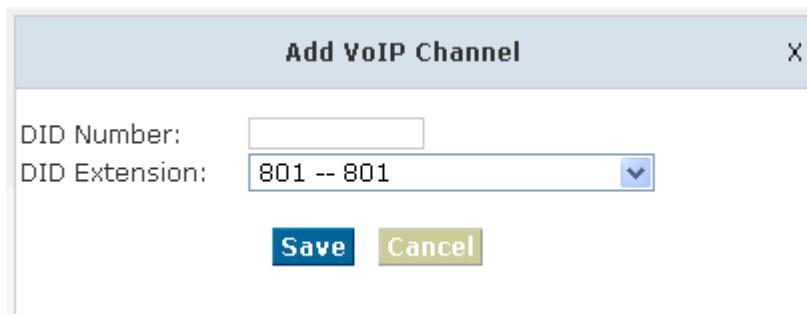
If you want to direct the inbound call from a trunk (Analog) to a specified extension, call queue, conference or IVR, please configure on this page: **【Add Analog Channel】**



- Channel Choose Analog Port of trunk
- DID Extension Select Extension, call queue, conference or IVR for DID.

VoIP Channel DID

If you want to direct the inbound call from a VoIP trunk to a specified extension, call queue, conference or IVR, please configure on this page: **【Add VoIP Channel】**

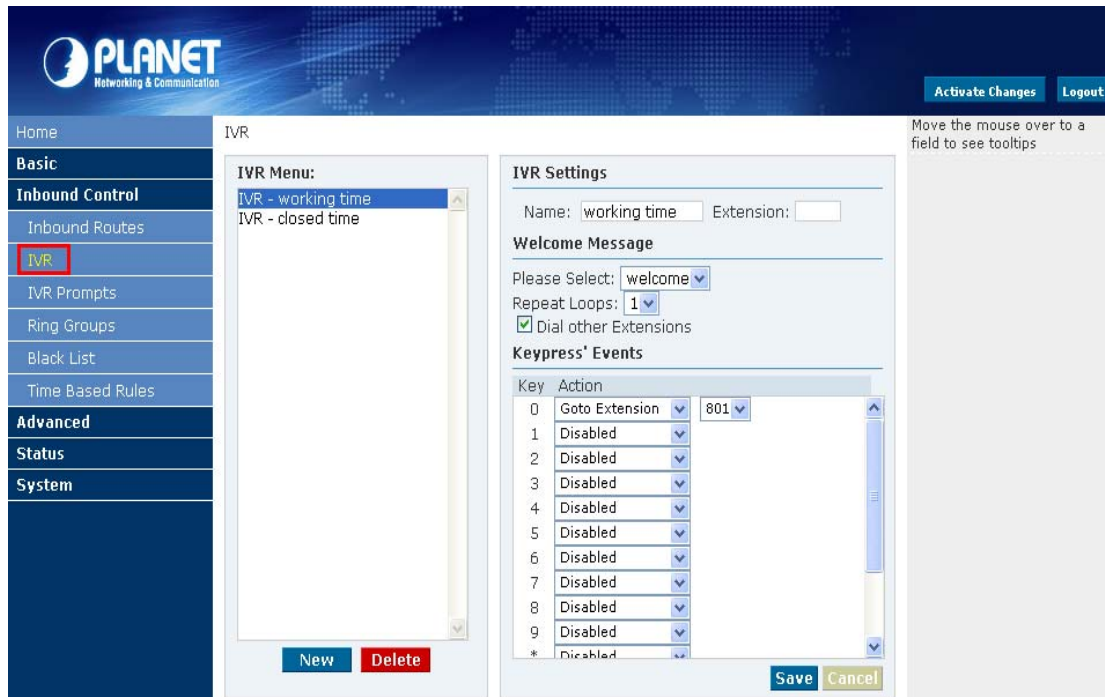


- DID Number DID number calling into VoIP (This number is configured in the advance option of VoIP trunk)
- DID Extension Choose a specified extension, call queue, conference or IVR to be directed to call.

3.4.2 IVR

IVR will improve office efficiency based on your requirement.


Please configure on this page **【IVR】**



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 extension besides of the ones listed as 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 Define a name for this ring group
- Strategy Select strategy: "Ring all" or "Ring in order"
- Ring Group Members Select ring group members in available channels, click  to add
- If not answered You can choose forward the call to extension, extension, Voicemail, RingGroup, IVR or Hangup.

3.5 Black List

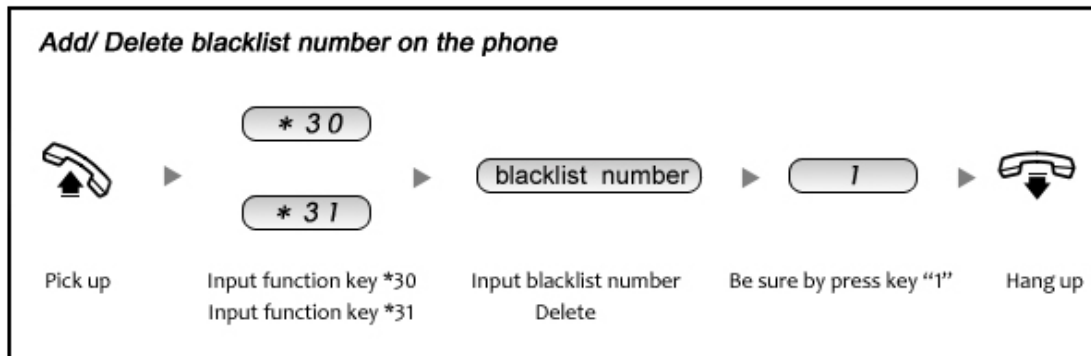
If some numbers need to be blocked, you can use this functionality.

Please configure in **【Black List】** , click **【New Blacklist】** to display this dialog as below:

Input caller's number in the blank, then this caller's number will be blocked when call again. Meanwhile, extension user can add or delete the blacklist number by function key on the

phone.

Please operate as the following diagram:

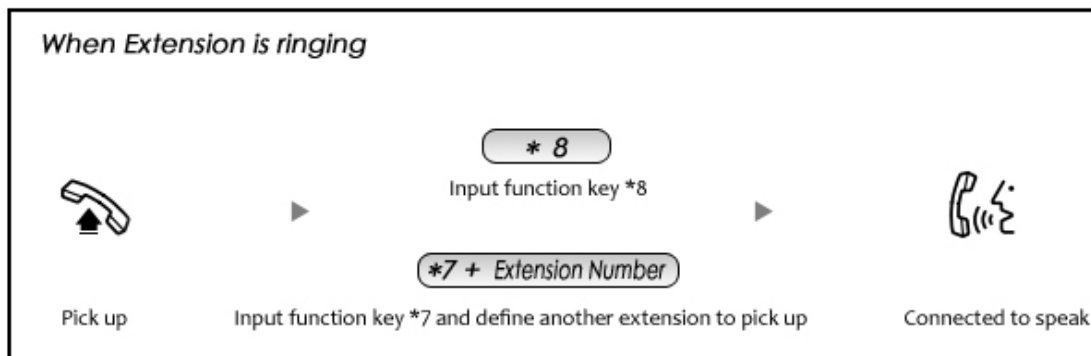


Reference Parameters and Explanation of Blacklist:

Item	Explanation
*30	When the extension user (in the system) input *30 to add a blacklist number, this number will be added to the "Black List"
*31	When the extension user input *31+ blacklist number, this number will be deleted from the "Black List".

3.5.1 Pickup Call

If an extension user is away from his/her desk, other extension users can pickup the call by function key on the phone. Please check the following diagram to learn:



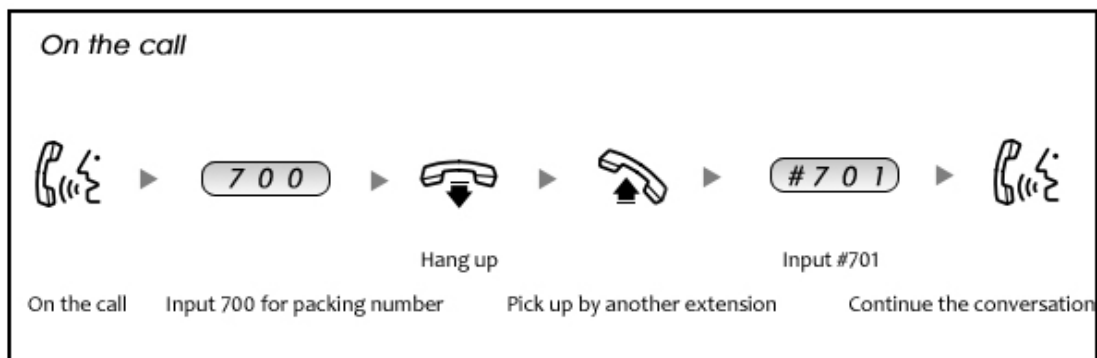
Reference Parameters and Explanation of Pickup Calls

Item	Explanation
*8	Pick up the ringing extension (in the system) at random. This can be defined in 【Feature Codes】
*7	Defined extension number must be inputted after *7. This can be defined in 【Feature Codes】 .

3.6 On The Call

3.6.1 Call Parking

If you picked 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 diagram as below to learn:

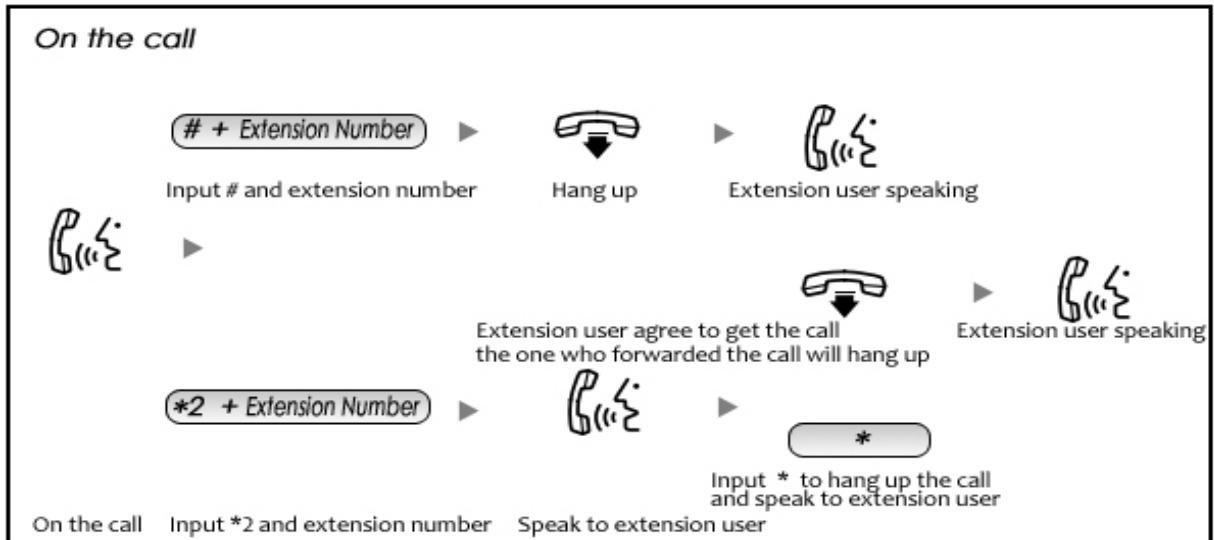


Reference Parameters and Explanation of Call Park:

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

3.6.2 Transfer

If an incoming call asked to speak to your colleague, you can transfer the call directly to your colleague or transfer the call after agreed by your colleague. Please check the diagram as below to learn:



Reference Parameters and Explanation of Transfer:

Item	Explanation
Blind Transfer	Default is #, it can be defined in 【Feature Codes】
Attended Transfer	Default is *2, it can be defined in 【Feature Codes】
Disconnect Call	Default is *, it can be used after you use function key " *2 ". it can be defined in 【Feature Codes】
Timeout for answer on attended transfer	Default is 15 seconds, it can be defined in 【Feature Codes】

3.6.3 Conference

If you wanted 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 into conference room. This model supports 3 conference rooms. Please configure on this page **【Conference】** :

The screenshot shows the PLANET Network & Communication web interface. The sidebar menu on the left includes options like Home, Basic, Inbound Control, Advanced, Options, VoiceMail, **Conference** (highlighted), Call Queue, Music Settings, DISA, Follow Me, Paging and Intercom, Monitor, Phone Book, Pin Set, Feature Codes, Auto Provision, Status, and System. The main content area is titled 'Conference(Default)' and has three tabs: 'Conference(Default)', 'Conference 2', and 'Conference 3'. The 'Conference(Default)' tab is active, showing the following configuration fields:

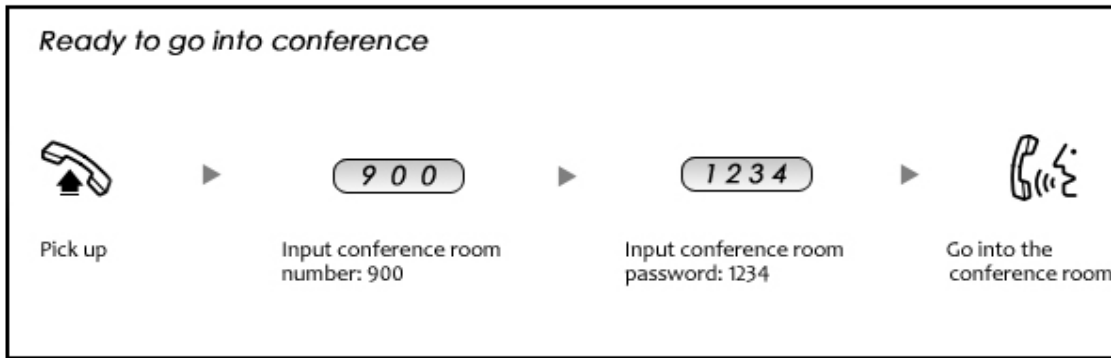
- Conference Number:** Extension: 900
- Conference Password:** PIN Code: 1234, Admin PIN Code: 2345
- Conference Options:**
 - Conference DialPlan: DialPlan1
 - Play hold music for first caller
 - Enable caller menu
 - Announce callers
 - Record conference
 - Quiet Mode
 - Leader Wait

At the bottom of the configuration area are 'Save' and 'Cancel' buttons. On the right side, there is a note: 'Record Conference: Record this conference by WAV format.'

Item	Explanation
Conference Number	The number that users call in order to access the conference room, the default number is "900".
PIN Code	Participants enter the conference room by this code.
Admin PIN Code	Administrator enter the conference room by this code.
Conference DialPlan	Use the dialplan when you invite the other participant.
Play hold music for first caller	Check this option, Asterisk will play Hold Music to the first user in a conference, until another user has joined the same conference.
Enable caller menu	Checking this option allows a user to access the Conference Bridge menu by pressing the * key on their dialpad.
Announce callers	Checking this option announces to all Bridge participants, the joining of any other participants.
Record conference	Recording format is WAV.
Quiet Mode	If this option was checked, all users entering this conference will be marked as quiet, and will be in Listen-Only mode.
Leader Wait	Wait until the conference leader (admin user) arrives before starting the conference.

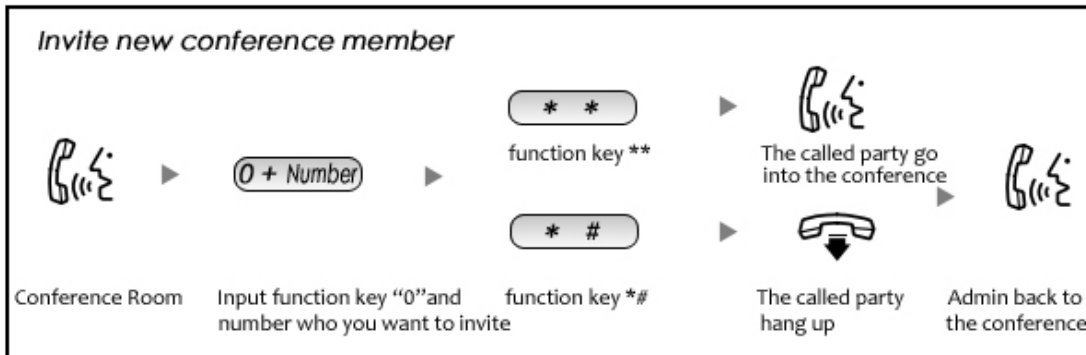
Please check the following diagram to learn:

Go to conference:



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

Add new participant:



3.6.4 Monitor

Monitor the specified extension, also you can monitor in different time.

Please click **【Monitor】** -- **【New Monitor】** to configure:

New Monitor X

Extension:

Monitoring Time

Always Monitor:

Start Time: : End Time: :

Start Day: End Day:

Monitor Settings

Inbound Record: Outbound Record:

Save Cancel

Item	Explanation
Extension	Select an extension which need to be monitored
Monitoring Time	Always monitor or monitor in different time.
Monitor Settings	Set inbound record and outbound record.

3.7 Settings before leaving office

3.7.1 Follow Me

If you don't want to lose any call, you can use this function.

Please click **【Follow Me】** --- **【New Follow Me】**

Item	Explanation	
Extension	Choose an extension	
Ring lasting for(s)	Default is 20 seconds, you can define it by yourself.	
Status	Always	All incoming calls will be forwarded
	Busy	Forward when extension is busy
	No answer	Forward when extension not answer
Set your Follow Me number	Forward to an Internal Extension	Incoming call will be forwarded to internal extension.
	Forward to an External Extension	Incoming call will be forwarded to external number or mobile number.
Set Internal Extension	Set an internal extension to pick up the call.	
Select DialPlan	Select DialPlan when forward the call to	

	external number.
Set External Number	Set external number, like Mobile number.

3.7.2 VoiceMail

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 Rein vite	<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

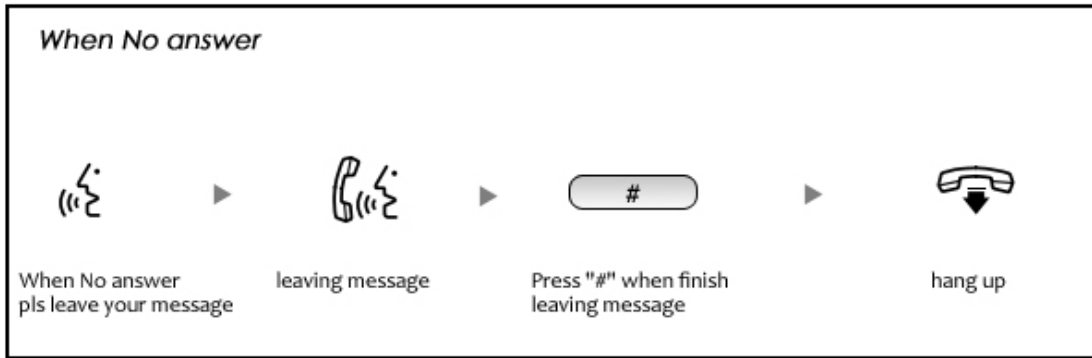
alaw
 ulaw
 G.729
 G.726
 GSM
 Speex

Video Codecs Configure

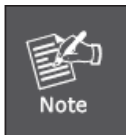
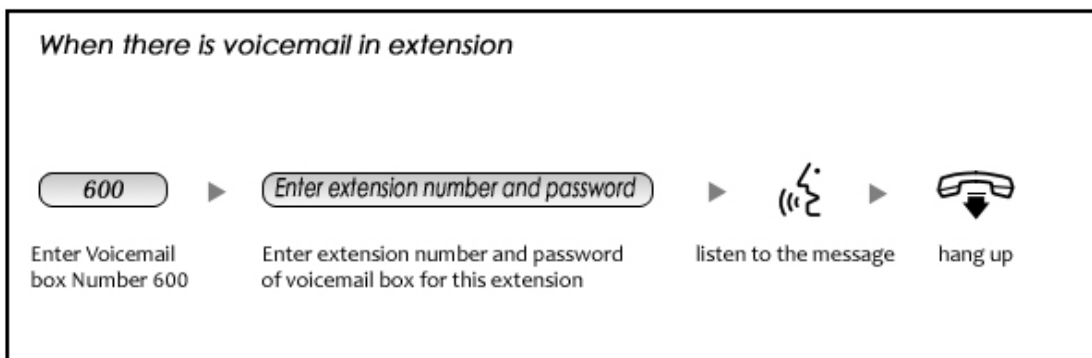
H.261
 H.263
 H.263+
 H.264

【VoiceMail】 must be opened and **【VM Password】** must be configured before using "VoiceMail". If no answer, when default ring time is over, the system will play and ask you to leave your message, press # to end recording. If you configured email, your voice message will be sent to your defined email.

Leave a message:



Listen to the message



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

3.8 Call Queue

3.8.1 Create Agent

Check agent in the **【Extension Settings】**---**【Advanced Options】** , then assign agent and Ring Strategy in **【Call Queue】** , please learn from the following configuration interface:

Call Queue Reference:

Queue Number:

Queue Name:

Ring Strategy: ▼

Agents:

Item	Explanation
Queue Number	This option defines the extension number that may be dialed to reach this Queue.
Queue Name	This option defines a name for this Queue, eg. "Sales"
Ring Strategy	RingAll -- Ring All available Agents until one answered (default). RoundRobin -- Take turns ringing each available Agent. LeastRecent -- Ring the Agent which was called least recently. FewestCalls -- Ring the Agent with the fewest completed calls. Random -- Ring a Random Agent. RRmemory --RoundRobin with Memory, and remember where it left off in the last ring pass.
Agents	All the users who are defined as Agent will be shown here. Selected agent will be a member of the current Queue.

Queue Options:	Announcements:
Agent TimeOut(s): <input type="text" value="15"/> <input type="checkbox"/> Auto Pause Wrap-Up-Time(s): <input type="text" value="10"/> Max Wait Time(s): <input type="text"/> Max Callers: <input type="text" value="8"/> <input type="checkbox"/> Join Empty <input type="checkbox"/> Leave When Empty <input checked="" type="checkbox"/> Auto Fill <input type="checkbox"/> Report Hold Time	Caller Position Announcements Frequency(s): <input type="text" value="30"/> Announce Hold Time: <input type="text" value="no"/> ▼ Periodic Announcements Repeat Frequency(s): <input type="text" value="0"/> Announcements Prompt: <input type="text" value="welcome"/> ▼

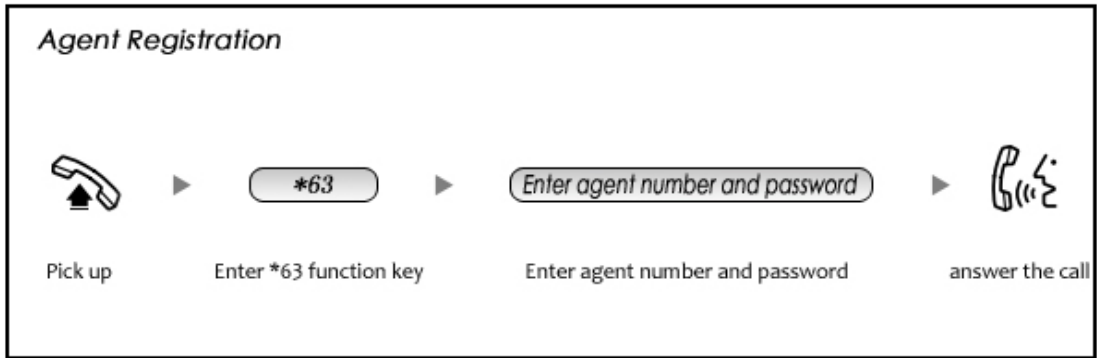
Note: Each agent needs to login to the queue using the login extension defined in Feature Codes.

