User's Manual

Internet Telephony PBX System

► **IPX-2100**

Copyright

Disclaimer

PLANET Technology does not warrant that the hardware will work properly in all environments and applications, and makes no warranty and representation, either implied or expressed, with respect to the quality, performance, merchantability, or fitness for a particular purpose. PLANET has made every effort to ensure that this User's Manual is accurate; PLANET disclaims liability for any inaccuracies or omissions that may have occurred. Information in this User's Manual is subject to change without notice and does not represent a commitment on the part of PLANET. PLANET assumes no responsibility for any inaccuracies that may be contained in this User's Manual. PLANET makes no commitment to update or keep current the information in this User's Manual, and reserves the right to make improvements to this User's Manual and/or to the products described in this User's Manual, at any time without notice. If you find information in this manual that is incorrect, misleading, or incomplete, we would appreciate your comments and suggestions.

Trademarks

The PLANET logo is a trademark of PLANET Technology. This documentation may refer to numerous hardware and software products by their trade names. In most, if not all cases, these designations are claimed as trademarks or registered trademarks by their respective companies.

CE mark Warning

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

Federal Communication Commission Interference Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Reorient or relocate the receiving antenna.
2. Increase the separation between the equipment and receiver.
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.

FCC Caution:

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-2100

Rev: 1.0 (Aug, 2013)

TABLE OF CONTENTS

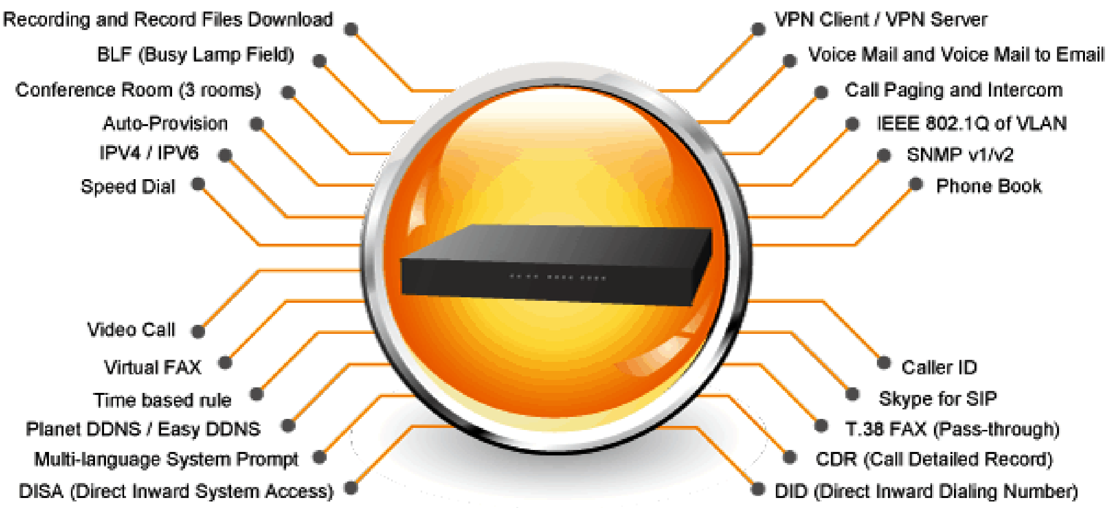
Chapter 1 Introduction

Intuitive, Ease-of-Use IP PBX Machine Management

PLANET IPX-2100 IP PBX telephony system is SIP based and optimized for the small and medium business in daily communications. The IPX-2100 is able to accept **100 user registrations**, and easy to manage a full voice over IP system with the convenience and cost advantages.

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

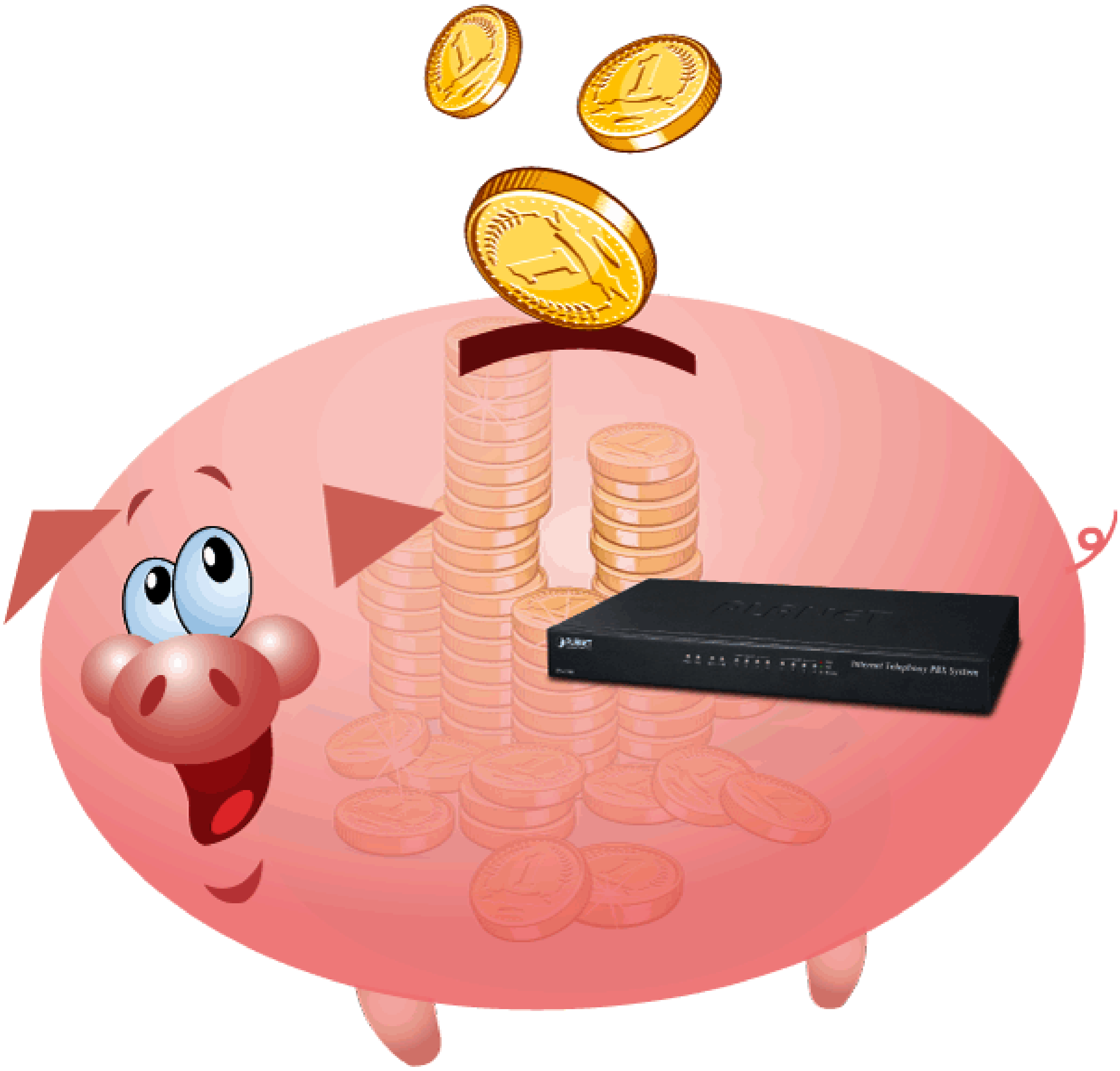
The IPX-2100 integrates up to 8 calls via the IPX-21FO (Foreign eXchange Office, FXO) module to become a feature-rich PBX system that supports seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints.



Replaces old PBX directly without any new wiring

Cost-effective, easy-to-install and simple-to-use, the IPX-2100 converts standard telephones into IP-based networks. It enables the service providers and enterprises to offer users traditional and enhanced telephony communication services via the existing broadband connection to the Internet or corporation network.

With the IPX-2100, home users and companies are able to save the installation cost and extend their past investments in telephones, conferences and speakerphones. The IPX-2100 can be the bridge between traditional analog systems and IP network with an extremely affordable investment.

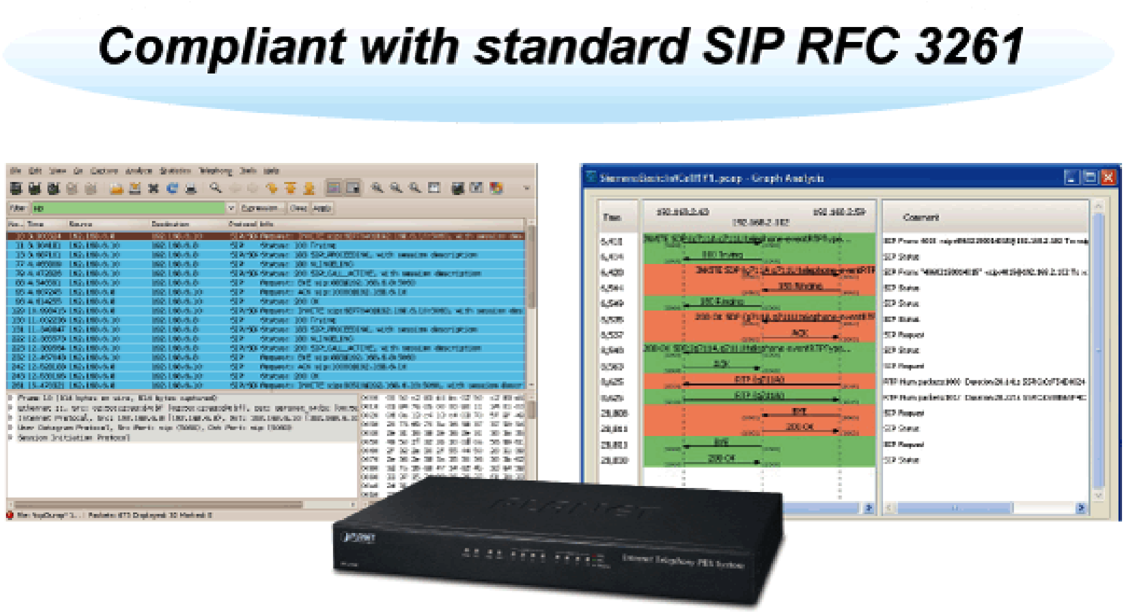


Distributed VoIP Network Infrastructure

For the new generation communication age, the IPX-2100 supports IPv6 and VPN (client / server) connection to provide users with more flexible and advantageous communication products. With PLANET DDNS function, the IPX-2100 also helps users to apply and remember the login information easier. Moreover, its multiple language features helps user to quickly and friendly manage the system.

Standard Compliance

Compliant with the Session Initiation Protocol 2.0 (RFC 3261), the IPX-2100 is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.



Green IP Office

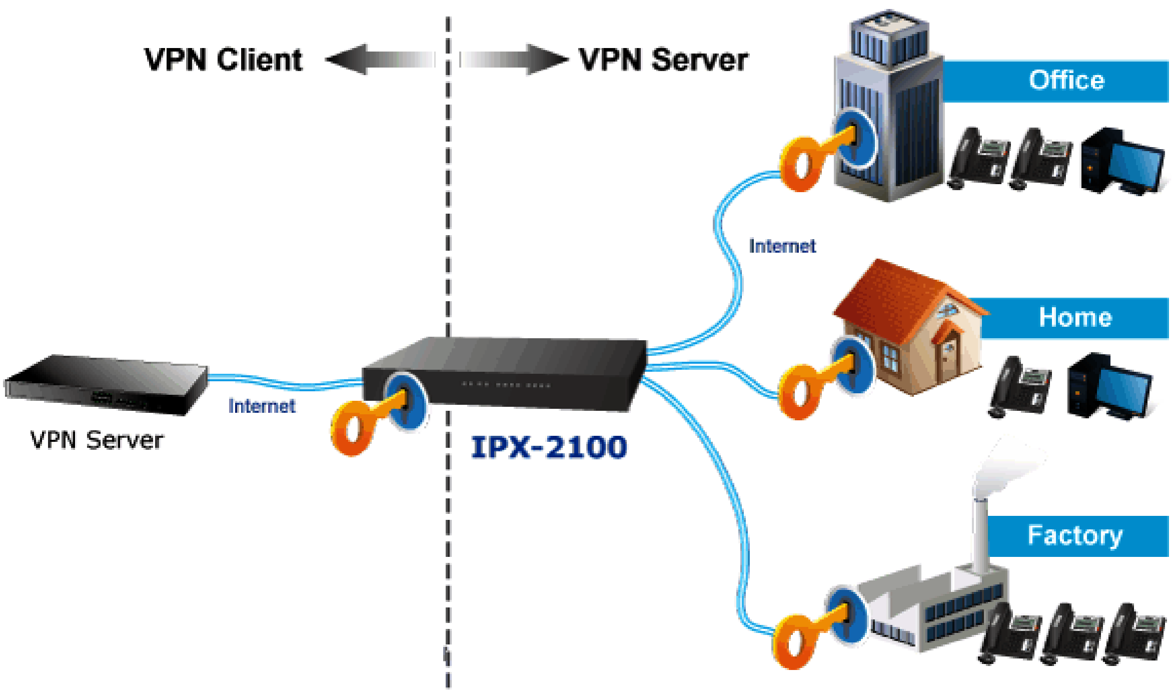
The Fax to Email / Email to Fax service by the IPX-2100 allows users to transfer / receive faxes directly to / from your email inbox as file attachments. It’s an easy and confidential way of receiving, storing and forwarding important fax documents, thus creating a paperless or green office.

Green Office (Fax to Email / Email to Fax)

Full Security with VPN Support

The IPX-2100 VPN securely and cost-effectively connects geographically disparate offices of an organization, creating one cohesive virtual network. The IPX-2100 VPN technology is also used by ordinary Internet users to connect to proxy servers for the purpose of protecting one's identity. It includes VPN server and client function that can support users full security login.

Supports Both VPN Client and Server Functions



1.1 Features

➢ **System Highlights**

- 20 concurrent calls and up to 100 registers
- HD voice codec G.722 for perfect voice quality
- Fax to Email / Email to Fax for Green Office
- Voicemail to Email for not missing any important message
- Paging and intercom function strengthens work efficiency
- Built-in SIP Proxy Server following RFC 3261
- Multiple Languages of GUI for international business
- Web based Control Panel for easy configuration and management of the system
- Hardware Echo Cancellation module for great and smooth communication
- Strong security features protect your system from hacking
- Supports maximum 8 ports FXO / FXS / GSM (on 2 slots)

➢ **Codec and Protocol**

- SIP 2.0 (RFC3261) / IAX2 compliant
- Audio Codec: G.722 / G.711-Ulaw / G.711-Alaw / G.726 / G.729 / GSM / SPEEX
- Video Codec: H.261 / H.263 / H.263+ / H.264
- DTMF: RFC2833, SIP INFO, In-band

➢ **Network and Security Features**

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

- **PBX Features**
  - Black List
  - BLF (Busy Lamp Field)
  - CDR (Call Detailed Record)
  - Conference Room (3 rooms)
  - DID (Direct Inward Dialing Number)
  - DISA (Direct Inward System Access)
  - DND / Feature Codes / Flash Operation Panel
  - Follow Me / Auto-Provision
  - IVR (Interactive Voice Responses)
  - Multi-language System Prompt
  - Multiple Languages of GUI
  - Phone Book / PIN Set
  - Record Files Download
  - Ring Group / SIP Trunk
  - Skype for SIP / Smart DID / System Log
  - T.38 Fax (Pass-through) / Time based rule
  - Virtual Fax / Voicemail & Voice Mail to E-Mail

- **Call Features**
  - Call Back / Call Forward / Call Group
  - Call Hold / Call Paging and Intercom
  - Call Park / Call Pickup / Call Queue
  - Call Record / Call Route / Blind Transfer
  - Attend Transfer / Call Waiting
  - Caller ID / Dial by Name
  - Customized IVR / on hold music / Transfer
  - Three-way Conference / Video Call

1.2 Package Contents

Thank you for purchasing PLANET Internet Telephony PBX system, IPX-2100. This Quick Installation Guide will introduce how to finish the basic setting of connecting the web management interface and the Internet. Open the box of the Internet Telephony PBX system and carefully unpack it. The box should contain the following items:

- Internet Telephony PBX System Unit x 1
- Quick Installation Guide x 1
- User's Manual CD x 1
- Power Adapter x 1 (12V)
- RJ-45 x 1
- Bracket x 2

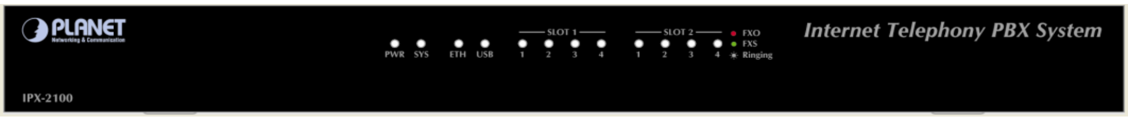
If any of the above items are damaged or missing, please contact your dealer immediately.

1.3 Physical Specifications

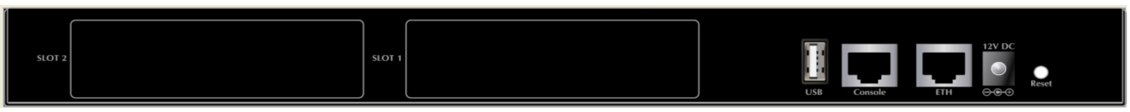
- **Dimensions**

Dimensions:	343 (L)x 154 (W)x 35 (H)mm
Net Weight:	1.4kg (without package)

- **Front Panel**



- **Rear Panel**



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

LED definitions

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

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	**ETH**	The ETH port supports auto negotiating Fast Ethernet 10/100 Base-TX networks. This port allows your IP PBX to be connected to an Internet Access device, e.g., router, cable modem and ADSL modem through a CAT.5 twisted pair Ethernet cable.
4	**FXO Port**	Connect to PSTN or PBX SLT line with RJ-11(Write) analog line.
5	**FXS Port**	Connect to PBX CO or single line telephone with RJ-11(Write) analog line.