Item	Explanation
Agent TimeOut (s)	This option defines the time in seconds that an Agent's phone rings before the next Agent is rung, eg. "15"
Auto Pause	Pause an Agent if they fail to answer a call.
Wrap-Up-Time(s)	After a successful call, how many seconds needed to wait before sending another call to a potentially free agent (Default is 0, which means No Delay).
Max Wait Time(s)	The maximum number of seconds a caller can wait in a queue before being pulled out (empty for unlimited).
Max Callers	This option sets the maximum number of callers that may wait in a Queue (Default is 0, Unlimited).
Join Empty	Defining this option allows 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	Defining this option forces all callers to exit the Queue if New Callers are also not able to Enter the Queue. This option should generally be set in concert with the "Join Empty" option.
Auto Fill	Defining this option causes the Queue, when multiple calls are in it at the same time, to push them to Agents simultaneously. Thus, instead of completing one call to an Agent at a time, the Queue will complete as many calls simultaneously to the available Agents.
Report Hold Time	Check this option if you wish to report the caller's hold time to the agent member before they are connected to the caller.
Frequency(s)	How often to announce queue position and estimated hold time (0 to Disable Announcements).
Announce Hold Time	Should we include estimated hold time in position announcements? Either yes, no, or only once; hold time will not be announced if <1 minute.
Repeat Frequency(s)	How often to announce a voice menu to the caller (0 to Disable Announcements).
Announcements Prompt	Select the 'Announcements Prompt' from IVR Prompts

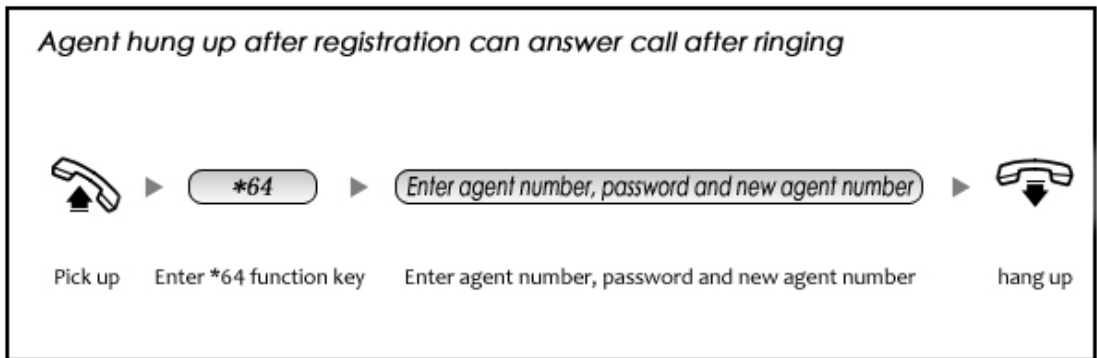
3.8.2 Agent Registration

You need register for using after creating agents.

Agent Registration when hook off



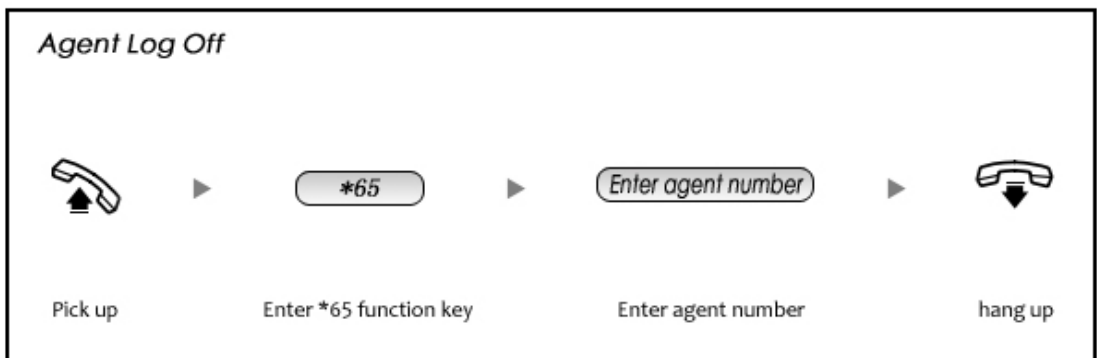
Agent Registration when hook on



3.8.3 Agent Log Off

If agent would leave and log off, none of agent will answer calls then.

Agent Log Off:



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 below:

Options

General	Global Analog Settings	Global SIP Settings
Local Extension Settings		
Local Extensions are <input type="text" value="Varying"/>		
Operator Extension: <input type="text"/>		
Global Ring Time Set(s): <input type="text" value="30"/>		
Enable Transfer: <input checked="" type="checkbox"/>		
Enable Music On Ringback: <input type="checkbox"/>		
<input type="checkbox"/> Allow multiple extensions to be assigned to one analog phone		
<input type="checkbox"/> Allow extensions to be AlphaNumeric (SIP/IAX users)		
Default Settings for a New User		
<input checked="" type="checkbox"/> SIP <input type="checkbox"/> IAX2		
<input type="checkbox"/> Agent <input checked="" type="checkbox"/> NAT		
<input type="text" value="1234"/> VM Password <input type="checkbox"/> Delete VMail		

Item	Explanation
Local Extensions	Set up the digit of local extensions
Operator Extension	Set up Operator Extension.
Global Ring Time Set(s)	Set Ring Time for each extension.
Enable Transfer	Enable transfer feature key.
Enable Music On Ringback	Enable music on ringback.
Allow multiple extensions to be assigned to one analog phone	Allow multiple extensions to be assigned to one analog phone.
Allow extensions to be Alpha Numeric (SIP/IAX users)	If extension is Alpha, outside line can't call in, but extension can call out.

VoiceMail	This user will have a voicemail account after choosing this option.
NAT	Check this option if the User or Phone is located behind a NAT (Network Address Translation) enabled gateway.
SIP	Check this option if the User or Phone is using SIP or is a SIP device.
IAX2	Check this option if the User or Phone is using IAX2 or is an IAX2 device.
Call Waiting	Check this option if the User or Phone should have Call-Waiting capability.
3-Way Calling	Check this option if the User or Phone should have 3-Way Calling capability.
VM Password	Voicemail Password for this user, eg: "1234".
Delete VMail	Voicemail will not be checkable by phone if you chose this option. Messages will be sent by e-mail only. **Note: you must configure SMTP server for this functionality.

4.1.2 General

General	Global Analog Settings	Global SIP Settings
Caller ID Detect		
<p>Caller ID Detection <input checked="" type="checkbox"/></p> <p>Caller ID Signalling <input type="text" value="Bell-US"/></p> <p>Caller ID Start <input type="text" value="Ring"/></p> <p>CID Buffer Length <input type="text" value="2500"/></p> <p>Opermode <input type="text" value="FCC"/></p>		

Item	Explanation
Caller ID Detection	For FXO trunk lines, this option causes PBX to look for Caller ID on incoming calls
Caller ID Signalling	Set the Type of caller ID signaling in use. Bell-US -- Used in the United States(FSK) DTMF -- Used for caller ID under DTMF mode, like: Denmark, Sweden and Netherlands etc; V23 -- used in the UK;

	V23-Japan -- used in Japan;
Caller ID Start	This option allows one to define the start of a Caller ID signal: Ring -- to start when a ring is received. Polarity -- to start when a polarity reversal is started.
CID Buffer Length	Buffer length is 2000, 2500, 3000 only in DTMF mode.
Opermode	Choice Caller ID operator mode.

General	
Relax DTMF	<input checked="" type="checkbox"/>
Echo Cancel	<input checked="" type="checkbox"/>
Echo Training	<input type="text" value="no"/> (yes/no/number)
Busy Detection	<input checked="" type="checkbox"/>
Busy Count	<input type="text" value="5"/>
Call Progress	<input type="checkbox"/>

Item	Explanation
Relax DTMF	If you are having trouble with DTMF detection, you can relax the DTMF detection.
Echo Cancel	Enable/Disable the Echo Cancel function.
Echo Training	Enabling echo training will cause the PBX system to briefly mute the channel, send an impulse, and use the impulse response to pre-train the echo canceller so it can start out with a much closer idea of the actual echo. Value may be "yes", "no", or a number of milliseconds to delay before training (default = 400)
Busy Detection	Used for detecting far end hang up or a busy signal.
Busy Count	If Busy Detection is enabled, it is also possible to specify how many busy tones to wait for before hanging up. The default is 4, but better results can be achieved if set to 6 or even 8. Mind that the higher the number, the more time that will be needed to hang up a channel, but lower the probability that a false detection may occur.
Call Progress	If turned on, call progress attempts to determine answer, busy, and ringing on phone lines.

4.1.3 Global SIP Settings

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

General	
UDP Port to bind to:	<input type="text" value="5060"/>
Start RTP Port:	<input type="text" value="10001"/>
End RTP Port:	<input type="text" value="10200"/>
DTMF Mode:	<input type="text" value="auto"/> ▼
Max Registration/Subscription Time:	<input type="text" value="3600"/>
Min Registration/Subscription Time:	<input type="text" value="60"/>
Default Incoming/Outgoing Registration Time:	<input type="text" value="120"/>

Item	Explanation
UDP Port to bind to	SIP standard port is 5060
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 registration/subscriptions we allow. Default 3600 seconds.
Min Registration/Subscription Time	Minimum duration (in seconds) of registrations/subscriptions. Default 60 seconds.
Default Incoming/Outgoing Registration Time	Default duration (in seconds) of incoming/outgoing registration

NAT Support	
External ip:	<input type="text"/>
External Host:	<input type="text"/>
External Refresh:	<input type="text"/>
Local Network Address:	<input type="text"/>

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:

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

Type of Service

TOS for Signalling packets:
TOS for RTP audio packets:
TOS for RTP video packets:
Enable Relaxed DTMF:
RTP TimeOut:
RTP HoldTimeOut:
Trust Remote Party ID:
Send Remote Party ID:
Generate In-Band Ringing:
Add 'user=phone' to URI:
Send Compact SIP Headers:

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:
Register Attempts:

Codecs

Disallowed Codecs:
Allowed Codecs: [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	Choice the codec system do allow

 <p>Note</p>	<p>1. In extension “Audio Codecs Configure” the priority is higher then “Allowed Codecs” items, “Allowed Codecs” items is default codec setting, if user mark the extension “Audio Codecs Configure” , then system will use it first, if not system will the “Allowed Codecs” to define what codec can be use in extension.</p>
---	---

4.2 Voice Mail

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

VoiceMail

	General	SMTP Settings	Email Settings
VoiceMail Reference			
Extension for checking messages:			600
Max greeting(seconds):			60
Direct to Voicemail:			<input type="checkbox"/>
Dial '0' for Operator:			<input type="checkbox"/>
Voice Message Options			
Message Format:			WAV (16-bit) ▾
Maximum messages:			100 ▾
Max message time(minutes):			5 ▾
Min message time(seconds):			No minimum ▾
Playback Options			
			<input checked="" type="checkbox"/> Say message Caller-ID <input type="checkbox"/> Say message duration <input type="checkbox"/> Play envelope <input type="checkbox"/> Allow users to review
<input type="button" value="Save"/> <input type="button" value="Cancel"/>			

Item	Explanation
Extension for checking messages	The number that users call in order to access their voicemail accounts, the default number is "600".
Max greeting(seconds)	Defining this option to set a maximum time for the greeting message.
Direct to Voicemail	Defining this option to go to voicemail box directly.
Dial "0" for Operator	Callers entering the voicemail application can leave for Operator by dialing "0".
Message Format	Choose the format of the voicemail messages in this selection box.
Maximum Messages	Choose the maximum number of messages in this selection box.