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

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

1.4 Specifications

Product	**IPX-2100** **Internet Telephony PBX System (100 SIP Users Registrations)**
Hardware	
ETH	1 x 10/100Mbps RJ-45 port
2 Slots	Supports maximum 8 ports (FXO / FXS / GSM)
USB	Future Feature
Console	Console Interface
Protocols and Standard	
Standard	SIP 2.0 (RFC3261), IAX2
Protocols	RFC 793 TCP RFC 826 ARP RFC 1034, 1035 DNS RFC 1631 NAT RFC 2068 HTTP RFC 2131 DHCP RFC 2516 PPPoE RFC 3261, RFC 3311, RFC 3515 RFC 3265, RFC 3892, RFC 3361 RFC 3842, RFC 3389, RFC 3489 RFC 3428, RFC 2327, RFC 2833 RFC 2976, RFC 3263
Voice Codec	G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX
Video Codec	H.261/ H.263 / H.263+ / H.264
Fax Support	T.38 FAX(Pass-through)
Management	HTTP Web Browser
Voice Processing	DTMF detection and generation In-Band and RFC 2833, SIP INFO
Protocols	SIP 2.0 (RFC-3261), TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE
Internet Sharing	
Network Features	DDNS Client(Planet DDNS and Easy DDNS), DHCP Server IEEE802.1Q of VLAN IP Assignment (PPPoE / DHCP / Static) IPv4 / IPv6 SNMP v1/v2 Manual Configuration of Static Route Table Troubleshooting (Ping, Traceroute) VPN Client (Supports N2N / L2TP/PPTP/OpenVPN) VPN Server(PPTP/L2TP/OpenVPN Server)
Security Feature	Refuse SIP Register DoS Refuse Abort Invite Dos Refuse SSH Login DoS FireWall SRTP
Features	
PBX Features	Black List BLF (Busy Lamp Field) CDR (Call Detailed Record) Conference Room(3 rooms) DID (Direct Inward Dialing Number) DISA (Direct Inward System Access)

	DND / Feature Codes / Flash Operation Panel Follow Me / Auto-Provision IVR (Interactive Voice Responses) Multi-language System Prompt Multiple Languages of GUI Phone Book / PIN Set Record Files Download Ring Group / SIP Trunk Skype for SIP / Smart DID / System Log T.38 Fax(Pass-through) / Time based rule Virtual FAX / Voicemail &Voice Mail to E-Mail
Call Features	Call Back / Call Forward / Call Group Call Hold / Call Paging and Intercom Call Park / Call Pickup / Call Queue Call Record / Call Route / Blind Transfer Attend Transfer / Call Waiting / Caller ID Dial By name Customized IVR / on hold music / Transfer Three way conference Video Call
System Capacity	
System Capacity	20 Concurrent Call Legs Up to 100 IP Phone Registers/Extensions Recording(GSM/ default): 21,000 minutes; Wav: 3000 minutes Voicemail(GSM/ default): 21,000 minutes; Wav: 3000 minutes
Network and Configuration	
Access Mode	Static IP, PPPoE, DHCP
LED Indications	SYS: 1, LNK/Off ETH: 1, LNK/Off PWR: 1, LNK/Off FXO: Red FXS: Green
Dimensions (W x D x H)	343 x 154 x 35 mm
Operating Environment	-10~45 degrees C, 10~80% humidity
Power Requirements	Input: 100 ~ 240 Vac Output: DC 12V / 2.0 A
EMC/EMI	CE, FCC Class B, RoHS
Remarks: T.30/ T.38 support is dependent on fax machine, SIP provider and network / transport resilience.	

Chapter 2 Installation Procedure

2.1 Web Login

Step 1. Connect a computer to an ETH port on the IPX-2100. Your PC must set up to the same domain of 192.168.0.X as that of the IPX-2100.

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-2100: 192.168.0.1 in the URL address box.

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

(Default IP)

Default ETH IP: **192.168.0.1**
Default User Name: **admin**
Default Password: **admin**

Figure 2-1. Login page of the IPX-2100

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

2.2 Configuring the Network Setting

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

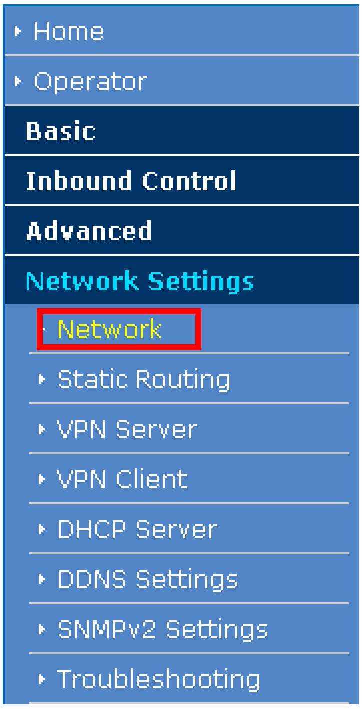


Figure 2-2. Network & Country Button

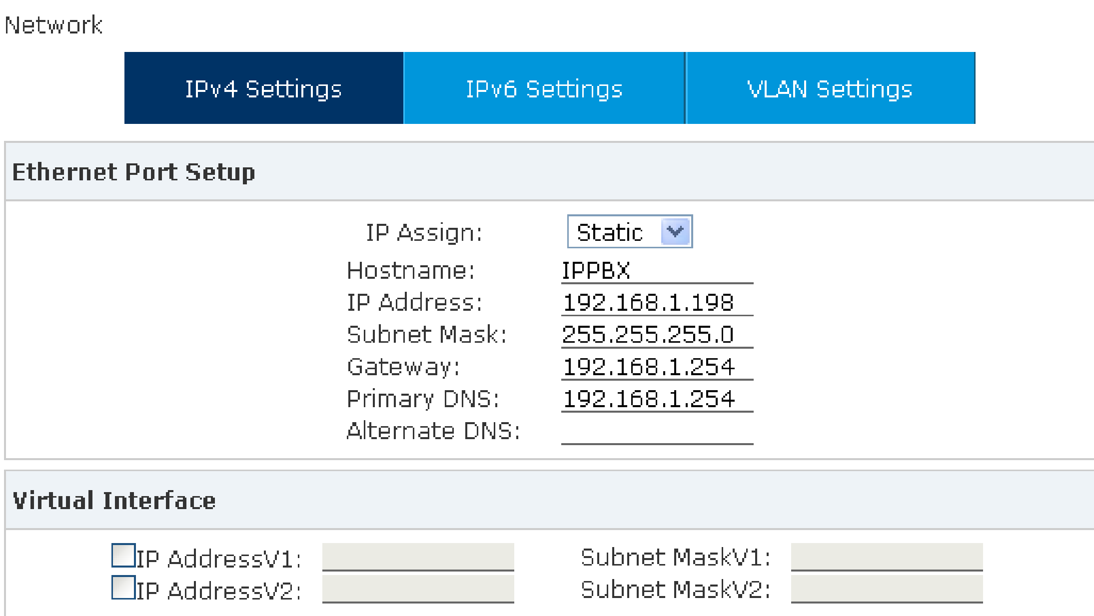


Figure 2-3. Network Setting page

Step 2. Edit your ETH port IP information .

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

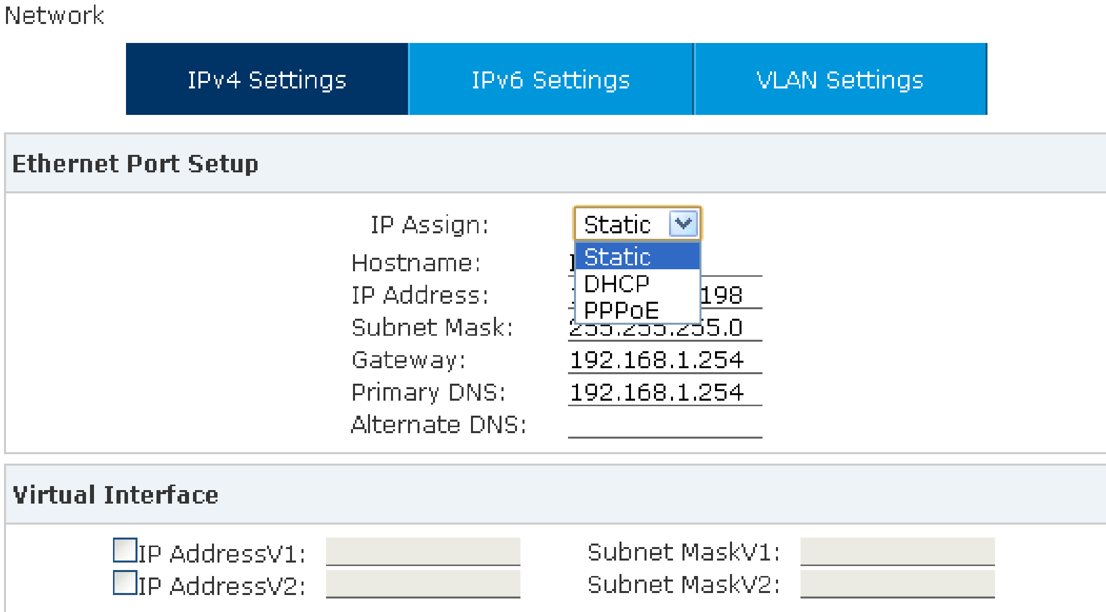
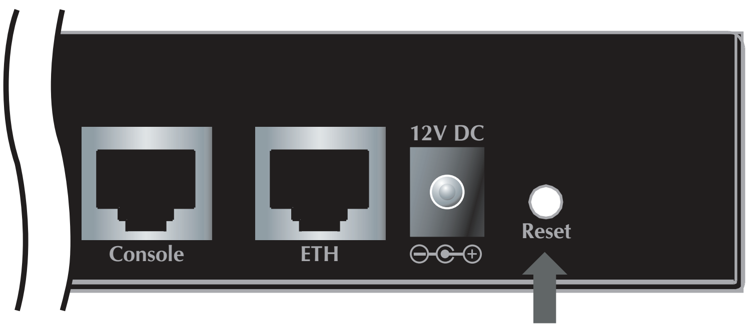


Figure 2-4. Selection of IP Connection Type

2.3 Changing IP Address or Forgotten Admin Password

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



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

Chapter 3 Basic Configuration

3.1 Preparation Before Operation

What kind of IP phone can be used with the IPX-2100 IP PBX?

- Our IPX-2100 is based on SIP 2.0 (RFC 3261); any IP phone model based on the same protocol can work with the IPX-2100.

3.2 Before Making a Call

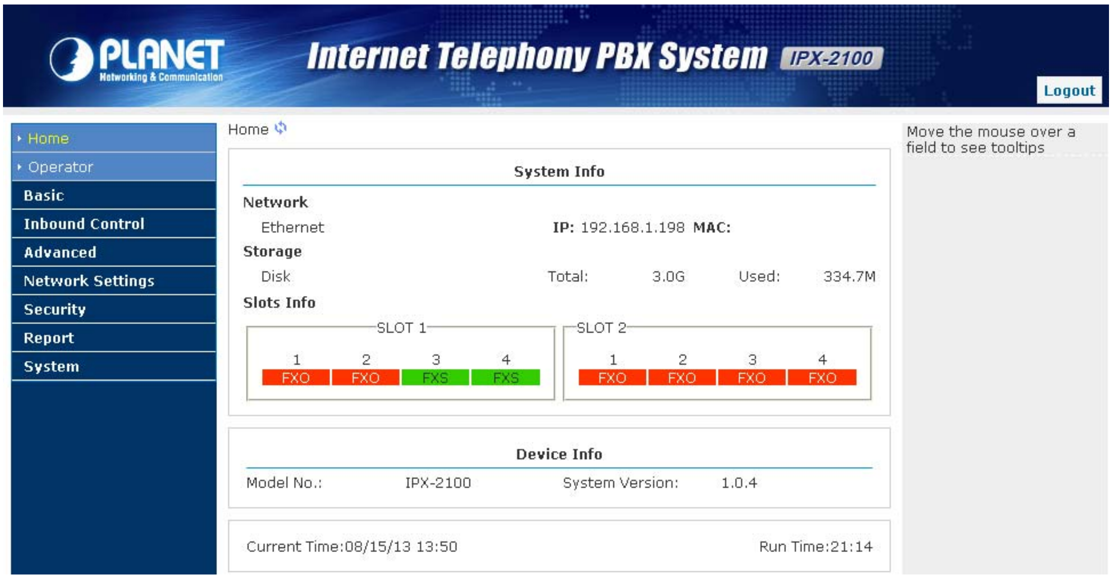
3.2.1 System Information

Default ETH IP: **192.168.0.1**
Default Name: **admin**
Default Password: **admin**

Note	1. To login to the IPX-2100, your PC must use the same domain as the EHT IP address of the IPX-2100. 2. For security reason, please modify the user name and password after you login. You can modify it on this page: “System”---“Management”

3. <<Warnning!>> **Every Time after saving the change, please press the Activate Changes to make modification effective.**

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



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

Note

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

Commonly Used Button

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

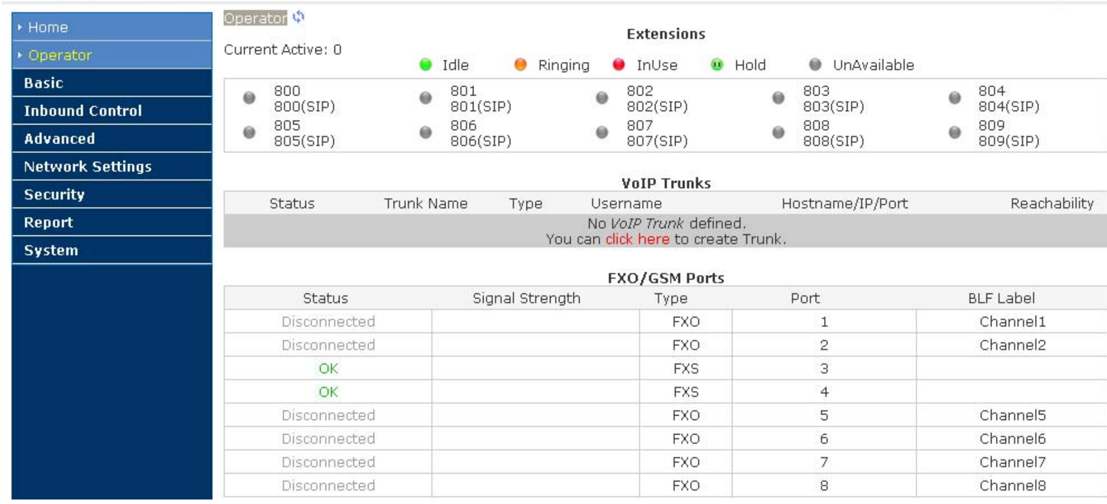
1	**Logout**	Logout the Web panel
2	**Activate Change**	Activate the changes for your current configuration

System Menu

System Menu includes the following sub menu:

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

3.2.2 Operator



Display all the Extension, VoIP Trunk and Slot information.

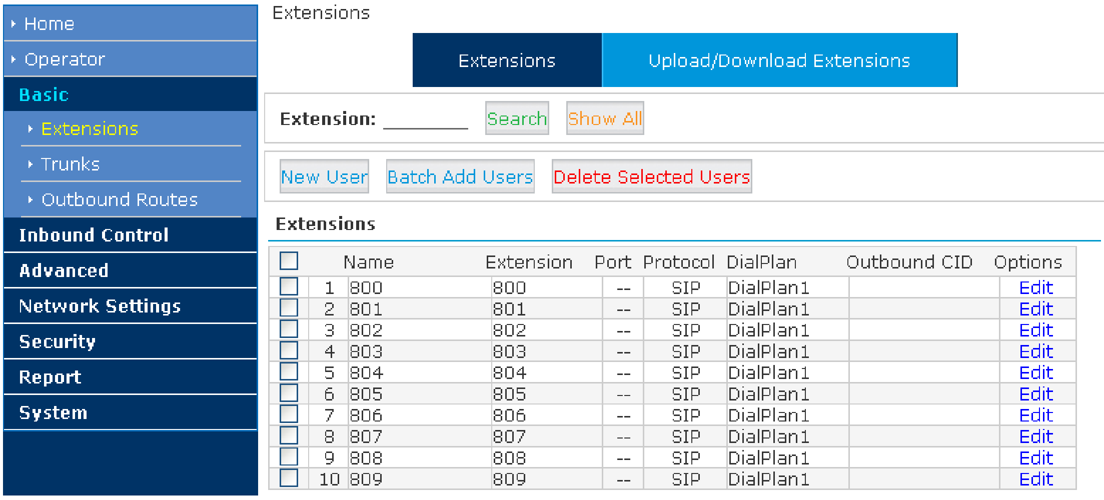
About extension:

1		Idle
2		Ringing
3		In use
4		Hold
5		Unavailable

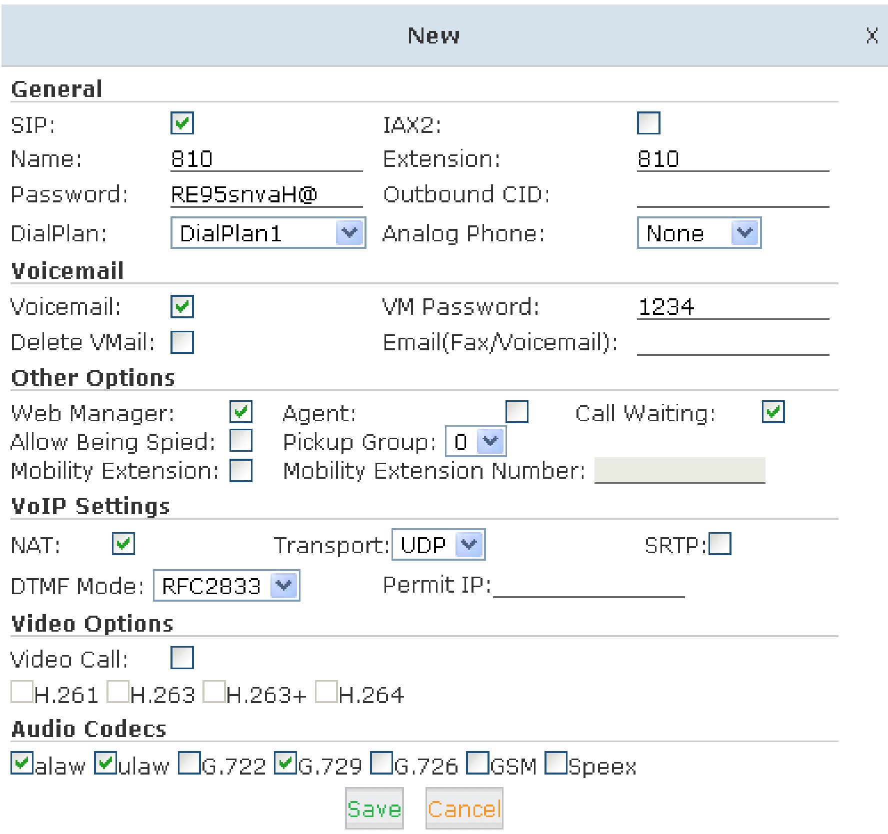
3.2.3 Basic Configuration

Configure Extensions

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



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



Extension Settings

Item	Explanation
SIP/IAX2	Choose extension protocol.
Name	Extension Name (English Character Only), e.g. Tom.
Extension	Extension Number connected to the phone, e.g. 888.
Password	Same password as voicemail. (4-16 digits, e.g.123456)
Outbound CID	Override the caller ID when dialing out with a trunk.
Dial Plan	Please choose the Dial Plan which is defined in the menu "Outbound Routes".
Analog Phone	Please select the related FXS port for your analog phone.
Voicemail	Select this option to open the voicemail account
VM Password	Set password for Voicemail, e.g. "1234"
Delete VMail	Check this option to delete voicemail from system after it's sent to mail box.
Email (Fax/Voicemail)	Extension user's mail box, which is used for receiving fax or voicemail (you need to open the function to fax to email/voicemail), e.g. Tom@gmail.com
Web Manager	It's allowed to login Extension Management Panel to manage extension like voicemail, call recording, call transfer, etc when you select this option.
Agent	Check this option to set this extension user as agent.
Call Waiting	Enable call waiting
Allowing Being Spied	Check this option to allow being spied.
NAT	Check this option if extension user or the phone is located after the NAT (Network Address Translation) available gateway.
Pickup Group	Select the Pickup Group which the extension user belongs to.
Mobility Extension	After checking this option, you must set mobility extension number. User can make calls to the IP PBX server with this mobility number, and have all rights of this extension, e.g. Outbound Call, Internal Call, Listen to the voicemail.
Transport	Select the Transport Protocol: UDP, TCP, TLS
SRTP	Enable SRTP
DTMF Mode	Default DTMF is rfc2833. It can be changed if necessary.
Video Call	Check to enable video call for this extension. And select the audio codecs you need to use.
Permit IP	Set computer permitted IP to visit this IP PBX, e.g.192.168.1.77or

	192.168.10.0/255.255.255.0. Computer with other IPs is not allowed to visit this IP PBX.
Audio Codec	Select what audio codec you need to use.

Note	1. There are few default extensions which number started with "8XX", you can add or delete extension by your requirement 2. Maximum extensions: 100. 3. For security reason the default password is random character or number e.g. BB%ChH64rI, and every time when you reset to default system, it will randomly have a new password again

Upload/Download Extensions

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

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

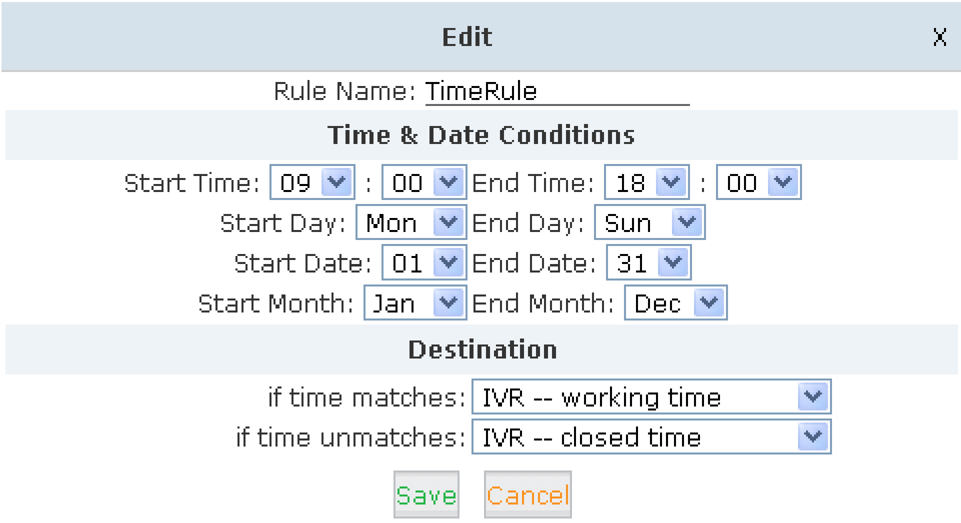
Select the extension file to upload from【Upload Extensions】

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

3.2.4 Time-based Rules

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

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



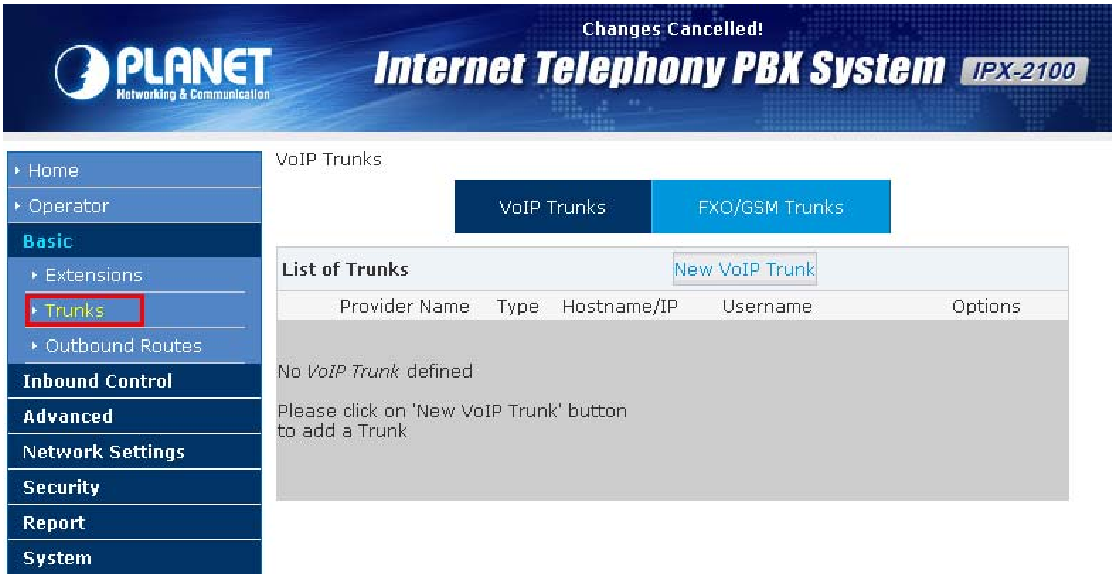
New Time Rule:

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

3.3 Outbound Call

3.3.1 Trunks

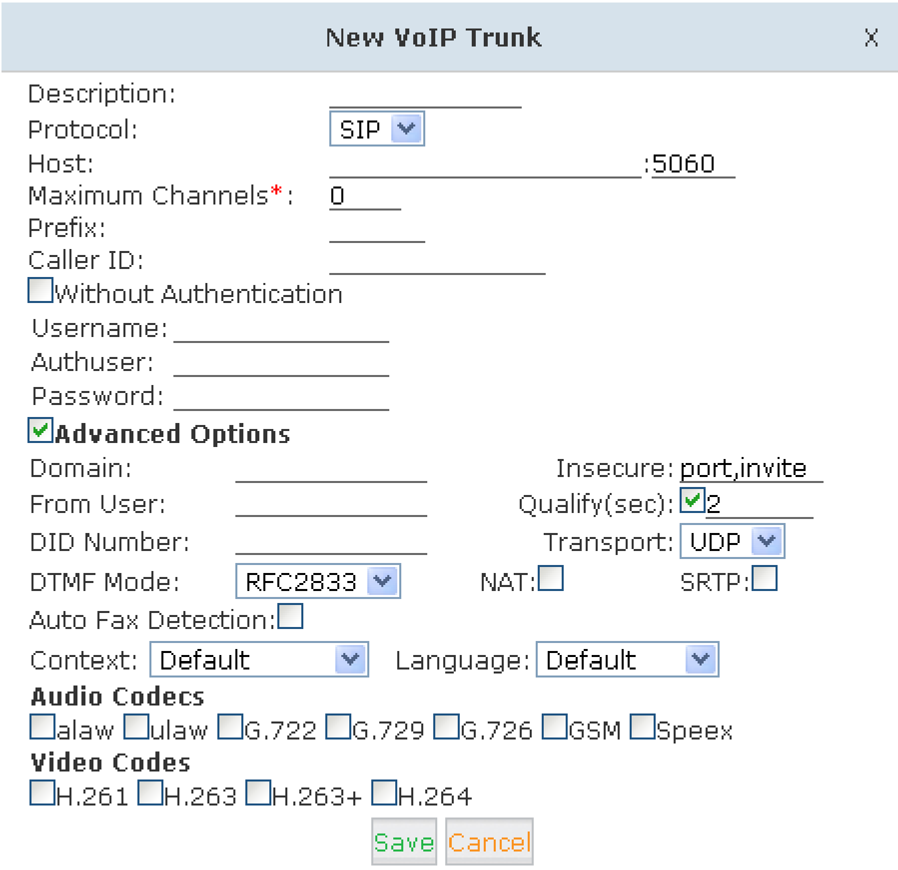
If you want to set up outbound call to connect to PSTN (Public Switch Telephone Network) or VoIP provider, please configure on this page: 【Basic】->【Trunks】



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

VoIP Trunks

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

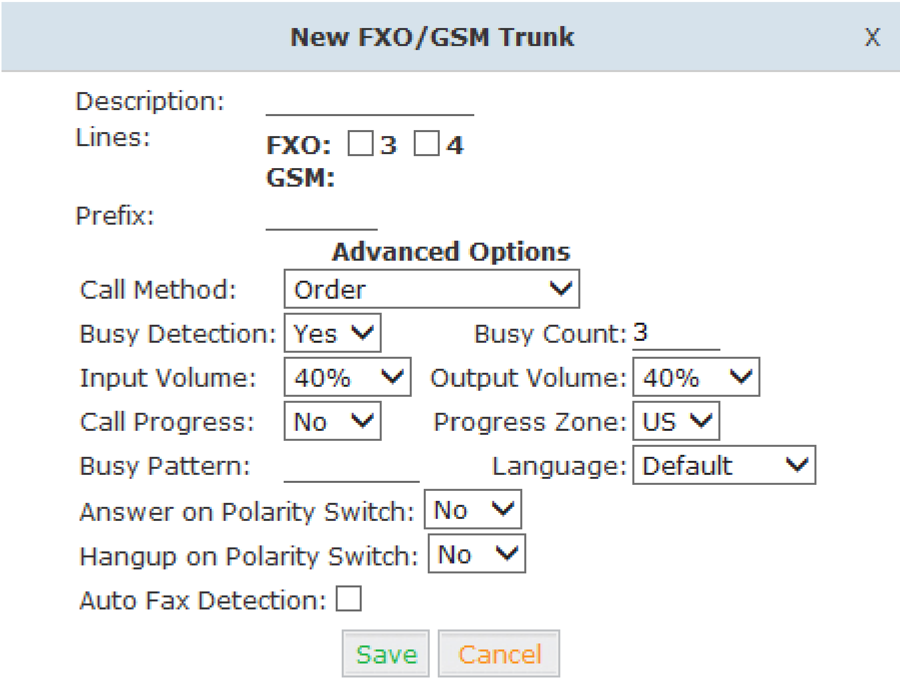


Item	Explanation
Description	Define the VoIP(figure or character).
Protocol	Select protocol for outbound route, SIP or IAX2.
Host	Set host address (provided by VoIP Provider).
Maximum Channels	Set maximum channels for simultaneous call. (Only for outbound call; “0” = no limitation).
Prefix	The prefix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

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

2) FXO/GSM Trunk

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



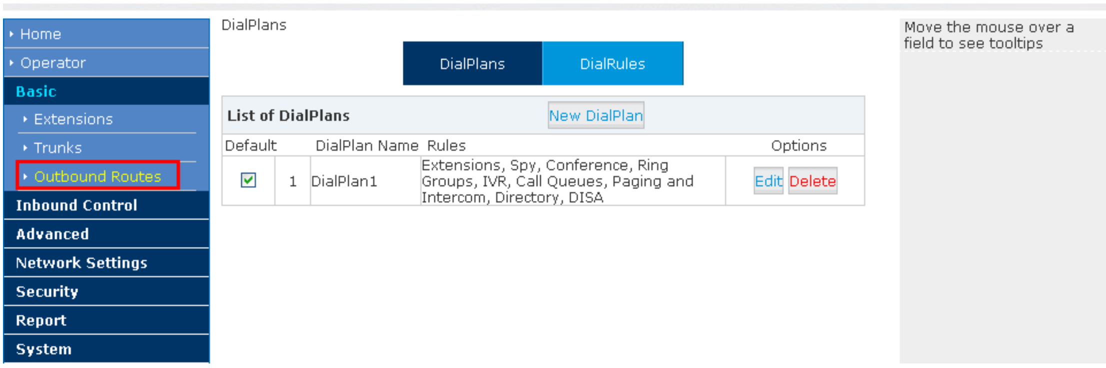
Item	Explanation
Description	Define the description for this trunk (figure or character).
Lines	Available line
Prefix	The prefix will be added to the dialed number automatically when this trunk is in use.
Advanced Options	Advanced Options for this trunk, e.g. Call Method, Busy Detection, etc.

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

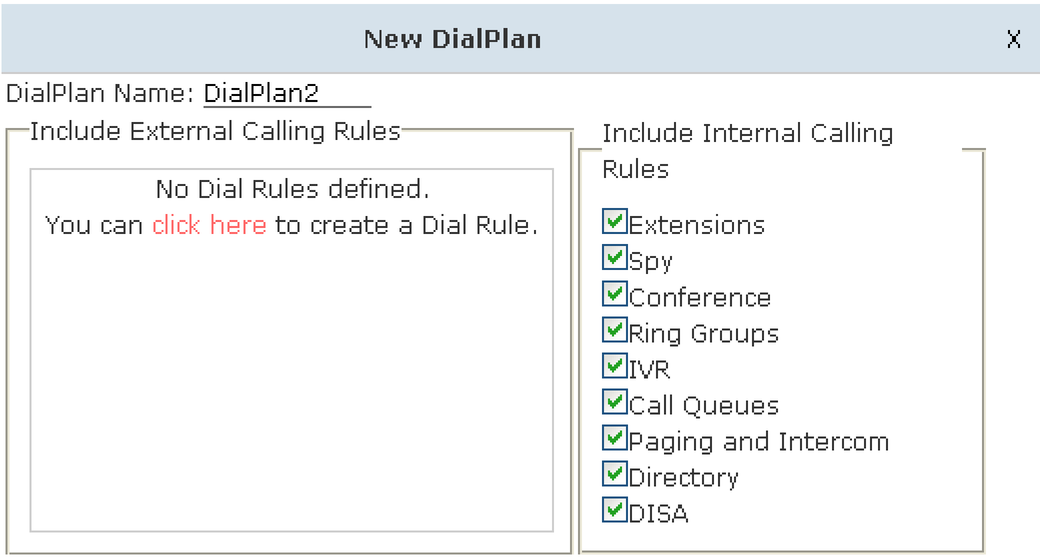
3.3.2 Outbound Routes

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

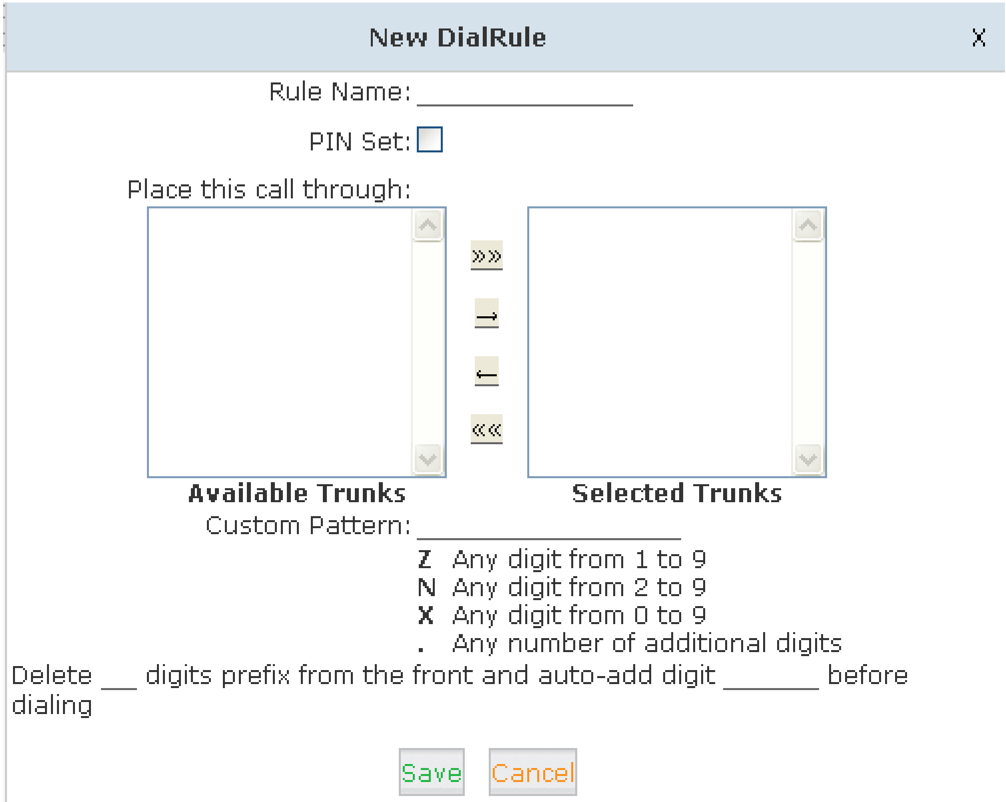
Please configure on this page: 【Basic】->【Outbound Routes】



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



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



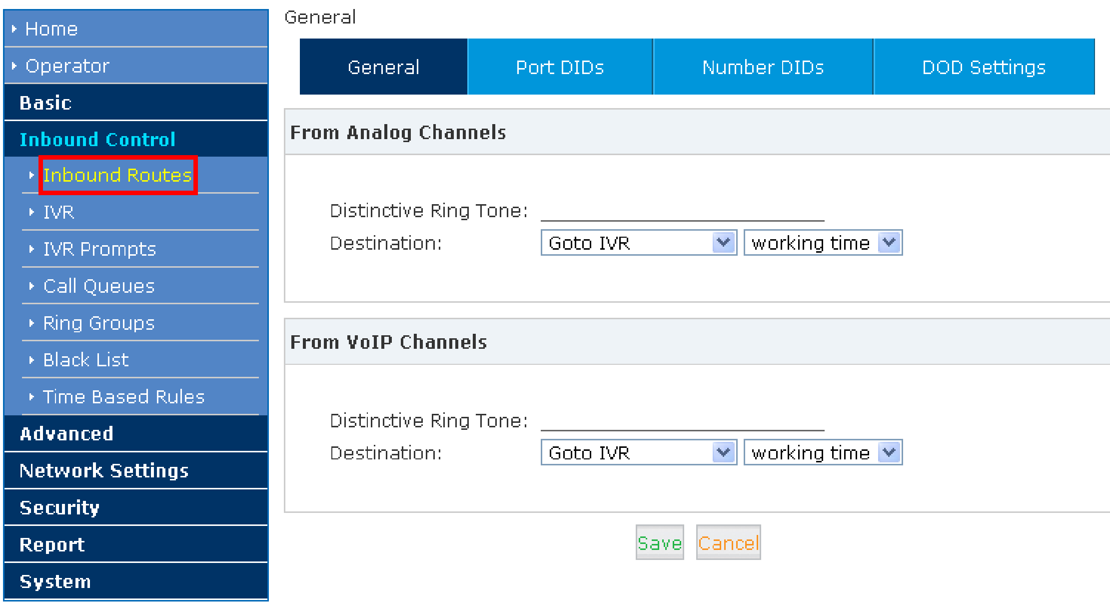
Item	Explanation
Rule Name	Define the name for the dial rule.
Pin Set	Input this Pin when you use this dial rule.
Place this call through	Select a trunk for this dial rule
Custom Pattern	**N** any figure from 2 to 9 **Z** any figure from 1 to 9 **X** any figure from 0 to 9 **.** One figure or multi-digit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

3.4 Inbound Call

3.4.1 Inbound Routes

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

Please configure it on this page:【Inbound Routes】



General

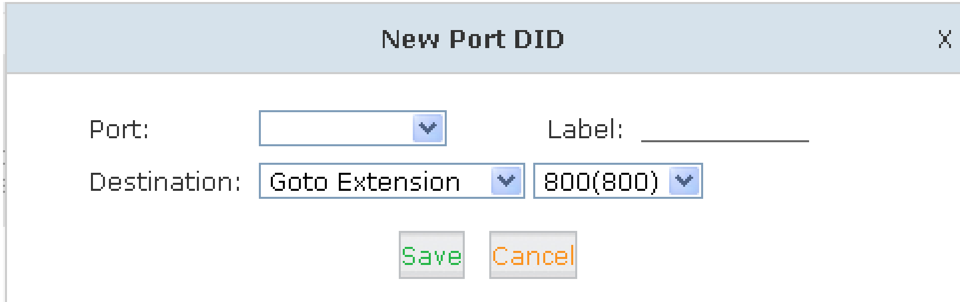
Distinctive Ring Tone: mapping the custom ring tone file, e.g. set distinctive ring tone as "External", the phone will play this ring tone when receiving the call. Note: The phone must support such feature as well.

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

Port DIDs

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

Click【Port DIDs】->【New Port DIDs】:

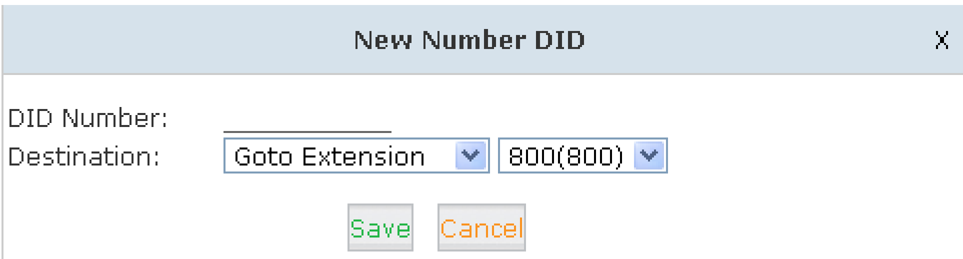


Item	Explanation
Port	Select the port for outbound line.
Label	Set a label for this port. When incoming calls are from this port, the label will be displayed.
Destination	Incoming calls will access directly to this destination (extension user, call queue, conference, or IVR).

Number DIDs

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

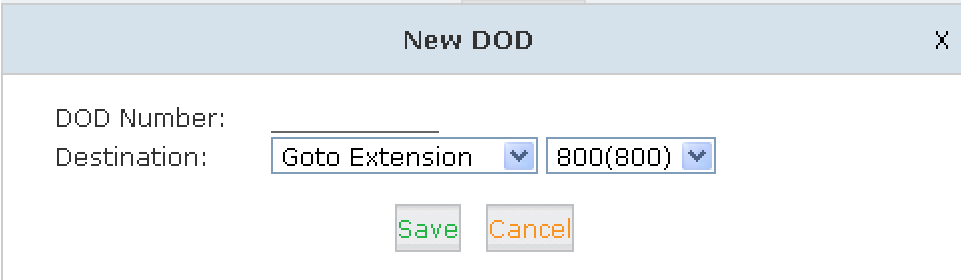
Click【 Number DID】->【New Number DID】:



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

DOD Settings

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

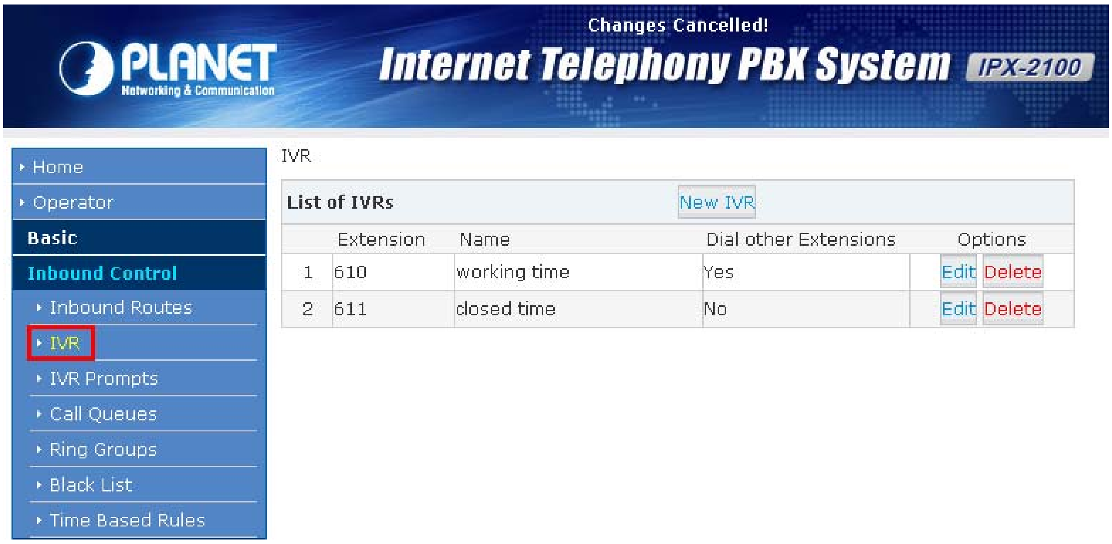


Item	Explanation
DOD Number	Set the DOD number, and use it to match the Caller ID. If matched, the call will access to the defined destination.
Destination	Outbound calls will access directly to this destination (extension user, call queue, conference, or IVR).

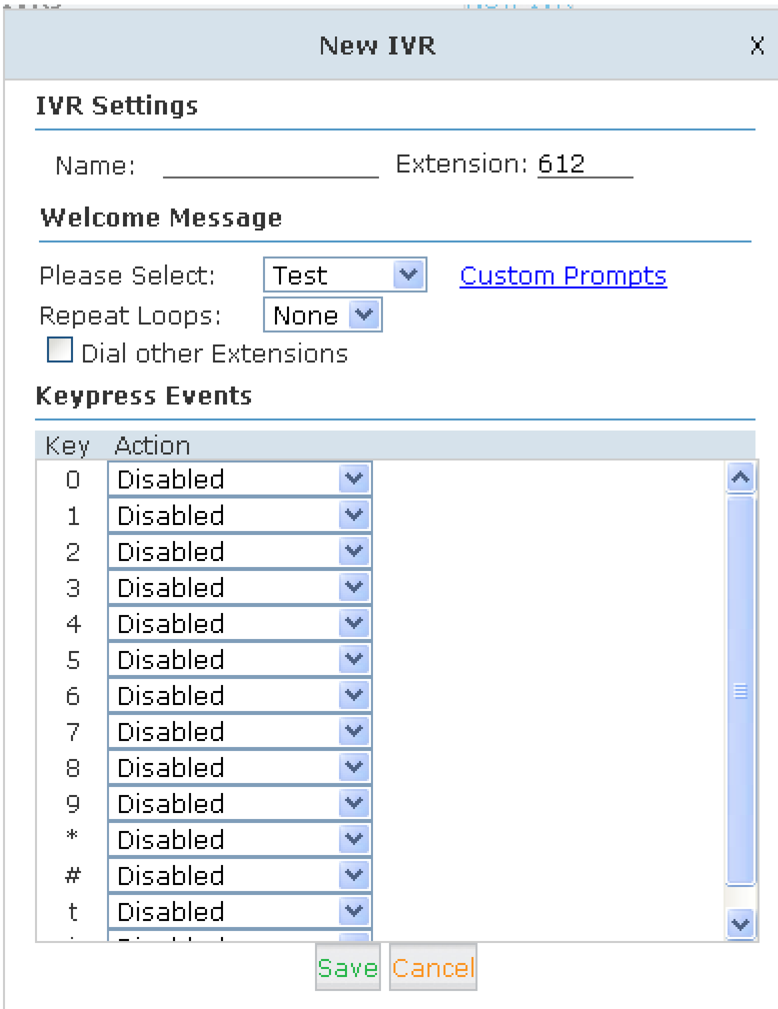
3.4.2 IVR

IVR will improve office efficiency based on your requirement.

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



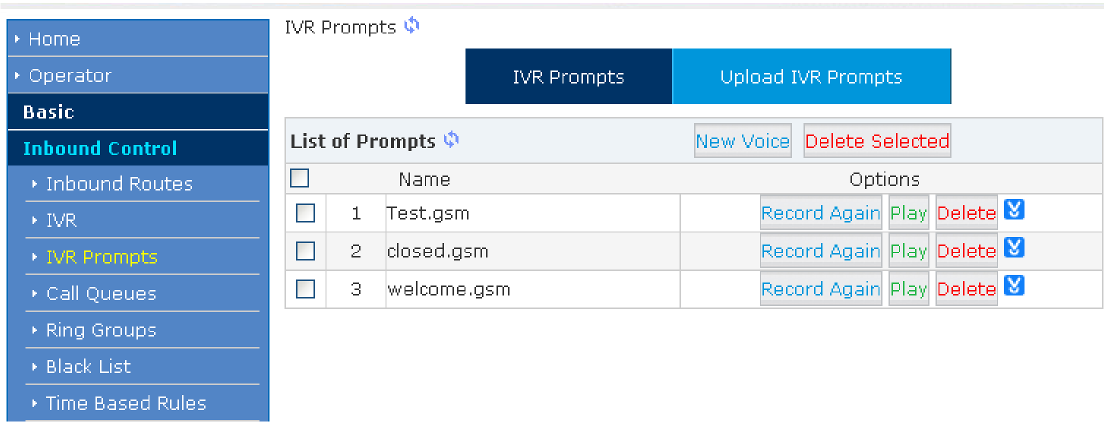
Click【New IVR】to create a new IVR:



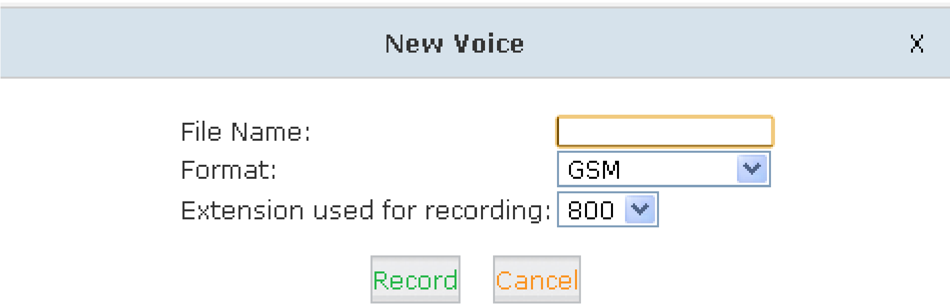
Item	Explanation
Name	Set a name for the IVR
Extension	If you want to listen to the IVR by dialing extension, please input an extension Number.
Please Select	Select IVR audio file, please configure in this page:【IVR Prompts】
Repeat Loops	Loop times to repeat playing the IVR prompt.
Dial Other Extensions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

3.4.3 IVR Prompts

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

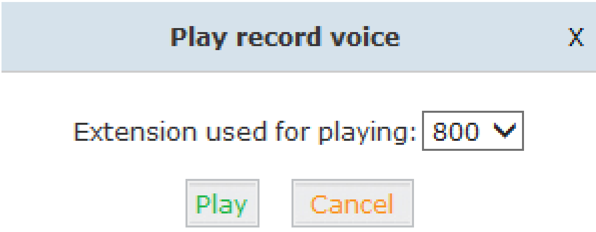


Click【IVR Prompts】----【New Voice】to create new IVR prompt:



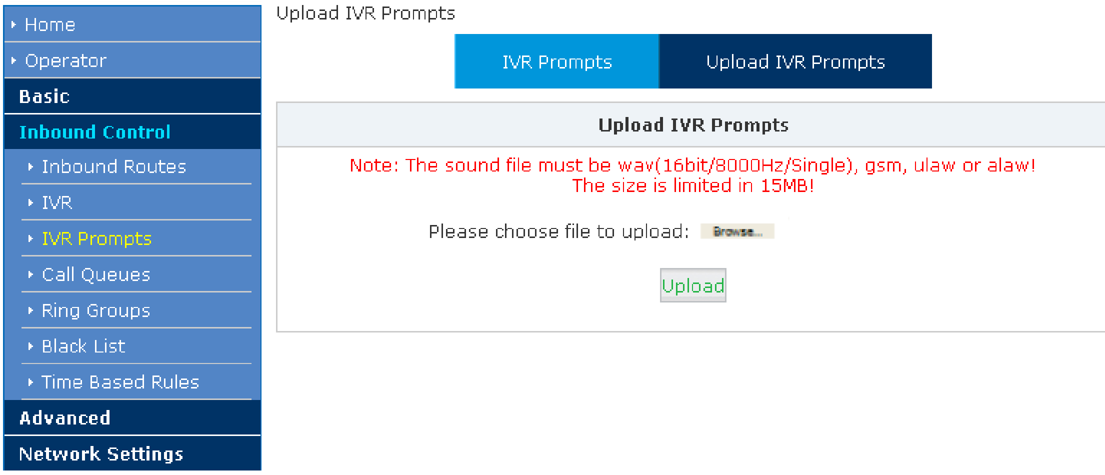
Item	Explanation
File Name	Define a name for this voice file.
Format	Select the voice format, GSM / WAV (16bit) supported only.
Extension used for recording:	Select the extension which is used for recording the IVR prompt. Click【Record】, this extension will ring, and then you can pick up the phone and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

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



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

Upload IVR prompt

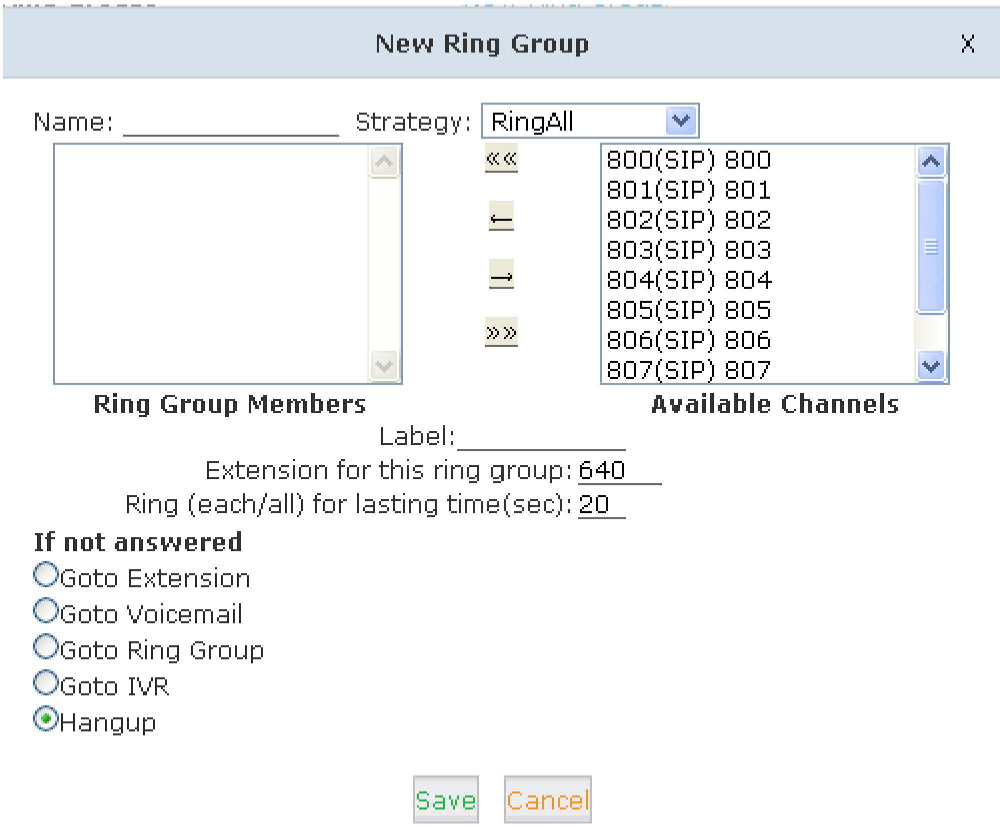


Note	Uploading customized audio file must be in the wav, gsm, ulaw, alaw format, and size must be less than 15MB.