Maximum message time (min)	Choose the maximum duration of a voicemail message. Message recording will be stopped when it's timeout.
Minimum message time (s)	Choose the minimum duration of a voicemail message in this selection box. Message time below this threshold will be deleted automatically.
Say message Caller-ID	Choose this option to play Caller's ID before voicemail message is played.
Say message duration	Choose this option to play the duration of message before the voicemail message is played.
Play envelope	Choose this option to play envelop (including date, time and caller ID).
Allow users to review	Choosing this option, the caller leaving the voicemail can review their recorded message before submitted.

SMTP Settings:

General	SMTP Settings	Email Settings
SMTP Settings:		
SMTP server: <input type="text"/> Port: <input type="text" value="25"/> SSL/TSL: <input type="checkbox"/> <input type="checkbox"/> Enable SMTP Authentication		
<input type="button" value="Save"/> <input type="button" value="Cancel"/>		

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. eg: mail.yourcompany.com
Port	The port number which the SMTP server running is generally port 25. If SSL is encrypted, please use port 465 instead.
SSL/TSL	Enable use 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.
Username	Input username of your email box.
Password	Input password of your email box.

Email Settings

General	SMTP Settings	Email Settings
Template for Voicemail Emails		
<input checked="" type="checkbox"/> Attach recordings to e-mail		
Sender Name	<input type="text" value="IPPBX Server"/>	
From	<input type="text" value="username@mailserver.com"/>	
Subject	<input type="text" value="you've a voicemail from \${VM_CALLERID}"/>	
Message	<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> Dear \${VM_NAME}, you have a new voicemail from \${VM_CALLERID}, the message time is \${VM_DUR}. </div>	
<input type="button" value="Save"/> <input type="button" value="Cancel"/>		
<p>Template Variables:</p> <ul style="list-style-type: none"> <code>\${VM_NAME}</code> : Recipient's firstname and lastname <code>\${VM_DUR}</code> : The duration of the voicemail message <code>\${VM_MAILBOX}</code> : The recipient's extension <code>\${VM_CALLERID}</code> : The caller id of the person who left the message <code>\${VM_MSGNUM}</code> : The message number in your mailbox <code>\${VM_DATE}</code> : The date and time the message was left 		

Item	Explanation
Attach recordings to e-mail	This option defines whether or not voicemails are sent to the Users' e-mail addresses as attachments.
Sender Name	Display the Sender name when you receive a voicemail.
From	Sender's email address
Subject	Subject of the mail
Message	The message pattern

4.3 Music Settings

Management for music on hold, music on ringback, 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 1 <input type="button" value="v"/>
Music On Ringback Reference	
Music:	Music 2 <input type="button" value="v"/>
Music On Call Queue Reference	
Music:	Music 3 <input type="button" value="v"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/> <input type="button" value="Music Reload"/>	

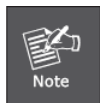
Please define different music file for different music folders.

Music Management:

Music Management

Music Settings	Music Management
Music Management	
Directory:	Music 1 <input type="button" value="v"/> <input type="button" value="Load"/>
Files:	<input type="button" value="v"/> <input type="button" value="Delete"/>
Upload Music File <input type="button" value="Music Reload"/>	
Select Music Directory: Music 1 <input type="button" value="v"/>	
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: <input type="text"/> <input type="button" value="Browse.."/>	
<input type="button" value="Upload"/>	

Item	Explanation
Directory	Load music in the music file.
Files	Display music in the music file, or you can delete it.
Enter The Music File Name	Input music file name which you want to upload.(GSM/ WAV format, If it's WAV, it must be accord with PCM 16 bits, 8000HZ format)
TFTP Server IP address	Please enter your TFTP server IP address.
Select Music Directory	Select directory where the uploaded music file will be saved.



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

4.4 DISA

A trunk call into the PBX, and call to another trunk through outbound route of the PBX. Eg: This trunk can make international call, you are out of the office and want to contact with your customer in foreign country, now you can dial DISA number, after PIN authentication, you are connected to your customer, and you can speak to your customer now.

Click **【DISA】** --- **【New DISA】** to display the dialog as 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	Give this DISA a brief name to help you identify it.
PIN	The user will be prompted for this number

Response Timeout(s)	The maximum amount of time it will wait before hanging up if the user has dialed an incomplete or invalid number. Default is 10 seconds.
Digit Timeout(s)	The maximum amount of time permitted between digits when the user is typing in an extension. Default is 5 seconds.
Extension for this DISA (Optional)	If you want this DISA to be accessible by dialing an extension, you can define an extension number for this DISA.
Select DialPlan	Set the DialPlan that calls will originate from.

4.5 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 **【Paging And Intercom】** --- **【Add Paging Group】** to display the dialog as below:

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) in 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".



1. For Paging/ Intercom function extension(IP phone) must enable **Auto Answer**

4.6 Monitor

Monitor is used for recording the defined extensions.

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

The 'New Monitor' dialog box includes the following fields and controls:

- Extension:** A dropdown menu.
- Monitoring Time:**
 - Always Monitor:**
 - Start Time:** [] : []
 - End Time:** [] : []
 - Start Day:** []
 - End Day:** []
- Monitor Settings:**
 - Inbound Record:**
 - Outbound Record:**

Buttons: **Save** (blue), **Cancel** (yellow)

Item	Explanation
Extension	Define an extension.
Monitoring Time	Set monitoring time
Inbound Record	Check to record inbound calls
Outbound Record	Check to record outbound calls

To check the record information please enter **【Status】** --- **【Record List】** .

4.7 Phone Book

If incoming call was matched with the number in the phone book, the incoming call will display the name of matched number.

Click **【Phone Book】** to display the dialog as below:

Phone Book			
Name:	<input type="text"/>	<input type="button" value="Search"/> <input type="button" value="Show All"/>	
S.No	Name	Number	Options
No Contact defined !!			

- Search Input contact name to search
- Show All Show all contacts

Create Contact
X

Name:

Number:

- Name Add contact's name, Alphabetic or numeric only.
- Number Add contact's number, international phone number is supported.

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

EX: Name: David / Number: 123456789.

When system receive the call 123456789, the extension answer this call it will display "David"



For now our IPX-1980 can only FSK caller ID mode.

4.8 PIN Set

PIN Set will distribute one PIN Code to different extension user, if you selected PIN Set on the Dial rule page in Outbound menu, the extension user who has the PIN code can dial long distance call. Click **【Pin Set】** to show the dialog as below:

Add a PIN Set
X

PIN Set Name:

PIN List:

保存
取消

- PIN Set Name Set the PIN Sets Name
- PIN List Enter a list of one or more PINs. One PIN per line.

4.9 Feature Codes

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

Feature Codes

Feature Codes Management

Call Parking

Extension to Dial for Parking Calls:

What extensions to park calls on:

How many seconds a call can be parked for:

Pickup Call

Pickup Extension:

Pickup Specified Extension:

Transfer

Blind Transfer:

Attended Transfer:

Disconnect Call:

Timeout for answer on attended transfer:

Black List

Blacklist a number:

Remove a number from the blacklist:

Conference

Invite Participant:

Create Conference:

Return to conference with participant:

Return to conference without participant:

Call Queue

Agent Login Extension:

Agent Callback Login Extension:

Agent Logoff Extension:

Pause Queue Member Extension:

Unpause Queue Member Extension:

Save
Cancel

Item	Explanation
Extension to Dial for Parking Calls	Set Call Parking number.
What extensions to park calls on	What extensions to park calls on, eg: (701-720)
How many seconds a call can be parked for	Set the call time by second, if it's time out, system will call the previous extension again.
Pickup Extension	Set Pickup Extension.
Pickup Specified Extension	Set Pickup Specified Extension, default: dial *7+extension to pickup the 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 the 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	Set the answer timeout value.
Blacklist a number	Add a black list number.
Remove a number from the black list	Remove a black list number.
Invite Participant	The administrator can invite another person by pressing 0 when he/she is in the conference. When you press 0, you will get a dial tone to enter the number of part A you also would like to invite. After the call has been established and you talk to B, you can press ** to direct him to the conference, or *# to hang up the current call and return to the conference yourself.
Create Conference	While you speak with another party you can press *0, you and the

	callee are immediately transferred to conference.
Return to conference with participant	The administrator can invite another person by pressing 0 when he/she is in the conference. When you press 0, you will get a dial tone to enter the number of part A you also would like to invite. After the call has been established and you talk to B, you can press ** to direct him to the conference, or *# to hang up the current call and return to the conference yourself
Return to conference without participant	The administrator can invite another person by pressing 0 when he/she is in the conference. When you press 0, you will get a dialtone to enter the number of part A you also would like to invite. After the call has been established and you talk to B, you can press ** to direct him to the conference, or *# to hang up the current call and return to the conference yourself.
Agent Login Extension	Logs the current caller into the queue as a call agent. Once logged in, the agent can take calls with the phone off-hook; each call is preceded by a warning tone. Calls are ended by pressing the "*" key.
Agent Callback Login Extension	Extension to be dialed for the Agents to Login to the Specific Queue. Same as Agent Login Extension, except you do not have to remain on the line.
Agent Logoff Extension	Agent logoff from the queue.
Pause Queue Member Extension	'Pauses' a queue member. So that the member can not receive calls.
Unpause Queue Member Extension	'Unpause' a queue member who is 'paused' previously. So that the member can receive calls again.

4.10 Auto Provision

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

There are two operation methods to fulfill this function, please see details as below:

Enable DHCP service

Click **【System】** -> **【Network Advanced】** , enable DHCP Server in the dialog as below:

DHCP Server Settings

DHCP Server Settings	DDNS Settings	VPN Settings
DHCP Server Settings		
DHCP Service:	<input checked="" type="checkbox"/>	
Start IP:	<input type="text" value="192.168.0.101"/>	
End IP:	<input type="text" value="192.168.0.200"/>	
Subnet Mask:	<input type="text" value="255.255.255.0"/>	
Gateway:	<input type="text" value="192.168.0.1"/>	
Primary DNS:	<input type="text" value="8.8.8.8"/>	
Lease Time:	<input type="text" value="1440minutes"/>	
<input type="button" value="Save"/> <input type="button" value="Cancel"/>		

Method 1:

Click **【 Extension 】** -> **【 Create New User 】**, select the relative IP Phone manufacture, and input relative MAC in the part of Auto Provision, Save and Activate.

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

Method 2:

Click **【 Auto Provision 】** to download auto provision script file model, this script file model support csv and txt format, Mac, Extension, Full name must be filled, <password>, <IP Phone version> could be optional. Save it in your local PC after you fill based on the model format, select the relative manufacture on this page and upload.

Auto Provision

Upload Auto Provision File
Delete all extensions: <input type="checkbox"/>
Manufacturer: <input type="text"/> <input type="button" value="v"/>
The upload file must be ".csv" or ".txt".
Please choose file to upload: <input type="text"/> <input type="button" value="Browse..."/>
<input type="button" value="Upload"/>

Download Auto Provision script file demo
Auto Provision script file demo:
Right Click here to Save as Demo File (.csv)
Right Click here to Save as Demo File (.txt)

Chapter 5 Status

This chapter will introduce you the status of record list, call logs, system info, register status etc.