3.4.4 Ring Groups

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

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

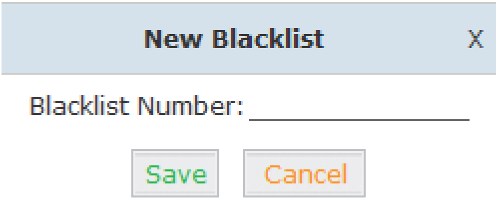


Item	Explanation
Name	Define a name for the Ring Group.
Strategy	Select “Ring All” or “Ring in order”.
Ring Group Members	Select the Ring Group Member from “the Available Channels”, click ← to add.
If not answered	You can choose to forward the call to extension, voicemail, ring group, IVR or hang up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

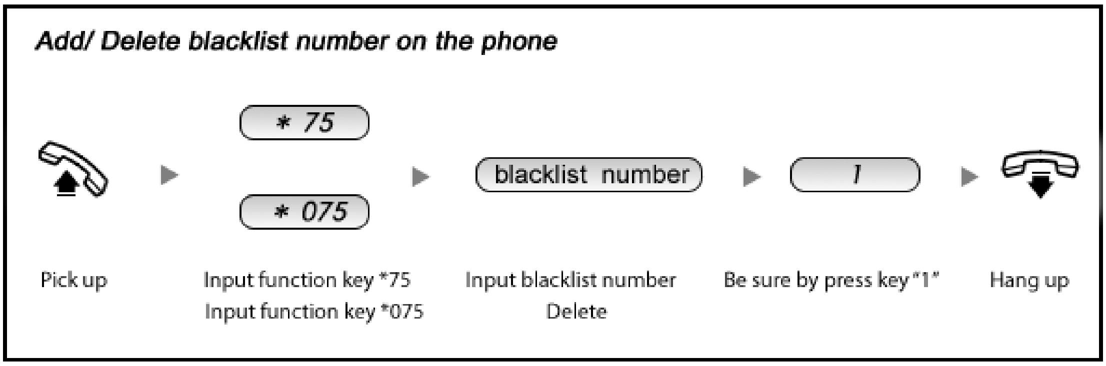
3.5 Black List

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

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



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

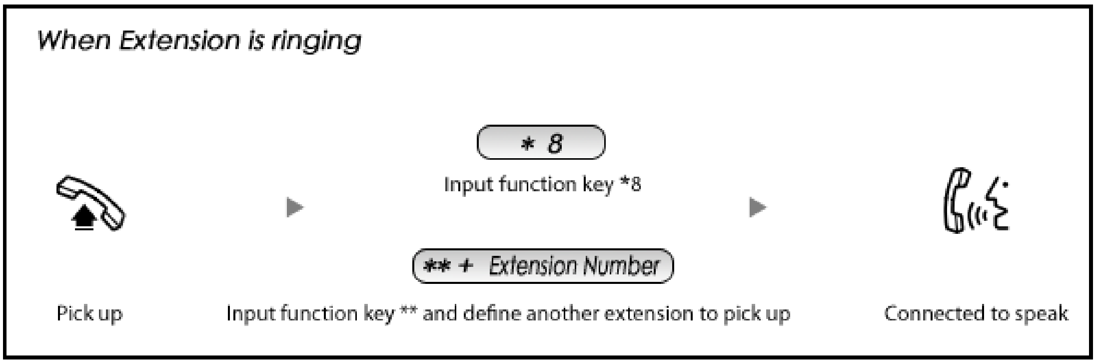


Reference Parameters and Explanation of the Blacklist:

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

3.5.1 Pick up Call

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



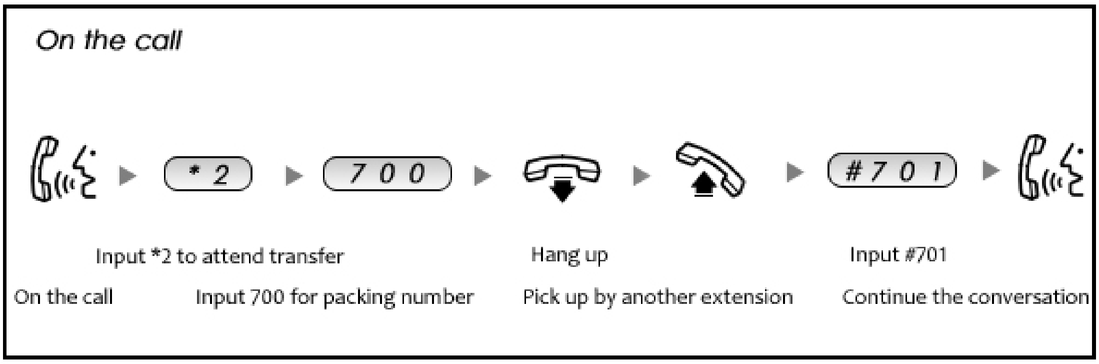
Reference Parameters and Explanation of Pickup Calls

Item	Explanation
***8**	Input function key *8 to pick up the registered extension which is in the ring at random. This can be defined in 【Feature Codes】
******	Input function key ** and define another extension to pick up. This can be defined in 【Feature Codes】.

3.6 On The Call

3.6.1 Call Parking

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

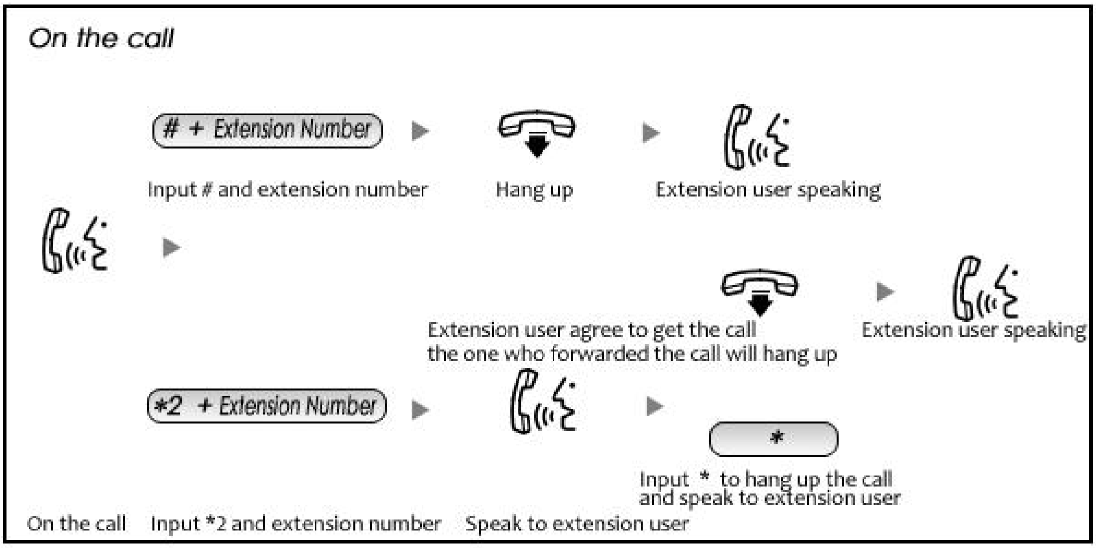


Reference Parameters and Explanation of Call Park:

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

3.6.2 Call Transfer

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

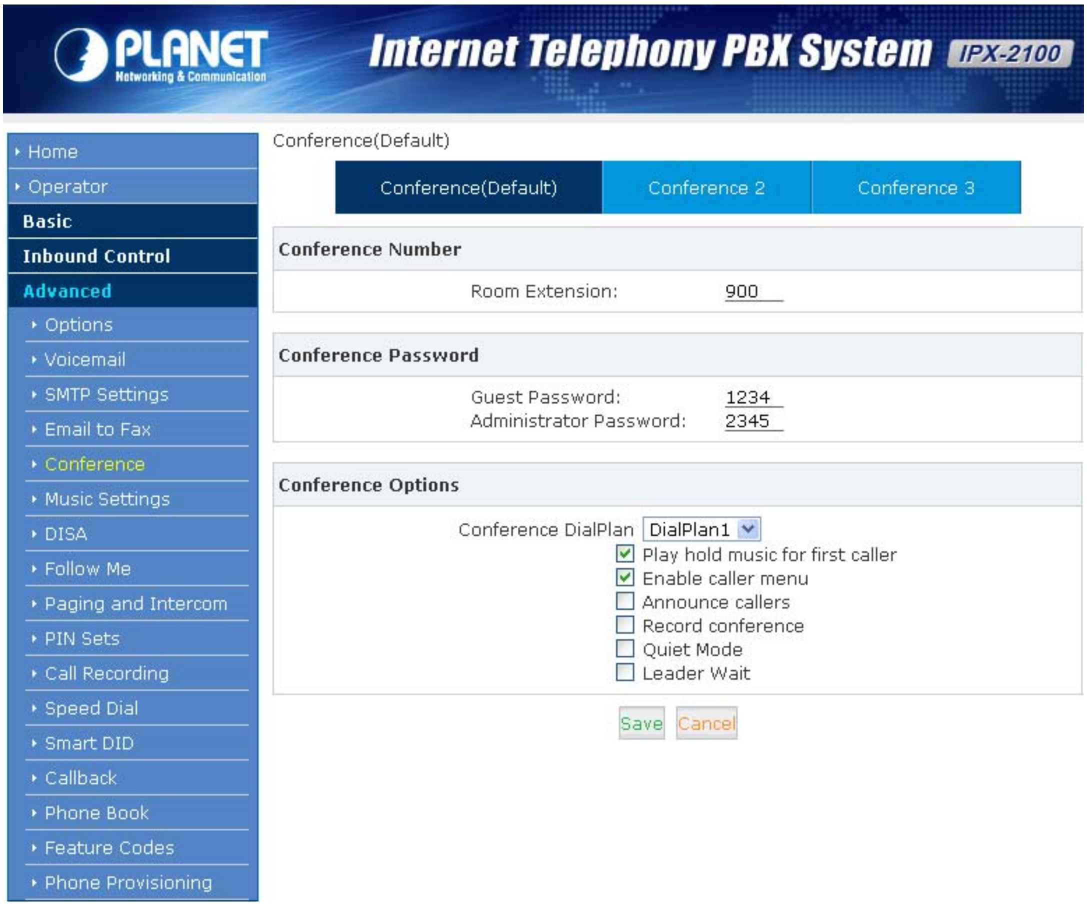


Reference Parameters and Explanation of Transfer:

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

3.6.3 Conference

If you want to create a conference room for some extension users or with external lines, you can input conference room number 900, input conference room password 1234 (Admin's password is 2345), then enter the conference room. This IPX-2100 supports 3 conference rooms. Please configure it on this page【Conference】:

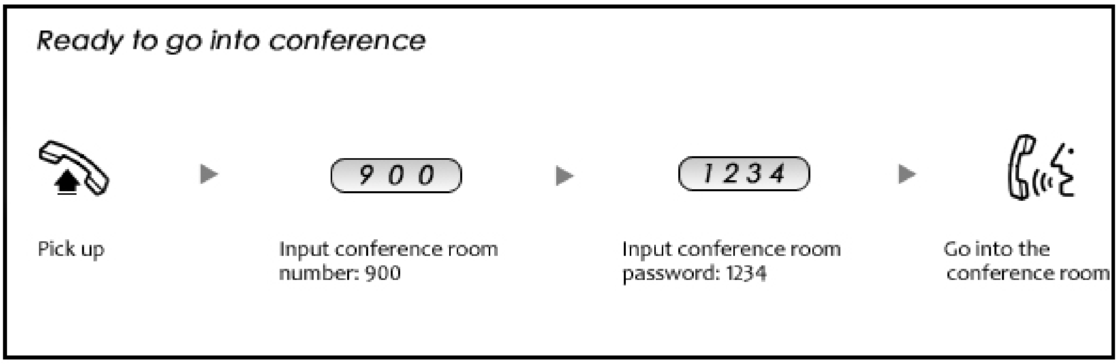


Item	Explanation
Conference Number	The number that users call in order to access the conference room; the default number is “900”.
Conference Password	Password for users to access the conference, e.g."1234".
Administrator Password	Password for administrator to access the conference.
Conference DialPlan	Use this dial plan to invite other participants.
Play hold music for the first participant	Check this option to play the hold music for the first participant in the conference until another participant enters this conference.
Enable caller menu	Check this option to allow the participant to access the Conference Bridge menu by pressing “*” on the dialpad.

Announce callers	Check this option to announce to all Bridge participants that a new participant is joining the conference.
Record conference	Recorded conference format is WAV.
Quiet Mode	If this option is checked, all the participants in the conference can hear only, but it is not allowed to speak.
Leader Wait	Wait until the conference leader (administrator) enters the conference before starting the conference.

Please check the following diagram to learn:

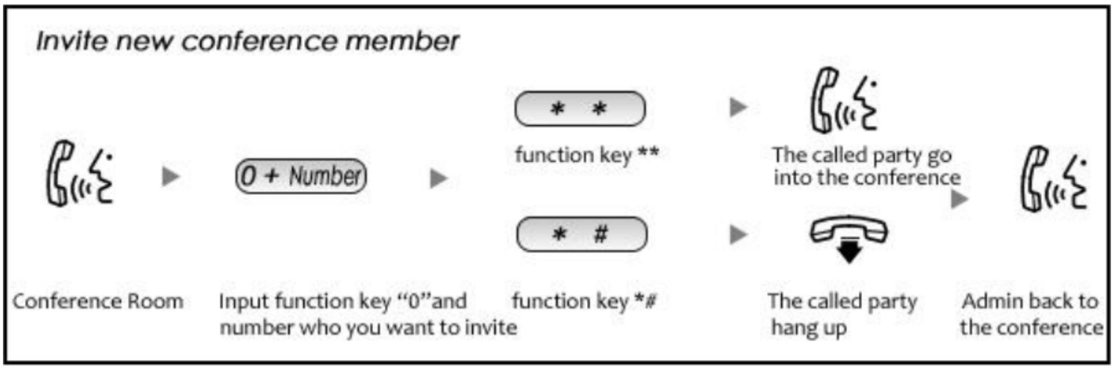
Go to conference:



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

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

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

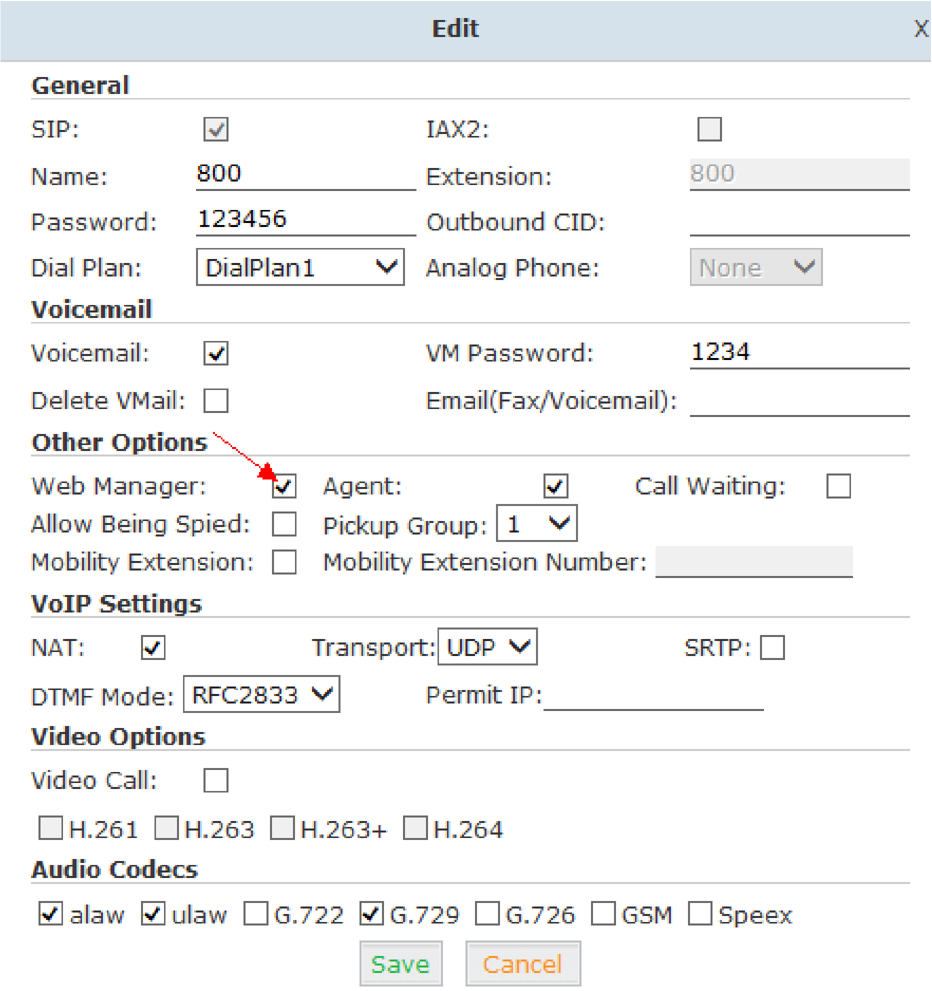


3.7 Settings before leaving office

3.7.1 Follow me

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

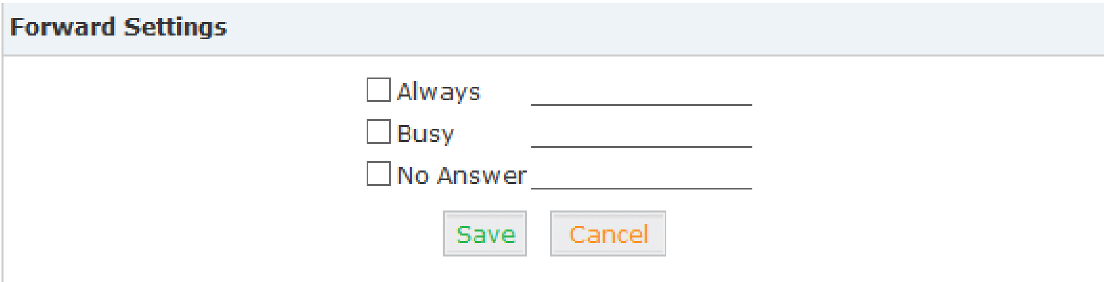
Click【Basic】->【Extension】->【Edit】the extension you want to configure.