5.1 Record List

Check the record list of defined extension or conference, you can delete the record list. Click **【Status】** ----- **【Record List】** --- **【Monitor】** and **【Conference】** will be displayed as below:

Monitor List Interface

Monitor

		Monitor	Conference	
Extension:	<input type="text"/>	Delete		
Date:	<input type="text" value="Jul"/>	<input type="text" value="5"/>	<input type="text" value="2011"/> Go	
List of Monitoring Files				
S.No	Caller ID	Destination	Date	Options

Conference List interface

Conference

		Monitor	Conference		
Date:	<input type="text" value="Jul"/>	<input type="text" value="5"/>	<input type="text" value="2011"/> Go	Delete All	
List of Conference Record Files					
S.No	Conference Room	Date		Options	

5.2 Call Logs

Check call logs of extension by caller ID or callee ID. Click **【Call Logs】** to display the dialog as below:

Call Logs Interface

Call Logs

Start Date:	Jun	22	2012	Field:	Caller ID	<input type="text"/>	Filter
End Date:	Jun	22	2012				Download Delete
Call Start	Caller ID	Destination	Account Code	Duration (sec)	Disposition		



Duration in the call logs is not real charged duration, if you need billing, PSTN must support polarity reversal function, meanwhile, you must configure relevance parameters of polarity reversal in trunk configuration for the IP PBX.
For now our IPX-1980 can only FSK caller ID mode.

5.3 Register Status

Check SIP/ IAX2 User, and SIP/IAX2 Trunk status. Click **【Register Status】** to display the dialog as below:

Register Status

SIP Users Status	IAX2 Users Status	SIP Trunks Status	IAX2 Trunks Status
SIP Users Status:			
Name/username	Host	Dyn Nat ACL Port	Status
ChenDu/ChenDu	(Unspecified)	D 0	UNKNOWN
202	(Unspecified)	D N 0	UNKNOWN
201	(Unspecified)	D N 0	UNKNOWN
112	(Unspecified)	D N 0	UNKNOWN
111	(Unspecified)	D N 0	UNKNOWN
830	(Unspecified)	D N 0	UNKNOWN
829	(Unspecified)	D N 0	UNKNOWN
828	(Unspecified)	D N 0	UNKNOWN
827	(Unspecified)	D N 0	UNKNOWN
826	(Unspecified)	D N 0	UNKNOWN
825	(Unspecified)	D N 0	UNKNOWN
824	(Unspecified)	D N 0	UNKNOWN
823	(Unspecified)	D N 0	UNKNOWN
822	(Unspecified)	D N 0	UNKNOWN
821	(Unspecified)	D N 0	UNKNOWN
820	(Unspecified)	D N 0	UNKNOWN
819	(Unspecified)	D N 0	UNKNOWN
818	(Unspecified)	D N 0	UNKNOWN
817	(Unspecified)	D N 0	UNKNOWN
816	(Unspecified)	D N 0	UNKNOWN
815	(Unspecified)	D N 0	UNKNOWN
814	(Unspecified)	D N 0	UNKNOWN
813	(Unspecified)	D N 0	UNKNOWN
812	(Unspecified)	D N 0	UNKNOWN

5.4 System Info

Check OS version, firmware version and memory, etc from here.

Click **【System Info】** to display the dialog as below:

The screenshot shows a web interface with a left sidebar and a main content area. The sidebar contains a menu with items: Home, Basic, Inbound Control, Advanced, Status, Record List, Call Logs, Register Status, System Info, and System. The 'System Info' item is highlighted. The main content area is titled 'System Info' and has two tabs: 'System Info' (active) and 'Resources'. The content under the 'System Info' tab displays the following information:

- OS Version:** Linux IP PBX 2.6.22.18
- Uptime:** 17:02:32 up 3 days, 14:29, Load Average: 1.33, 1.17, 1.16
- Firmware Version:** Zycoo System v3.1
- Server Date & TimeZone:** Wed May 16 17:02:32 WST 2012 [Refresh](#)
- [Synchronize](#)
- Hostname:** IPPBX

Chapter 6 System

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

6.1 Network And Country

Configure WAN/ LAN IP, and tone zone.

Click **【System】** ---- **【Network And Country】** to display the dialog as below:

The screenshot shows the PLANET IP PBX configuration interface. The left sidebar contains a menu with the following items: Home, Basic, Inbound Control, Advanced, Status, System, Network & Country (highlighted), Troubleshooting, Network Advanced, Time Settings, Management, Data Storage, Backup, and Update. The main content area is titled 'Network & Country' and contains three sections: 'WAN Port Setup', 'LAN Port Setup', and 'Tone Zone Setting'. At the bottom are 'Save' and 'Cancel' buttons.

Item	Value
IP Assign	Static
Hostname	IPPBX
IP Address	192.168.1.100
Subnet Mask	255.255.255.0
Gateway	192.168.1.1
Primary DNS	8.8.8.8
Alternate DNS	
HTTP Port	80
Remote Administration	<input checked="" type="checkbox"/>
LAN IP Address	192.168.0.98
LAN Subnet Mask	255.255.255.0
Country	TW - Taiwan

Item	Explanation
IP Assign	Support Fixed IP, DHCP, PPPoE
Primary DNS	Specify a name server to resolve domain names
Alternate DNS	Specify a name server to resolve domain names
HTTP Port	HTTP login port
Remote Administration	Allow user to login form WAN or not.
LAN port setup	LAN IP address and subnet mask.
Tone Zone Setting	Define the tone zone for home country or place.



Please be remind: DNS response delay or no answer will effect the BLF function un-normal working, so if your network connection not so stable, but you still want the BLF function can works normal, we recommend to keep the DNS item "**Blank**".

6.2 Trouble Shooting

You can ping other network device through PLANET IP PBX and track network route by command "Traceroute".

Click **【System】** -- **【TroubleShooting】** to display the dialog as below:

The screenshot displays the PLANET IP PBX web interface. On the left is a navigation menu with the following items: Home, Basic, Inbound Control, Advanced, Status, System, Network & Country, TroubleShooting (highlighted), Network Advanced, Time Settings, Management, Data Storage, Backup, and Update. The main content area is titled 'TroubleShooting' and contains two tabs: 'Ping' and 'Traceroute'. The 'Ping' tab is selected, showing a text input field with '8.8.8.8', a 'Packets: 4' field, and 'Start' and 'Stop' buttons. Below this, the output of the ping command is displayed: 'PING 8.8.8.8 (8.8.8.8): 56 data bytes', followed by four lines of results: '64 bytes from 8.8.8.8: icmp_seq=0 ttl=50 time=435.3 ms', '64 bytes from 8.8.8.8: icmp_seq=1 ttl=50 time=606.1 ms', '64 bytes from 8.8.8.8: icmp_seq=2 ttl=50 time=474.4 ms', and '64 bytes from 8.8.8.8: icmp_seq=3 ttl=50 time=584.2 ms'. At the bottom, there is a summary: '--- 8.8.8.8 ping statistics ---', '4 packets transmitted, 4 packets received, 0% packet loss', and 'round-trip min/avg/max = 435.3/525.0/606.1 ms'.

6.3 DHCP Server Settings

ZX 50 series support DHCP , Click **【System】** -- **【Network Advanced】** -> **【DHCP Server Settings】** to show the following dialog:

DHCP Server Settings

DHCP Server Settings	DDNS Settings	VPN Settings
DHCP Server Settings		
DHCP Service:	<input type="checkbox"/>	
Start IP:	<input type="text" value="192.168.0.101"/>	
End IP:	<input type="text" value="192.168.0.200"/>	
Subnet Mask:	<input type="text" value="255.255.255.0"/>	
Gateway:	<input type="text" value="192.168.0.1"/>	
Primary DNS:	<input type="text" value="8.8.8.8"/>	
Lease Time:	<input type="text" value="1440minutes"/>	
<input type="button" value="Save"/> <input type="button" value="Cancel"/>		

6.4 DDNS & VPN

After configure DDNS, you can visit by domain remotely.

Click **【System】** -- **【DDNS & VPN】** to display the dialog as below:

DDNS Settings:

DDNS	
DDNS Enable:	<input checked="" type="checkbox"/>
DDNS Server:	<input type="text" value="planetddns.com"/>
Username:	<input type="text" value="testxx"/>
Password:	<input type="password" value="••••••"/>
Domain:	<input type="text" value="testxx"/> ,planetddns.com
Update Time(s):	<input type="text" value="120"/>
<input type="button" value="Save"/>	
Status:Disabled	

VPN Settings:

VPN	
VPN Mode:	N2N <input type="radio"/> L2TP <input checked="" type="radio"/>
VPN Enable:	<input checked="" type="checkbox"/>
Server Address:	<input type="text" value="192.168.1.1"/>
Username:	<input type="text" value="admin"/>
Password:	<input type="password" value="•••••"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	



1. DDNS supports the domain provided by Planet DDNS / Dyndns.org/ No-ip.com/ Zoneedit.com
2. VPN supports N2N/L2TP only

6.5 Time Settings

Click **【System】** -- **【Time Settings】** to display the dialog as below:

Time Settings

NTP Manual Time Set

NTP Server:

Time Zone:

Time Settings

NTP Manual Time Set

Year: (YYYY, eg: 2010)

Month: (MM, eg: 05)

Day: (DD, eg: 08)

Hour: (HH, eg: 09)

Minute: (MM, eg: 30)

Synchronize current PC time

Item	Explanation
NTP Server	Specify the NTP server that you wish to use. You may type either the domain name or the IP address of the server, and it may be either remote or local. The default server is pool.ntp.org. Be aware that the PBX needs to be able to connect to a NTP server for perfect function.

Time Zone	Select your time zone so that the system will set time based on the time zone.
Synchronize with current PC time	Click the button to synchronize the PBX time with the current PC time.

6.6 Management

Management on username, password, access permit, etc. Click **【System】 -- 【Management】** to display the diagram as below:

Management

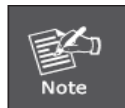
Management	Access Permit
Change Password	
Username: <input type="text"/> Password: <input type="password"/> New Username: <input type="text"/> New Password: <input type="password"/> Retype New Password: <input type="password"/>	
Apply	
Set Language	
Set Voice Language: <input type="text" value="English"/>	
Save	

(Show Advanced Options)

- **Change Password** You can change the password of admin here (default password is admin)
- **Set Language** Set voice language of the system. And you can set the SIP & Analog channel here by clicking "Show Advanced Options"

Click **【System】 - 【Management】 --- 【Access Permit】** to display the diagram as below:

Management	Access Permit
Deny all access GUI except for the ones in the list below <input type="checkbox"/>	
Save	
List of Permitted IP Address	
No Access Permit address defined!!	



After you added a permitted IP, you can only login the system by this IP, other IP address isn't effective to login the system

6.7 Data Storage

Upload the voicemail, monitor, conference, call logs, etc to the defined FTP server for storage. Click **【System】** - **【Data Storage】** to display the diagram as below:

FTP Data Storage

Data Storage	Data Storage Log
FTP Data Storage	
Enable Uploading: <input type="checkbox"/> Server Address: <input type="text"/> User Name: <input type="text"/> Password: <input type="text"/> Directory: <input type="text"/> Save	
Status: Disabled	

Upload Voicemail,Conference record,Monitor and Call logs to the defined FTP Server automatically when flash storage is over 40%.Then the history files will be removed out automatically.

(Note: NOT upload in working time).

Item	Explanation
Enable Uploading	Enable periodical FTP uploading.
Server Address	Set FTP Server address (IP address or Domain).
User Name	FTP account name.
Password	FTP account password.
Directory	Define a directory on the FTP server.




1. Upload Voicemail, Conference record, Monitor and Call logs to the defined FTP Server automatically when flash storage is over 40%. Then the history files will be removed out automatically
2. NOT upload in working time by default

6.8 Backup

Backup all the settings. Click **【System】 - 【Backup】** to display the diagram as below:

Backup

Backup		Upload Backup File	
List of Configuration Backups			
S.No	Name	Date	Options
1	test	Jun 25, 2012	Restore Delete 

- **Restore** Restore your selected backup file to system.
- **Delete** Delete your selected backup file.
-  Download your selected backup file to your PC. (Note: Please don't change the backup file name.)