Check【Web Manager】 and【Save】

Then login to the Extension Web Panel:

Click【Call Forward】:



Reference

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

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

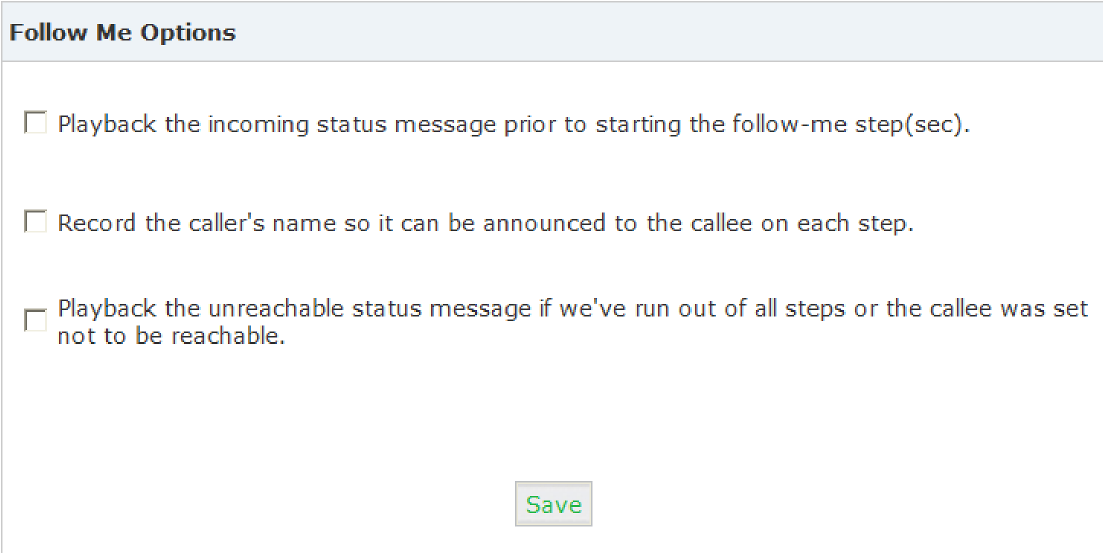
List Format: Extension Number, Ring Duration

E.g.: 806,30

808,20

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

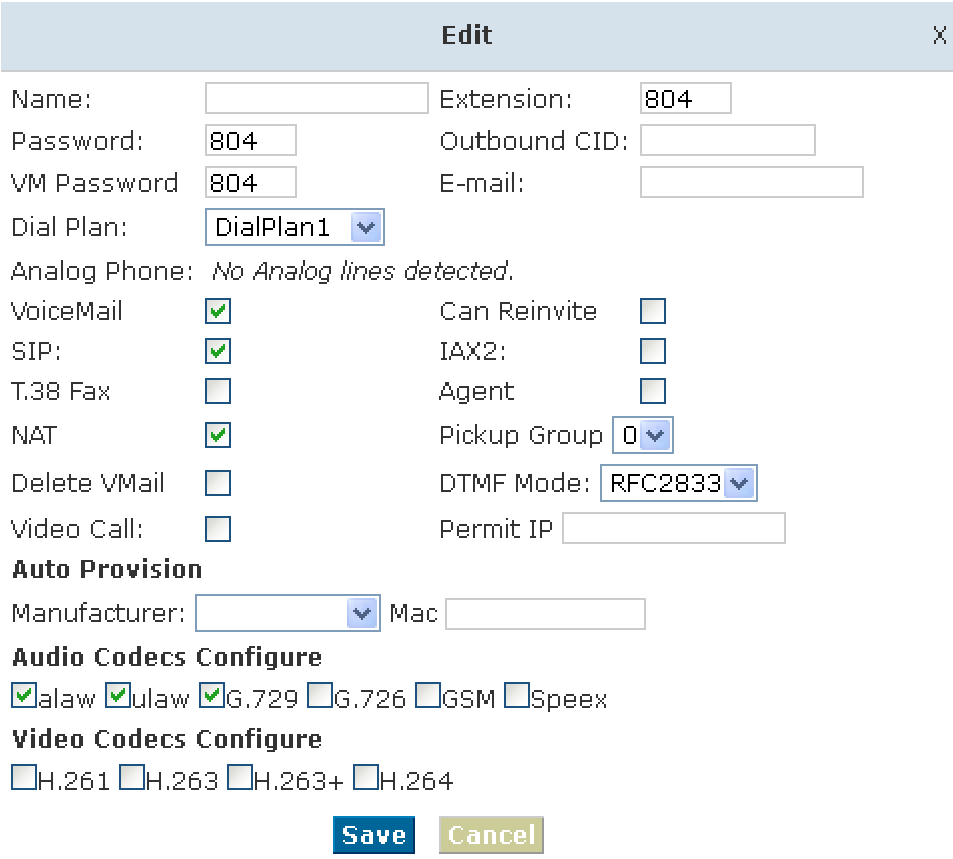
【Follow Me Option】



3.7.2 Voice Mail

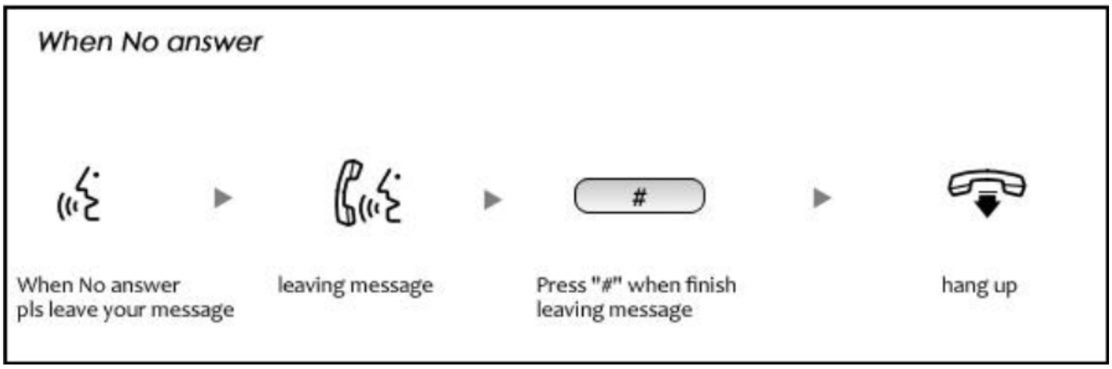
If you don't want to configure "Follow Me", you can record the message of incoming call, and email the message to your defined mailbox.

Click【Extension】---【Extension Settings】

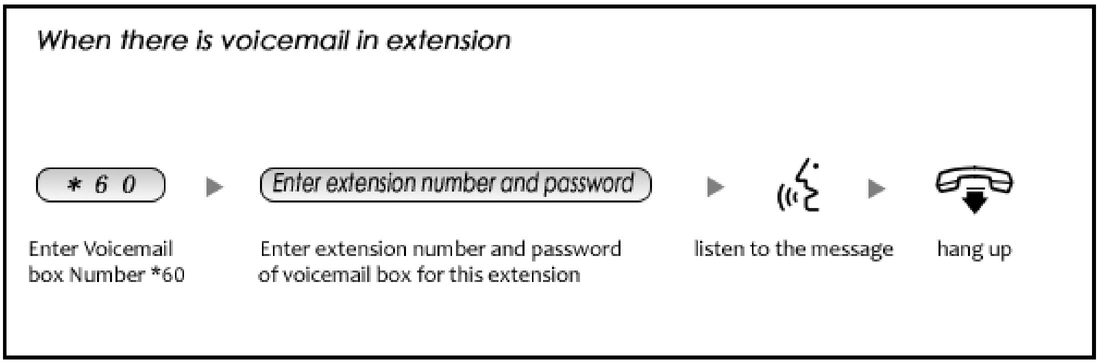


Please enable【Voice mail】before configuration, and configure【VM Password】and【Email】. If there is no answer for the incoming call and when the default ring time is over, the system will play: “Please leave your message and press the “#” key. Then voicemail will be sent to the specified mailbox by email.

Leave a message:



Listen to the message



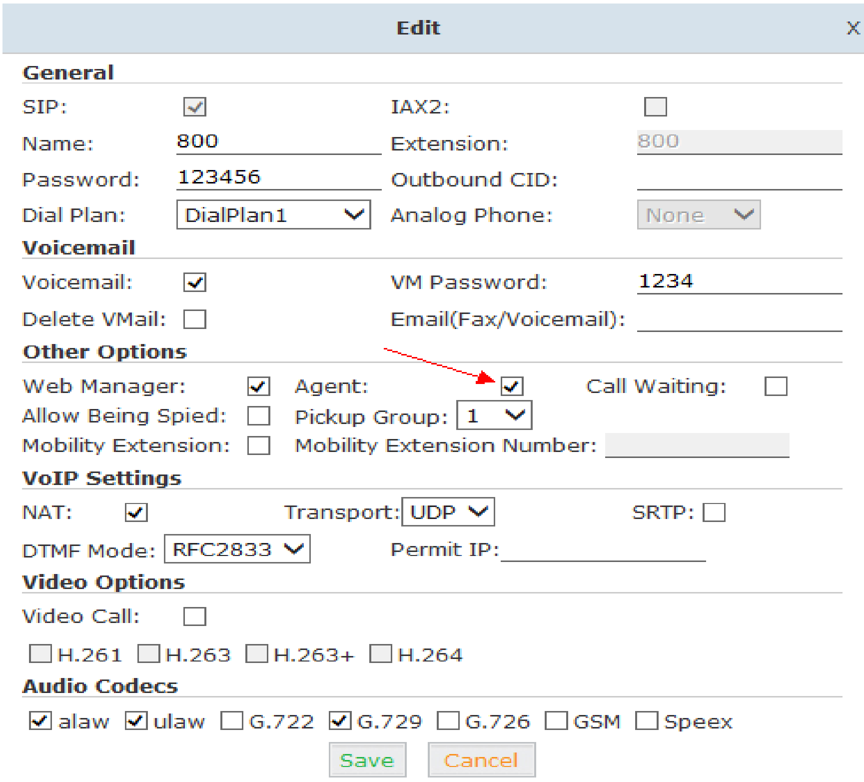
Note

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

3.8 Call Center(Call Queues)

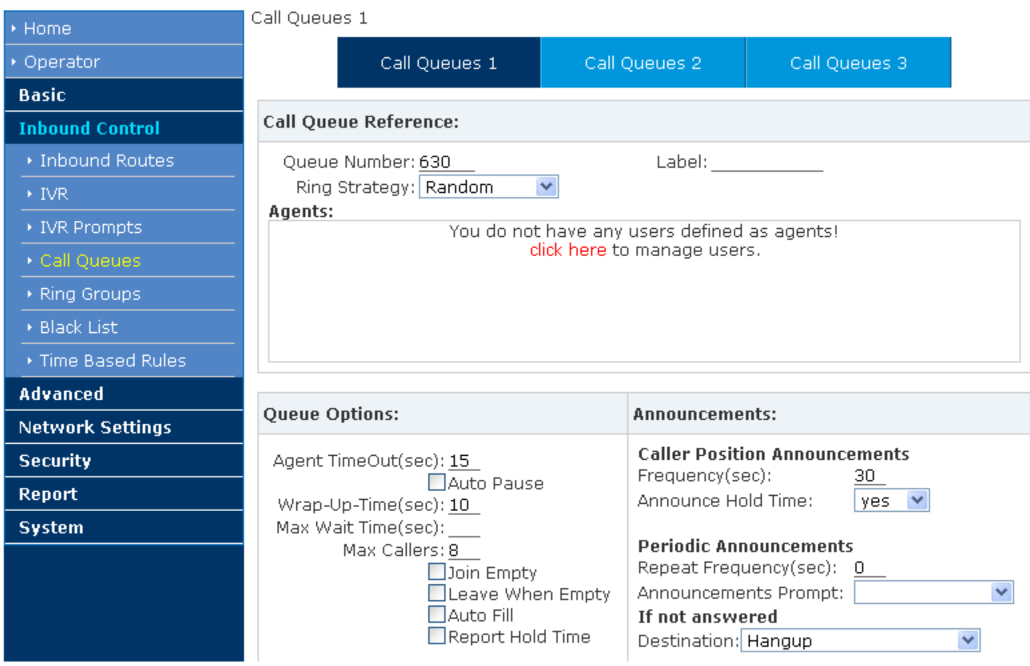
3.8.1 Create Agent

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

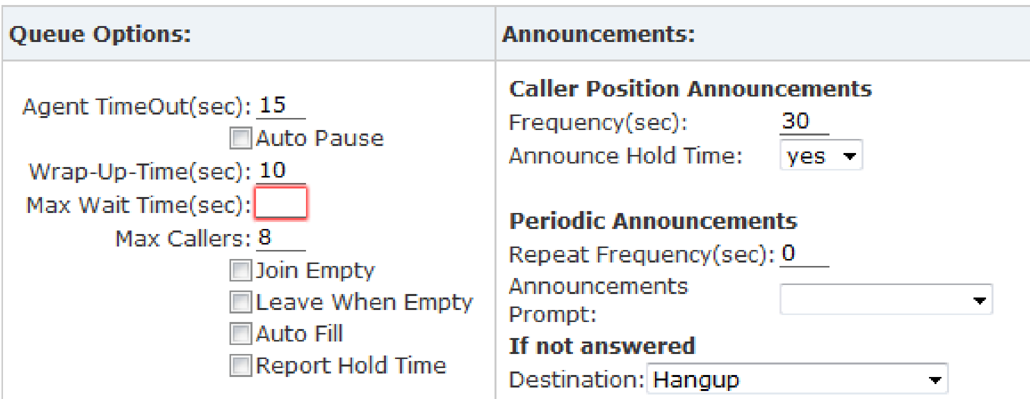


Step1: Check【Agent】and【Save】

Step2: Click【Inbound Control】->【Call Queues】



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



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

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

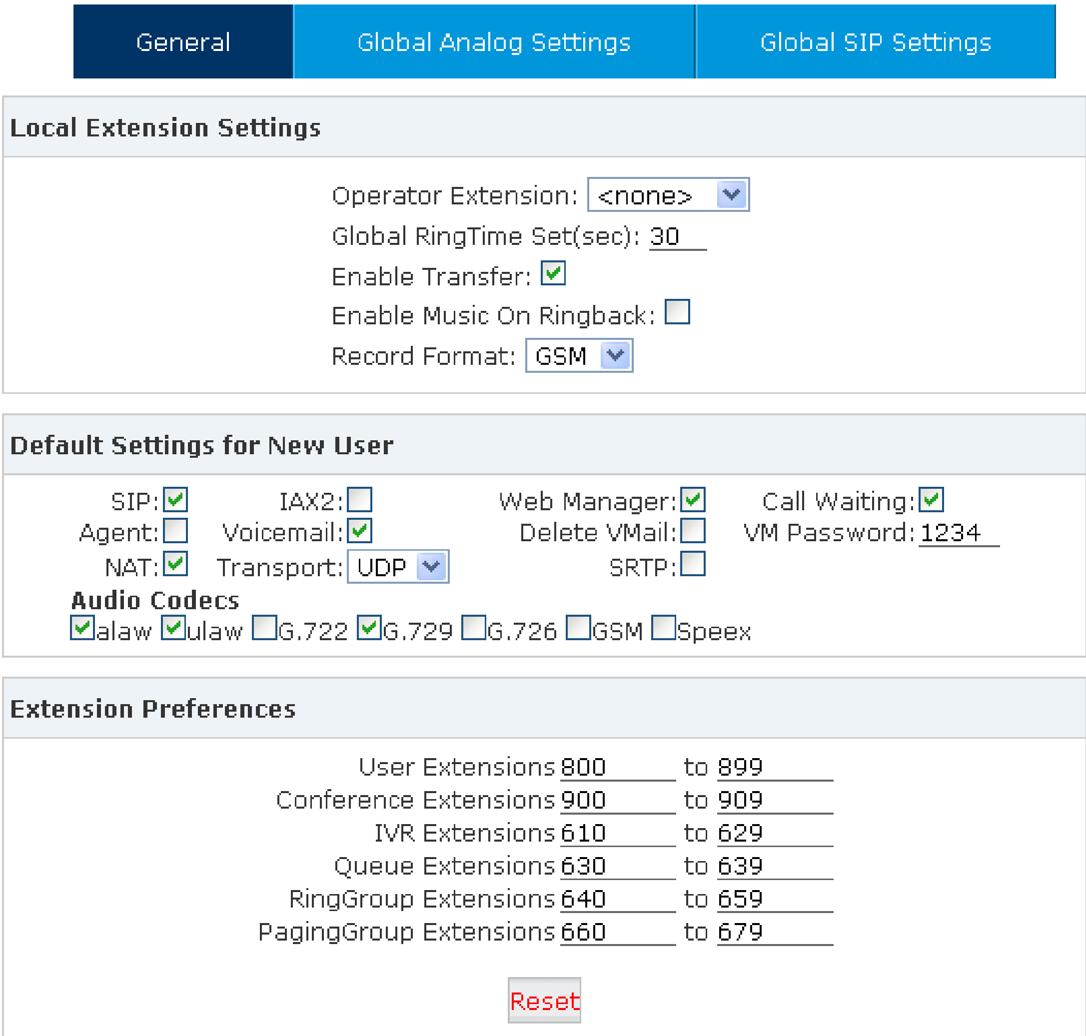
Chapter 4 Advanced

4.1 Options

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

4.1.1 General

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

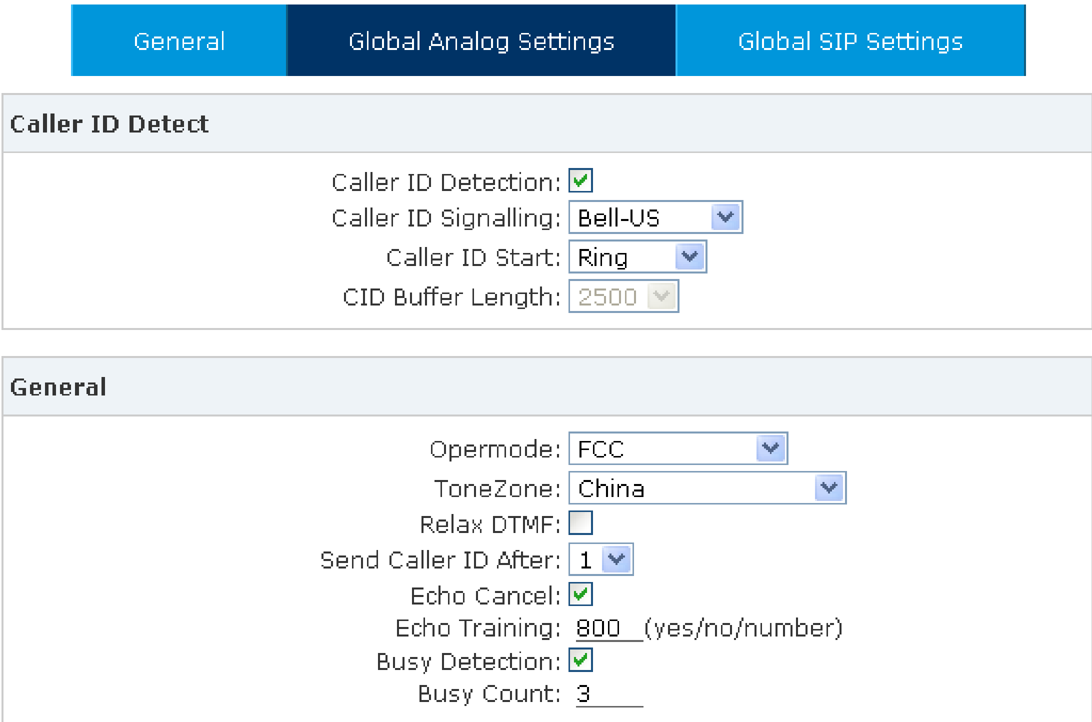


Item	Explanation
Operator Extension	Set extension number for Operator.
Global Ring Time Set	Set Ring time for every extension.
Enable Transfer	Check to enable Transfer.

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

4.1.2 Global Analog Settings

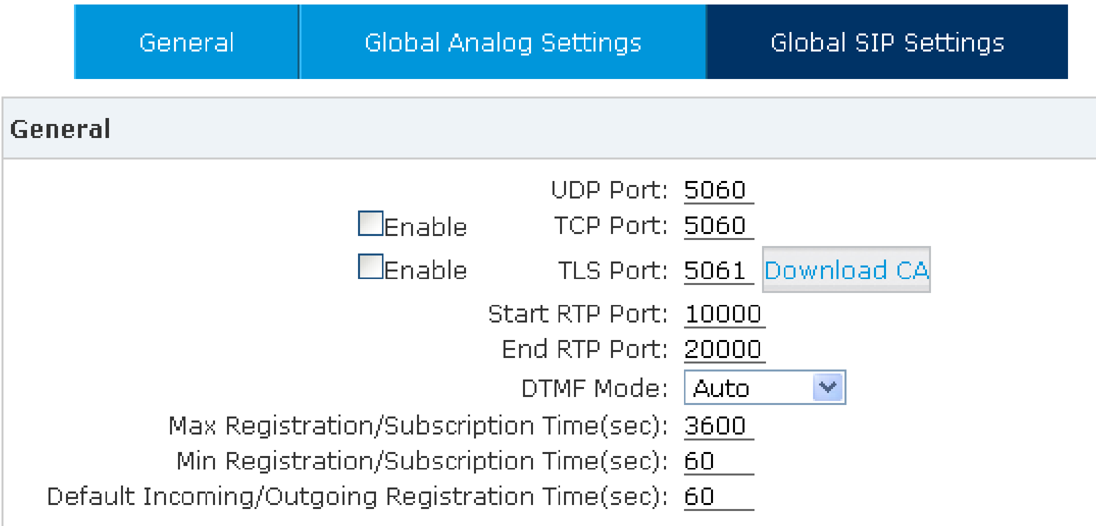
Click【Advance】->【Options】->【Global Analog Settings】:



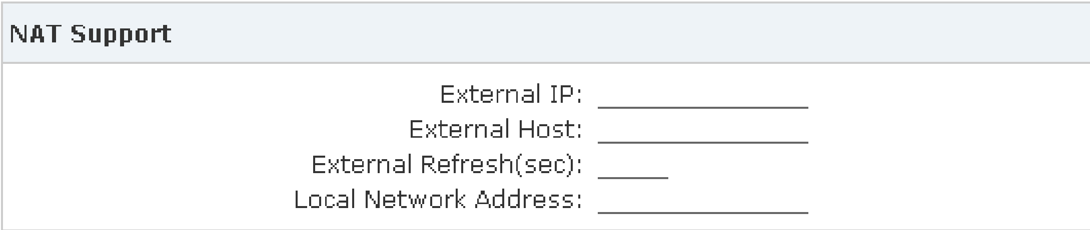
Item	Explanation
Caller ID Detection	Enable/Disable Caller ID Detection
Caller ID Signaling	Select the mode of Caller ID Signaling.
Caller ID Start	Ring--Caller ID start before ring. Polarity--Caller ID start when polarity reversal starts.
CID Buffer Length	Default CID Buffer Length
Opermode	Set the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

4.1.3 Global SIP Settings

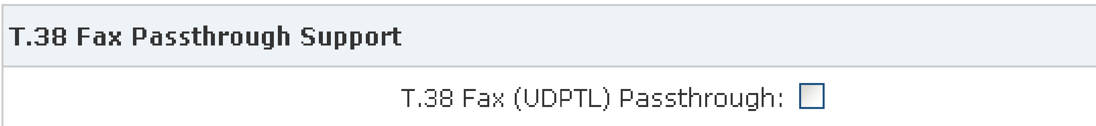
【Global SIP Settings】is appropriate for professionals. If anything needs to be modified, please contact our tech-support people.



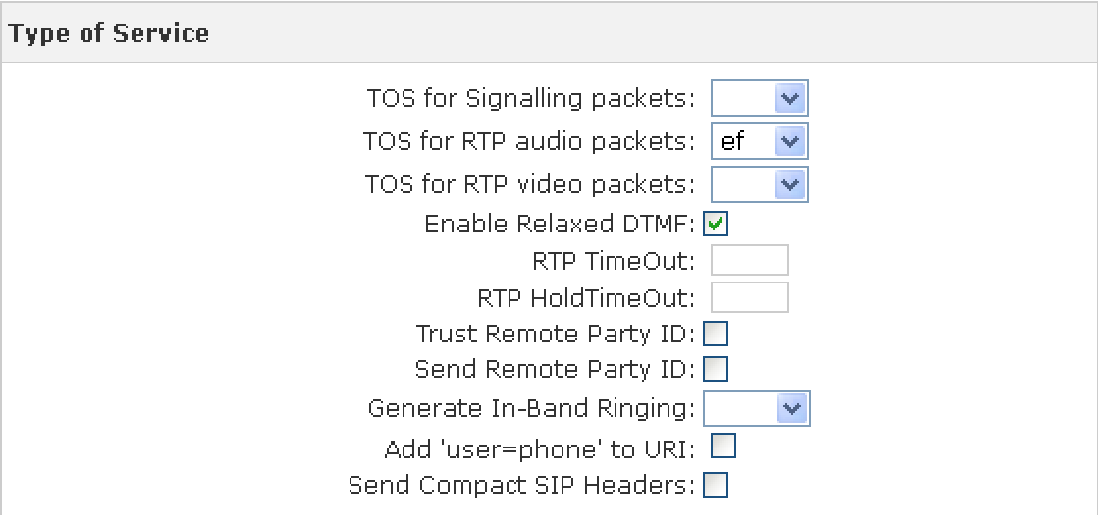
Item	Explanation
UDP Port to bind to	SIP standard port is 5060
TCP Port	Default TCP port is 5060
TLS Port	Default TLS port is 5061
Start RTP Port	RTP port range
End RTP Port	RTP port range
DTMF Mode	Set default DTMF mode for sending DTMF, support auto, RFC2833, inband, info. Default: RFC 2833
Max Registration/Subscription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



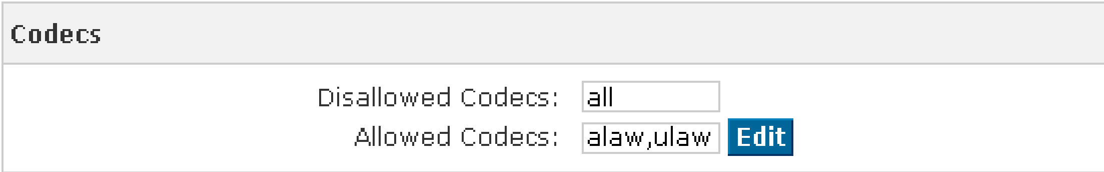
Item	Explanation
External IP	Address that we're going to put in outbound SIP messages if we're behind a NAT
External Host	Alternatively, you can specify an external host, and Asterisk will perform DNS queries periodically. Not recommended for production environments! Use external IP instead
External 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



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



Item	Explanation
TOS for Signaling packets	Sets Type of Service for SIP packets
TOS for RTP audio packets	Sets Type of Service for RTP audio packets
TOS for RTP video packets	Sets Type of Service for RTP video packets
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

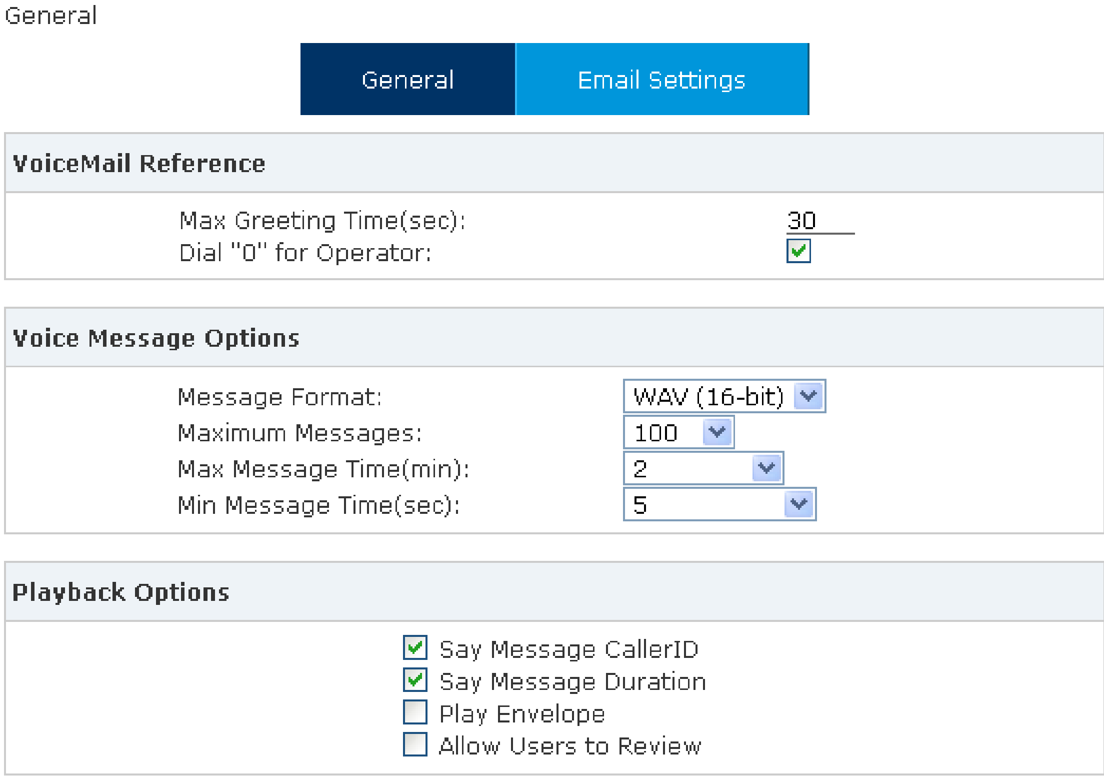


Item	Explanation
Register Time Out	Retry registration calls at every 'x' seconds (default 20)
Register Attempts	Number of registration attempts before we give up; 0 = continue forever
Disallowed Codecs	Default is disallowed = all
Allowed Codecs	Choose the codec that system allows