6.9 Update

- Click **【Update】** to upgrade your system by uploading backup file:

Update

Upgrade System Package

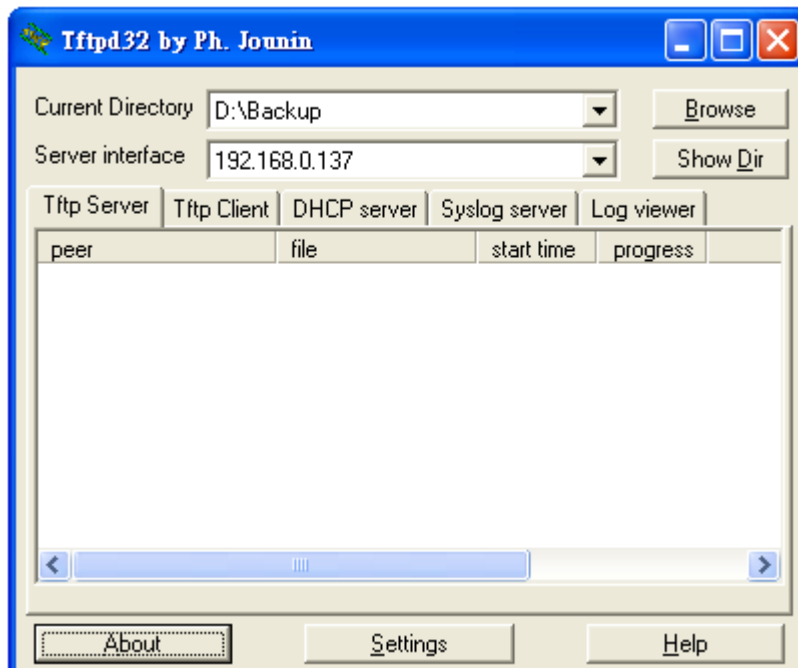
Restore Default Settings:

Please choose file to upload:

Extract the downloaded firmware package which includes one TFTP server and one upgrading file.



Run TFTP server, you will see the following interface:



Go into the "update" page, and upload firmware;

Enter the package name `uImage-md5`

Enter **TFTP Server IP address**, in this example we are "192.168.0.137"

Click **Update** button to finish upgrading system package after entering the TFTP Server IP.
Then system will reboot automatically.

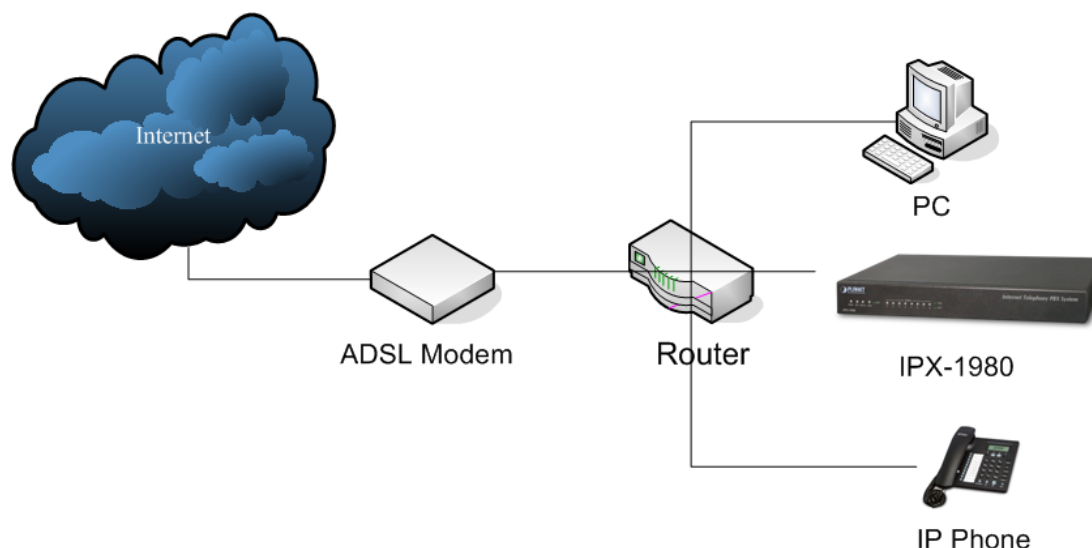
Chapter 7 Operating Instruction

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

7.1 How to connect the IPX-1980 IP PBX to the Internet

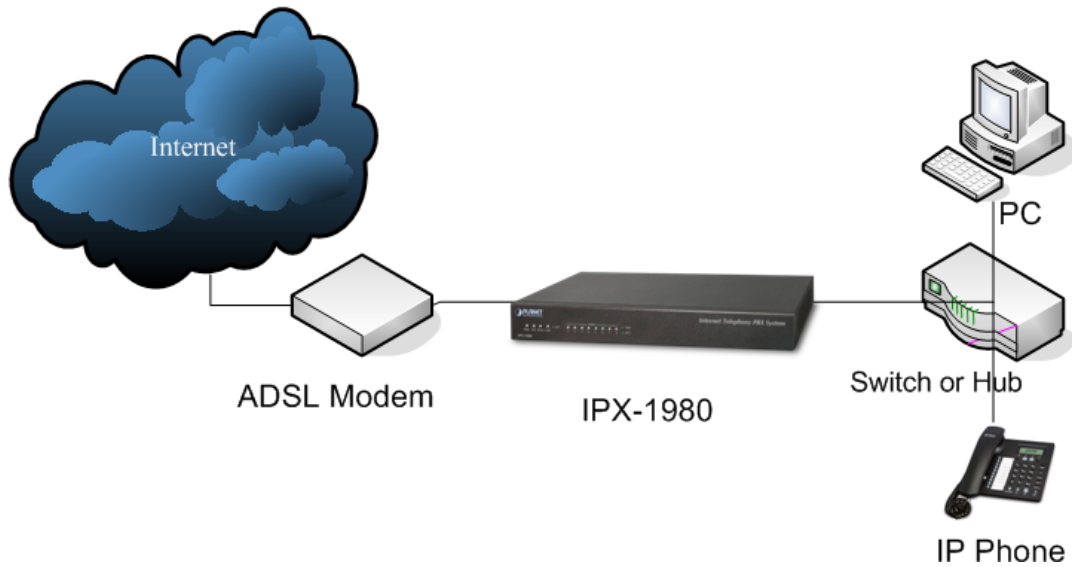
7.1.1 IP PBX behind the Router

If your office access the public network through router, you can put the IP PBX behind the router. You should connect the WAN port of the IP PBX to the LAN ports of the router, and you can also connect HUB or Switch to the LAN port of the IP PBX to enable some PC or IP Phone to access the public network..



7.1.2 IP PBX behind the Modem

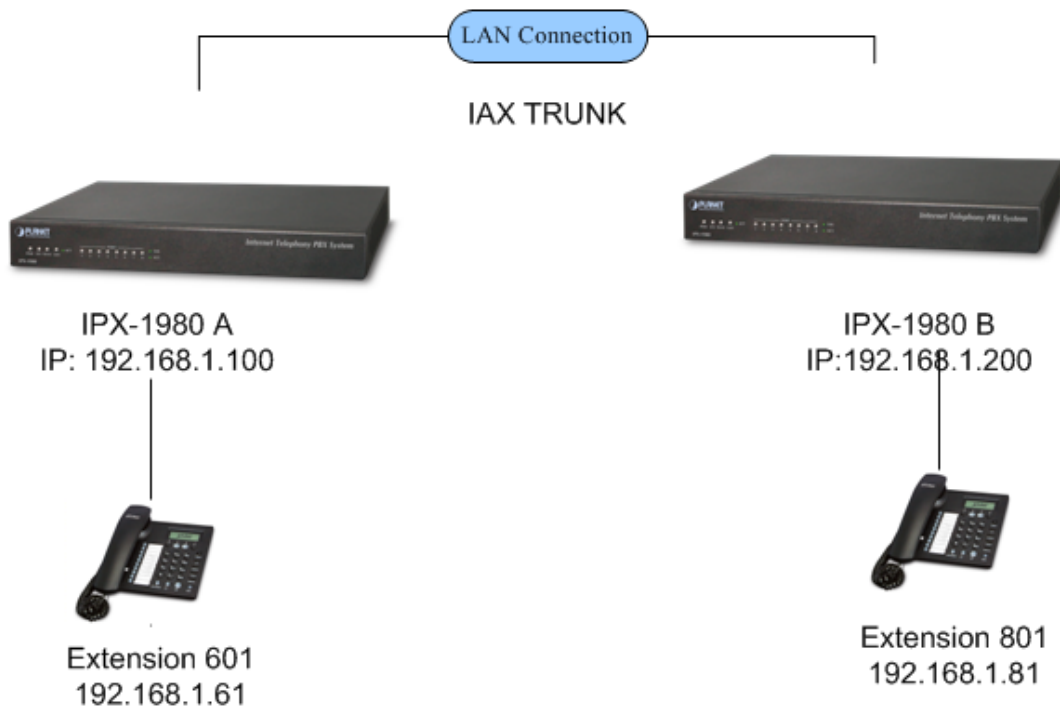
If you have the public IP and want to enable the IP PBX access the public network directly without router, then you should connect the Wan port of the IP PBX to the public network and connect HUB or Switch to the LAN ports of the IP PBX to enable your PC access the public network. (If you want to access the public network through Modem, then you should use the PPPoE function of the IP PBX and make the IP PBX dial-up to connect to the public network)



7.2 How to combine two IPX-1980 IP PBX in the same network

We start combining two IP PBX in the same network and then try to expand to different network.

Below is the structure of how to combine two IP PBX in the same LAN:



Register the IPX-1980-A as a peer in IPX-1980-B (via IAX2 trunk), so the extensions in IPX-1980-A can make calls to IPX-1980-B's extensions via this "special" trunk.

In above structure:

1. IP PHONEA registers to IPX-1980-A as extension 601.
2. IP PHONEB registers to IPX-1980-B as extension 801.
3. All the extensions under IPX-1980-A are in the format 6XX.
4. All the extensions under IPX-1980-B are in the format 8XX
5. Extensions under IPX-1980-A can make calls to extension under IPX-1980-B with format 8XX.
6. Extensions under IPX-1980-B can make calls to extension under IPX-1980-A with format 6XX.

Step 1: Set up a peer 699 in IPX-1980-A

In the page Trunks → Add a Trunk

Add a Trunk
✕

Provider Type:

Analog/GSM

VoIP Trunk

Peer

Peer Name:

Protocol:

Dial Plan:

Host:

NAT:

Prefix:

Without Authentication

Username:

Password:

Save
Cancel

Home	Options
Basic	
Inbound Control	
Advanced	
Options	
VoiceMail	
Conference	
Call Queue	
Music Settings	
DISA	
Follow Me	
Paging and Intercom	
Monitor	
Phone Book	
Pin Set	
Feature Codes	
Auto Provision	

Local Extension Settings

Local Extensions are

Operator Extension:

Global Ring Time Set(s):

Enable Transfer:

Enable Music On Ringback:

Allow multiple extensions to be assigned to one analog phone

Allow extensions to be AlphaNumeric (SIP/IAX users)

Default Settings for a New User

SIP IAX2

Agent NAT

VM Password Delete VMail

Save
Cancel

Peer Name:	IPX1980B
Peer Username:	699, Account of this Peer
Password:	699, IAX2 Log on password
Advance Options:	Select IAX protocol

Step 2: Set up an IAX trunk in IPX-1980-B to connect to IPX-1980-A via this IPX-1980B Peer.

In the page Trunks--> Add a Trunk

Add a Trunk
X

Provider Type:

Analog/GSM

VoIP Trunk

Peer

Description:

Protocol:

Dial Plan:

Register:

Host:

Outboundproxy:

Proxy Port:

Prefix:

Without Authentication

Username:

Password:

Step 3: Set Dial Rule in IPX-1980-B, all calls starting with 6 will be sent to IPX-1980-A.

In the page: Outbound Routes --> Add a Dial Rule

X

Rule Name:

PIN Set:

Place this call through:

Failover:

Dialing Rules: If the number begins with and followed by (more than) digits
[\(Define a custom pattern\)](#)

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

Step 4: Set the user 601 and Dial Plan in IPX-1980-A.

In the page: Extensions → Dial Plan

Create New User X

Name: Extension:

Password: Outbound CID:

VM Password E-mail:

Dial Plan:

Activate the change and apply the test:

1. Register an IP phone IP PHONEB to IPX-1980-B with 801 extension.
2. Register an IP phone IP PHONEA to IPX-1980-A with 601 extension.
3. 801 call 601. And you can see 601 will ring and you can pick up the call.

Above is the way to route IPX-1980-B's call to IPX-1980-A,

Accordingly, if you want to call from IPX-1980-A to IPX-1980-B, continue as below:

Step 5: Set Dial Rule in IPX-1980-A all calls starting with 8 will be sent to IPX-1980-B.

Rule Name:

PIN Set:

Place this call through:

Failover:

Dialing Rules: If the number begins with and followed by (more than) digits
(Define a custom pattern)

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

Step 6: Set the user 801 and Dial Plan in IPX-1980-B

Edit

Name: Extension:

Password: Outbound CID:

VM Password E-mail:

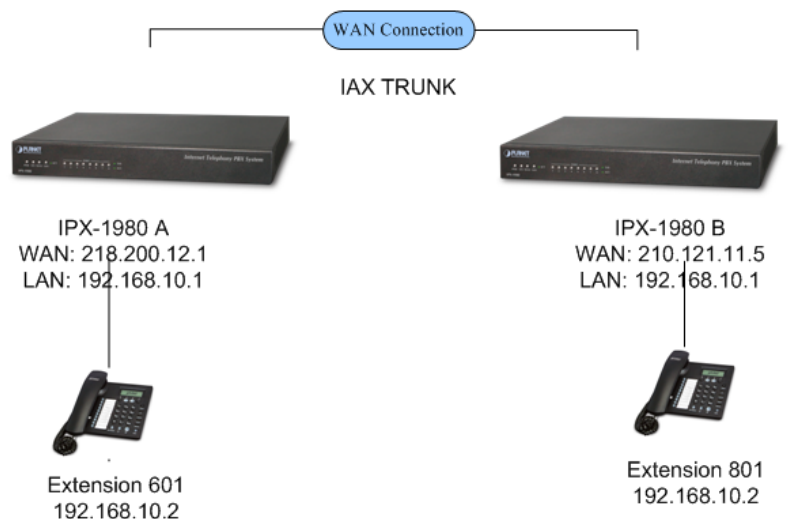
Dial Plan:

Activate the change and apply the test:

601 call 801, and 801 will ring and you can pick up the call.

7.3 How to combine two IPPBX in different network

The general environment for two IPX-1980 in different locations is: two IPX-1980 IP PBX are both in the Internet and using the public IP.



The configuration is same as above guide (7.2 Combine two IPX-1980 IP PBX in the same network), but use the public IP address as the "HOST" settings, set as below:

In the page Trunks of *IPX-1980-B*--> Add a Trunk

Add a Trunk
X

Provider Type:

Analog/GSM

VoIP Trunk

Peer

Description:

Protocol:

Dial Plan:

Register:

Host:

Outboundproxy:

Proxy Port:

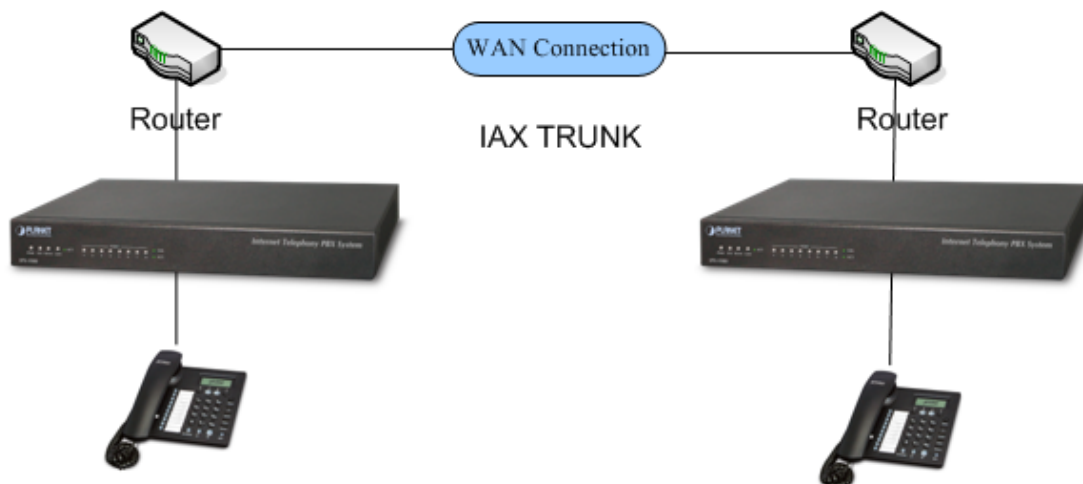
Prefix:

Without Authentication

Username:

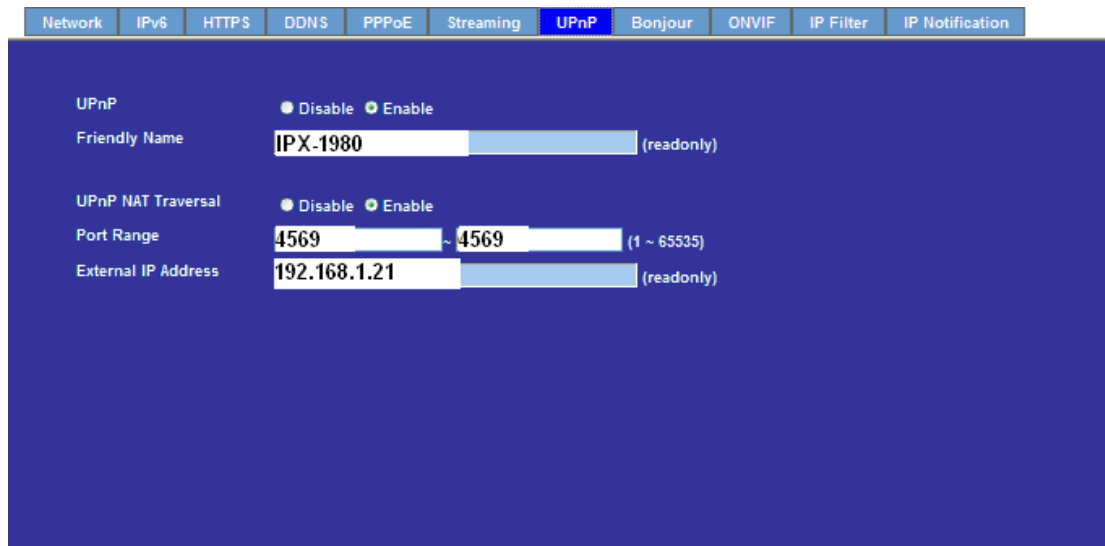
Password:

The general environment for two IPX-1980 IP PBX in different location and one or both two are behind router and using the private IP. So we need to make port forwarding in the router and make IPX-1980 IP PBX reach to each other.



Step 1: Set port forwarding in the router for IPX-1980-A

For the IPX-1980-A is behind the router, you need forward the IAX2 port in your router, so all the packets received on the router WAN port (210.11.25.127:4569) will be forwarded to the IPX-1980-A (192.168.1.21:4569). Below is the setting page in a Planet router (SG-4800) to make example:



Step 2: *Set up the Provider Host in IPX-1980-B*


Set up the service provider and calling rule in IPX-1980-B to make it register to IPX-1980-A. This method is almost the same as above, EXCEPT you need to use the 210.11.25.127 as the service provider instead of 192.168.1.21.

Step 3: *Set port forwarding in the router for IPX-1980-B*

Use the same method as Step 1 to do port forwarding in router-B for IPX-1980-B as above.

Setp4: *Combine two IPX-1980 and make calls*

Accordingly, set the 601 users in IPX-1980-A and 801 users in IPX-1980-B, and build the correct dial rules as above, you can make calls between two the IPX-1980 IP PBX.

 <p>Note</p>	<p>1. You can also apply a DDNS to get one fixed domain for both IPX-1980 IP PBX and connect to each other rather than using the Port Forwarding in the router.</p>
---	---

7.4 Remotely register IP phone to the IPX-1980

1. Build one DDNS if you have no static IP for accessing the internet.

To get one free domain from www.dyndns.org.

Like ours:

DDNS	
DDNS Enable:	<input checked="" type="checkbox"/>
DDNS Server:	dyndns.org
Username:	planetest
Password:	••••••••
Domain:	planetest.dyndns.org
Update Time(s):	120
Save	
Status: Disabled	

2. If the IPPBX is behind the Router, please open port 5060 for signaling and the 10001-10200 port for RTP. And 80 port for you web (default is 80 port), and make the port map for IPPBX.

3. Now ,if your dyndns and the port mapping is successfully set up, you can access your web Like:

<http://plnaetest.dyndns.org>

4. Then set the sip support.

Advanced--> Global SIP Settings → NAT Support

Home	Options																																																
Basic																																																	
Inbound Control																																																	
Advanced																																																	
Options	<table border="1" style="width: 100%;"> <thead> <tr> <th style="background-color: #e0e0e0;">General</th> <th style="background-color: #e0e0e0;">Global Analog Settings</th> <th style="background-color: #e0e0e0; border: 2px solid red;">Global SIP Settings</th> </tr> </thead> <tbody> <tr> <td colspan="3">General</td> </tr> <tr> <td>Caller ID:</td> <td colspan="2">Unknown</td> </tr> <tr> <td>Realm for digest authentication:</td> <td colspan="2"></td> </tr> <tr> <td>UDP Port to bind to:</td> <td colspan="2">5060</td> </tr> <tr> <td>Start RTP Port:</td> <td colspan="2">10001</td> </tr> <tr> <td>End RTP Port:</td> <td colspan="2">10200</td> </tr> <tr> <td>DTMF Mode:</td> <td colspan="2">auto</td> </tr> <tr> <td>Max Registration/Subscription Time:</td> <td colspan="2">3600</td> </tr> <tr> <td>Min Registration/Subscription Time:</td> <td colspan="2">60</td> </tr> <tr> <td>Default Incoming/Outgoing Registration Time:</td> <td colspan="2">120</td> </tr> <tr> <td colspan="3">NAT Support</td> </tr> <tr> <td>External ip:</td> <td colspan="2" style="border: 2px solid red;"></td> </tr> <tr> <td>External Host:</td> <td colspan="2"></td> </tr> <tr> <td>External Refresh:</td> <td colspan="2"></td> </tr> <tr> <td>Local Network Address:</td> <td colspan="2"></td> </tr> </tbody> </table>	General	Global Analog Settings	Global SIP Settings	General			Caller ID:	Unknown		Realm for digest authentication:			UDP Port to bind to:	5060		Start RTP Port:	10001		End RTP Port:	10200		DTMF Mode:	auto		Max Registration/Subscription Time:	3600		Min Registration/Subscription Time:	60		Default Incoming/Outgoing Registration Time:	120		NAT Support			External ip:			External Host:			External Refresh:			Local Network Address:		
General	Global Analog Settings	Global SIP Settings																																															
General																																																	
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Realm for digest authentication:																																																	
UDP Port to bind to:	5060																																																
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End RTP Port:	10200																																																
DTMF Mode:	auto																																																
Max Registration/Subscription Time:	3600																																																
Min Registration/Subscription Time:	60																																																
Default Incoming/Outgoing Registration Time:	120																																																
NAT Support																																																	
External ip:																																																	
External Host:																																																	
External Refresh:																																																	
Local Network Address:																																																	
VoiceMail																																																	
Conference																																																	
Call Queue																																																	
Music Settings																																																	
DISA																																																	
Follow Me																																																	
Paging and Intercom																																																	
Monitor																																																	
Phone Book																																																	
Pin Set																																																	
Feature Codes																																																	
Auto Provision																																																	
Status																																																	

NAT Support

External ip:
External Host:
External Refresh:
Local Network Address:
NAT mode:
Allow RTP Reinvite:

- **Extern IP** Replace with your external IP address this your public IP or domain
- **Extern Host** Replace with your external IP address this your public IP or domain
- **Extern Refresh** Set time for fresh, default 10
- **Local Network Address** Replace with your local network address and mask
- **NAT mode** If your IPPBX behind the Router, set default yes

5. If your settings are accomplished, now you can try to register one extension.

We try our phone and soft phone eyebeam.

1) If your phone is behind the router, and you must open the NAT for your extension.

Usually we open it.

Like:

Edit

Name:	<input type="text" value="801"/>	Extension:	<input type="text" value="801"/>
Password:	<input type="text" value="801"/>	Outbound CID:	<input type="text"/>
VM Password:	<input type="text" value="801"/>	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

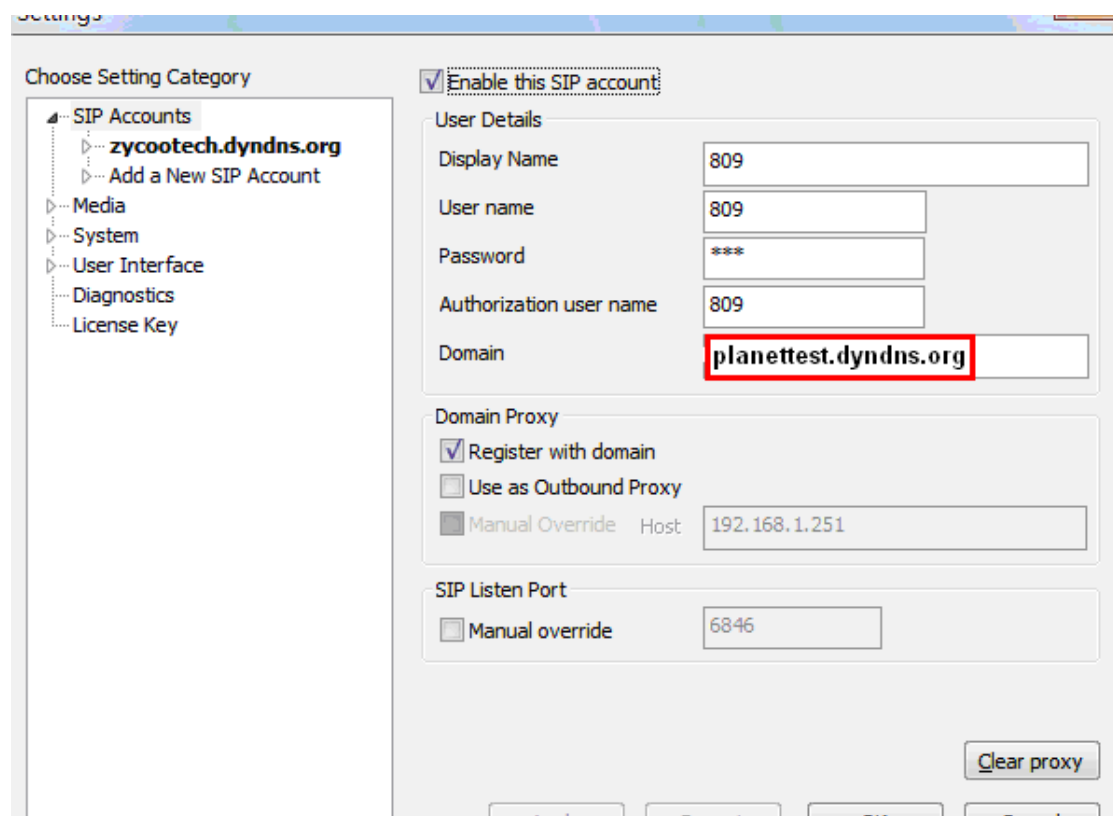
Audio Codecs Configure

alaw ulaw G.729 G.726 GSM Speex

Video Codecs Configure

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

Then we begin to register the soft phone.



The screenshot shows a web-based configuration interface for SIP accounts. On the left, a tree view under 'SIP Accounts' shows 'zycootech.dyndns.org' selected. The main area is titled 'Enable this SIP account:' and contains several sections:

- User Details:** Fields for Display Name (809), User name (809), Password (***), Authorization user name (809), and Domain (planettest.dyndns.org, highlighted with a red box).
- Domain Proxy:** Includes a checked 'Register with domain' checkbox, an unchecked 'Use as Outbound Proxy' checkbox, and a 'Manual Override' section with a 'Host' field containing '192.168.1.251'.
- SIP Listen Port:** Includes an unchecked 'Manual override' checkbox and a field containing '6846'.

Buttons for 'Apply', 'Reset', 'OK', and 'Cancel' are visible at the bottom, along with a 'Clear proxy' button.



It is registered.

Now try our own IP phone.



The screenshot shows a 'SIP Line Select' configuration window. At the top, there are tabs for 'SIP', 'IAX2', 'STUN', and 'DIAL PEER'. Below the tabs, a dropdown menu shows 'SIP 1' and a 'Load' button. The main section is titled 'Basic Setting' and contains a table of configuration parameters:

Register Status	Registered	Display Name	chen gang
Server Name	sip1	Proxy Server Address	
Server Address	planettest.dyndns.org	Proxy Server Port	
Server Port	5060	Proxy Username	
Account Name	809	Proxy Password	
Password	***	Domain Realm	
Phone Number	809	Enable Register	<input checked="" type="checkbox"/>

An 'APPLY' button is located at the bottom right of the configuration table.

It is registered.

7.5 How to customization on hold music

1. in [Music Management], [Upload music file] item,

1.1 Select music directory: in here you can choice Music 1-10.

1.2 Choice and Upload the music file (The sound file must be wav(16bit/8000Hz/Single), gsm, ulaw or alaw !! The size is limited in 15MB)

Music Management

Music Settings	Music Management
Music Management	
Directory:	Music 1 <input type="button" value="Load"/>
Files:	<input type="button" value="Delete"/>
Upload Music File <input type="button" value="Music Reload"/>	
Select Music Directory: Music 1 <input type="button" value="Load"/>	
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: <input type="text"/> <input type="button" value="Browse..."/>	
<input type="button" value="Upload"/>	

1.3 In music settings, choice the file you want to replace, and press save.

Music Settings

Music Settings	Music Management
Music On Hold Reference	
Music:	Music 1 <input type="button" value="Load"/>
Music On Ringback Reference	
Music:	Music 2 <input type="button" value="Load"/>
Music On Call Queue Reference	
Music:	Music 3 <input type="button" value="Load"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/> <input type="button" value="Music Reload"/>	

1.4 Please be remind: **[Music Reload]**, is last step to upload the music file to system, even user already press **[Upload]** in step 1.2 or press **Music reload** in step 1.3, it still need to press **[Music Reload]** to finished the last step and make music file replace.

Chapter 8 How to use Skype account in IPX-1980

[Answer]:

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

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

Sign in with the business account.

Create an account or sign in

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.


[Sign in](#) [Create an account](#)

Skype Name

[Forgotten your Skype Name?](#)

Password

[Forgotten your password?](#)







- Safe & Secure
- Quick & Easy
- Manage your account
- Change your settings

[Sign me in](#)

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

Settings and extras

 Payment settings	Stored payment details and Auto-recharge settings. View det
 Skype Manager	You are the administrator of Planet . Skype Manager · Membr
 Redeem voucher	Redeem your voucher or prepaid card. Redeem
 Skype WiFi	Learn about Skype WiFi

om

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](#)

W
th
S
C
M
e
s
fc
M

- 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 Manag 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.

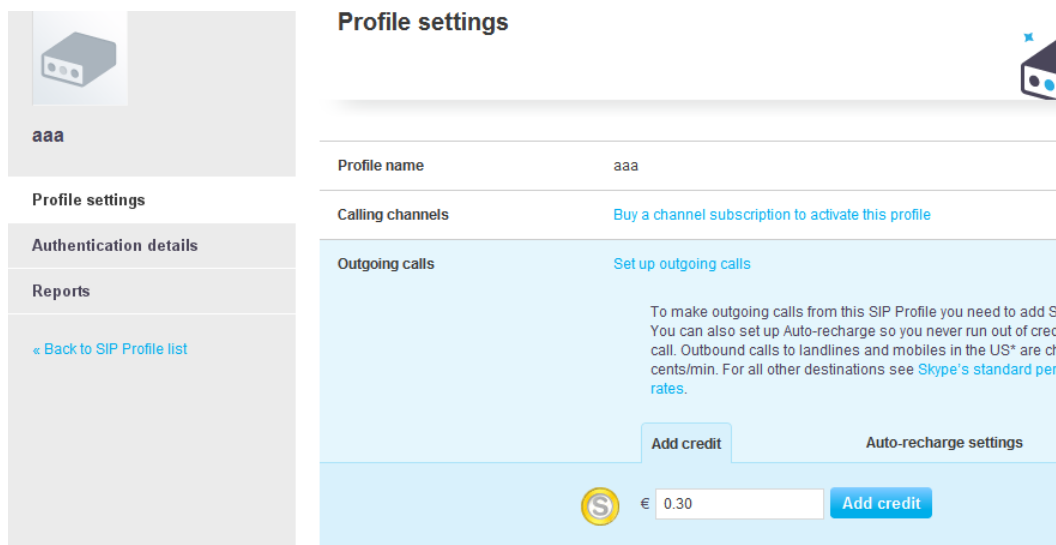
Choose a profile name

aaa 

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


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

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



The screenshot shows the 'Profile settings' page for a SIP profile named 'aaa'. The page is divided into a left sidebar and a main content area. The sidebar contains a profile icon (a grey box with three circles) and the name 'aaa'. Below the name are links for 'Profile settings', 'Authentication details', and 'Reports', along with a link to 'Back to SIP Profile list'. The main content area has a title 'Profile settings' and a small icon of a SIP profile. It contains several sections: 'Profile name' with the value 'aaa', 'Calling channels' with a link 'Buy a channel subscription to activate this profile', and 'Outgoing calls' with a link 'Set up outgoing calls'. Below these sections is a text block explaining that outgoing calls require adding credit and that auto-recharge can be set up. At the bottom, there is a section for adding credit, showing a balance of € 0.30 and an 'Add credit' button.

Then you can see the sip account information, 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

A. 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

In the trunk of our IPPBX setting:

Provider Type:

Analog Trunk

Custom Trunk

Peer

Description:

Protocol:

DialPlan:

Register:

Host:

Without Authentication

Username:

Password:

B. Set one outbound rule

X

Rule Name:

Place this call through :

Failover :

PIN Set:

Dialing Rules : If the number begins with and followed by (more than) digits (define a custom pattern)

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

C. Make an outbound call

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

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

D. Set inbound rule

Inbound Routing of our IPPBX:

X

Route

from Trunk

Destination