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

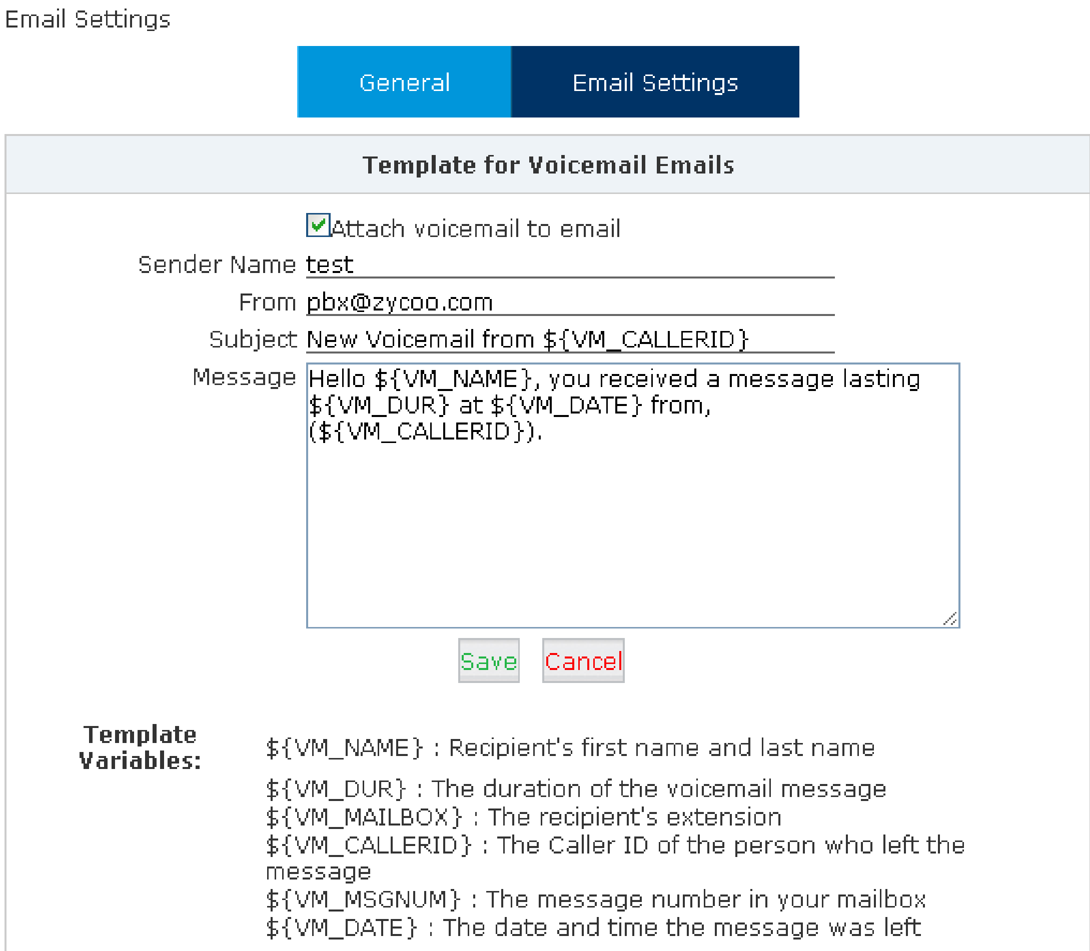
4.2 VoiceMail

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



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

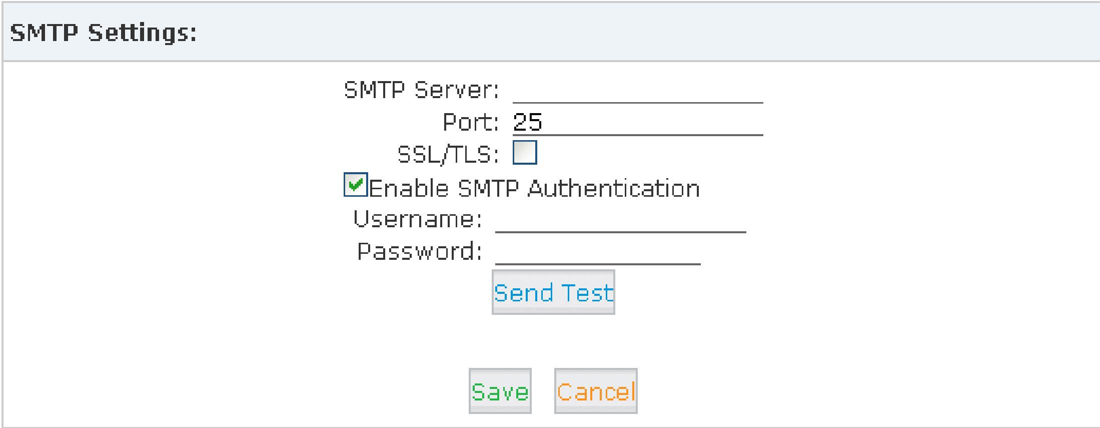
Email Settings



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

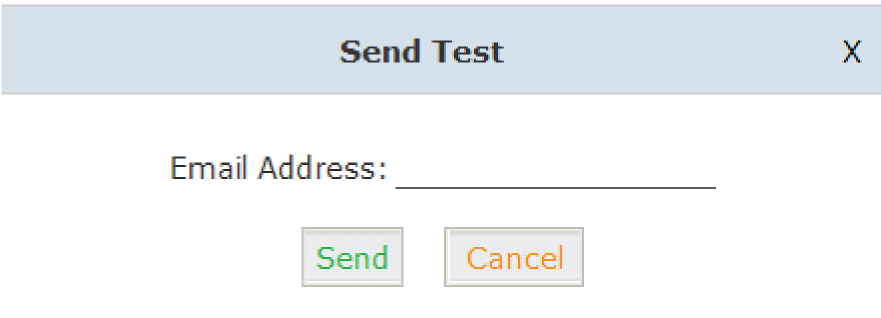
4.3 SMTP Setting

SMTP Settings



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

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

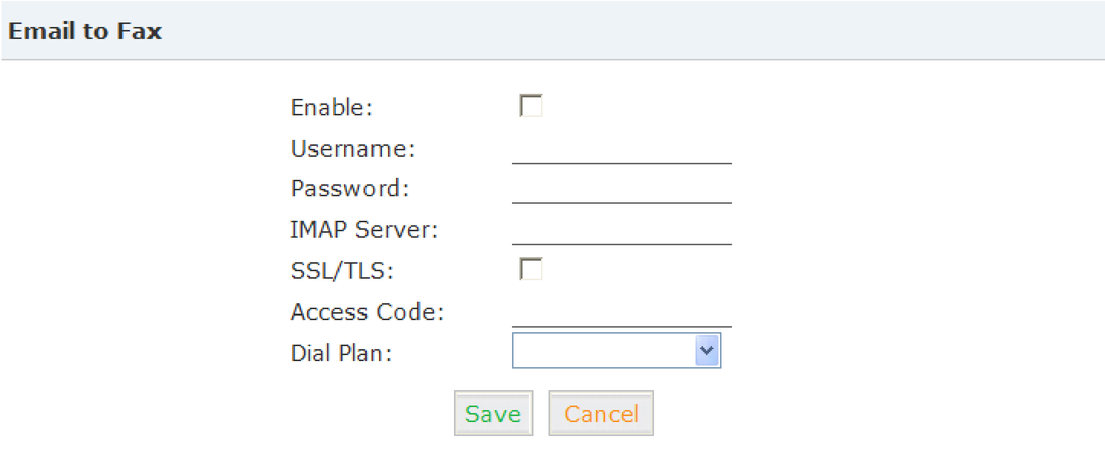


Input the Email and click【Send】to send the test email. Login your Email to check; configuration is successful if you receive the test email; otherwise, it fails. Please check your email settings.

4.4 Email to Fax

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

Click【Advanced】->【Email to Fax】



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

Practical Case:

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

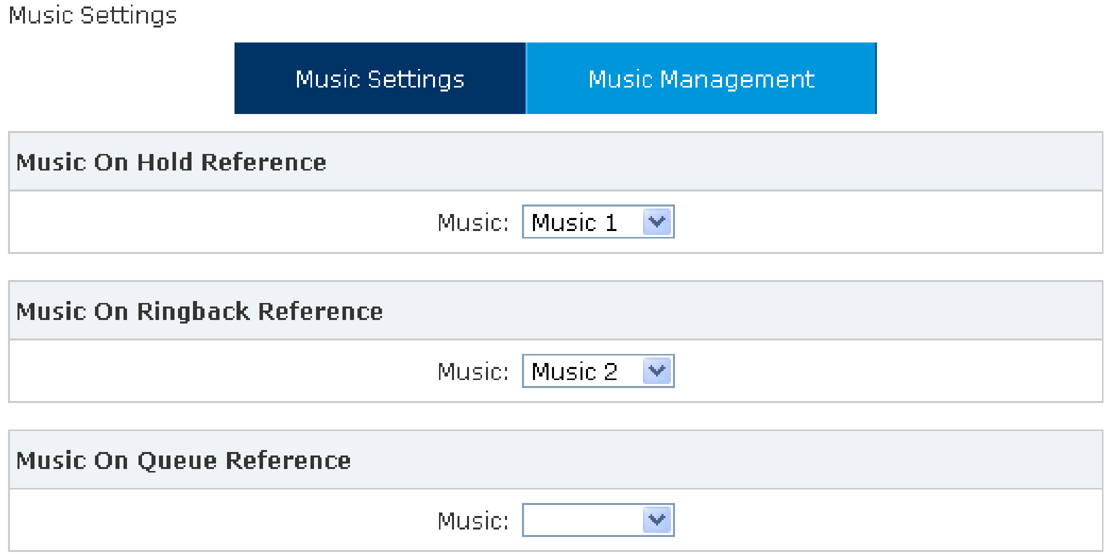
If you need to dial the extension when sending fax, e.g. fax number: 85337096 ext.800, you need to use the 【Access Code】: 985337096-800 as subject.

4.5 Music Settings

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

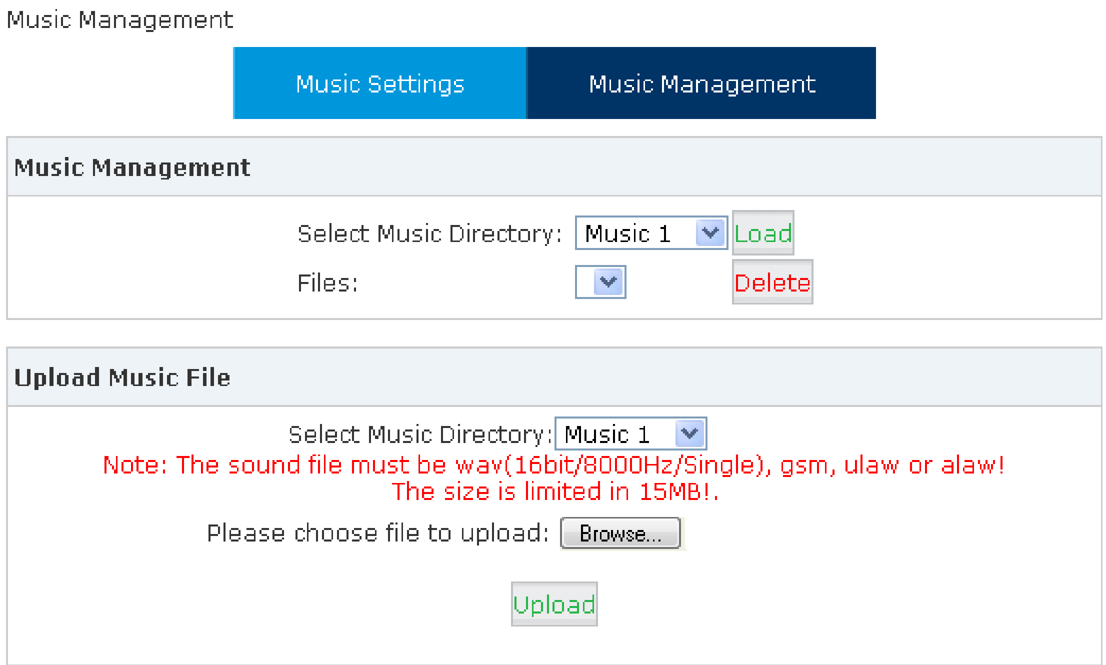
Click【Music Settings】 to display the dialog as below:

Music Settings:



Please define different music files for different music folders.

Music Management:



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

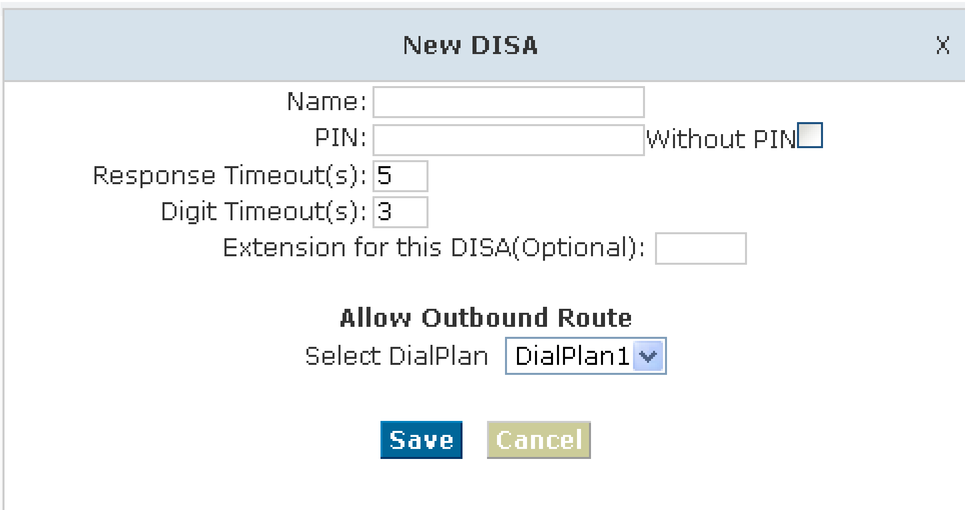
	15MB.

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

4.6 DISA

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

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

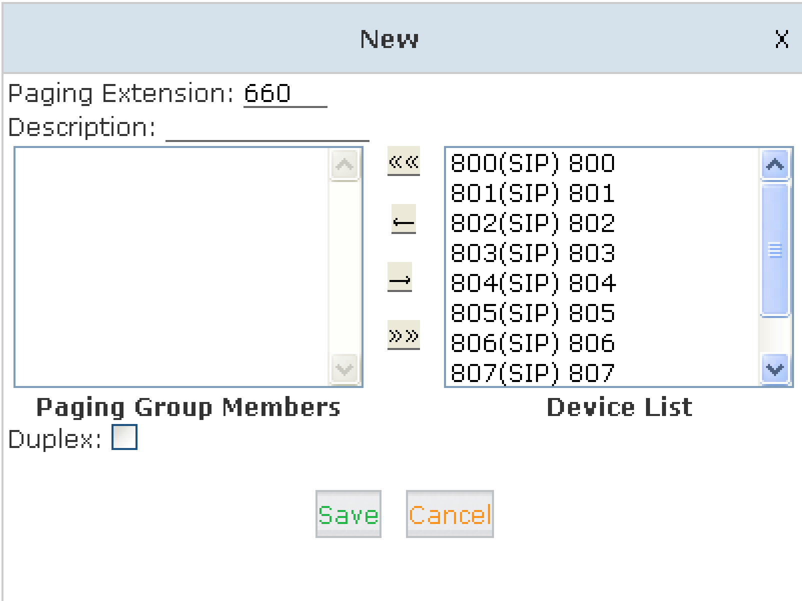


Item	Explanation
Name	Define a name for DISA.
PIN Set	User will be prompted to input this number when PIN Authentication is needed.
Record in CDR	Check to record.
Response Timeout(sec)	The maximum time for waiting before hanging up if the dialed number is incomplete or invalid. Default is 10 seconds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

4.7 Paging And Intercom

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

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



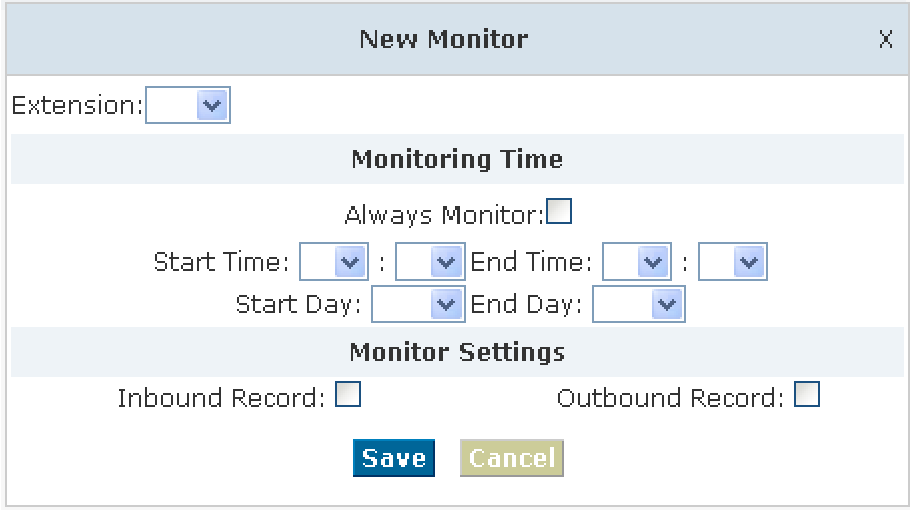
Item	Explanation
Paging Extension	The number users will dial to page this group.
Description	Provide a descriptive title for this Page Group.
Paging Group Members	Selected device(s) on this page
Device List	Select Device(s) to page.
Duplex	Paging is typically one way for announcements only. Checking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

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

4.8 PIN Set

Monitor is used for recording the defined extensions.

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

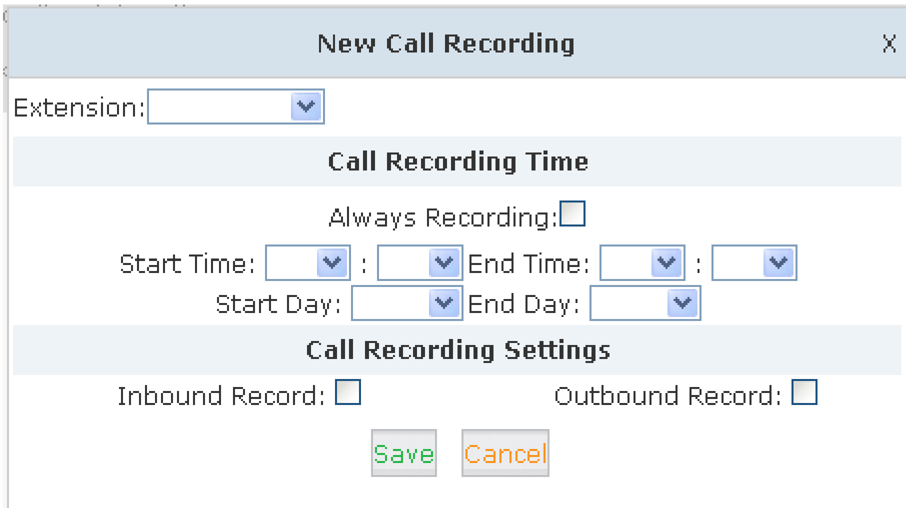


Item	Explanation
PIN Set Name	Define the name for this PIN Set.
PIN List	Define PIN codes in this list.

4.9 Call Recording

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

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



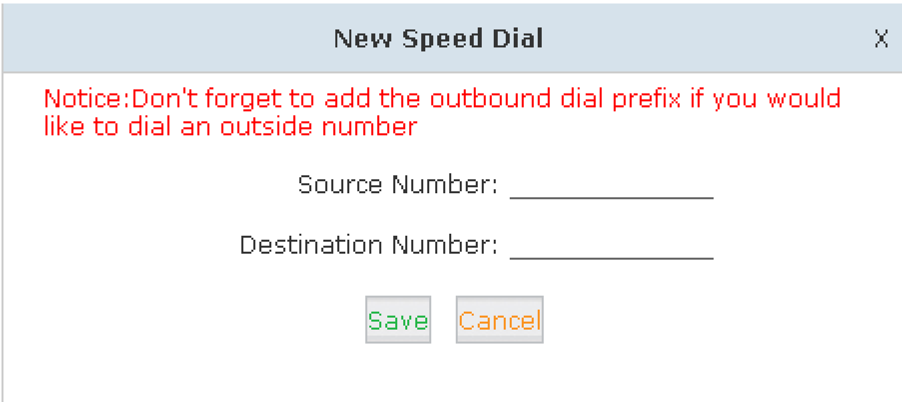
Reference:

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

4.10 Speed Dial

Please configure as shown below:

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



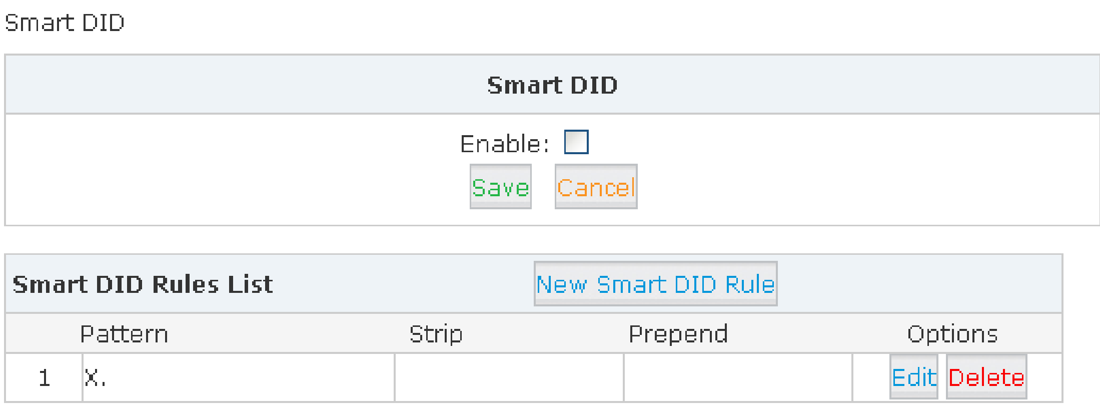
E.g. prefix is *99, speed number is 00, destination telephone number is 85337096.

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

4.11 Smart DID

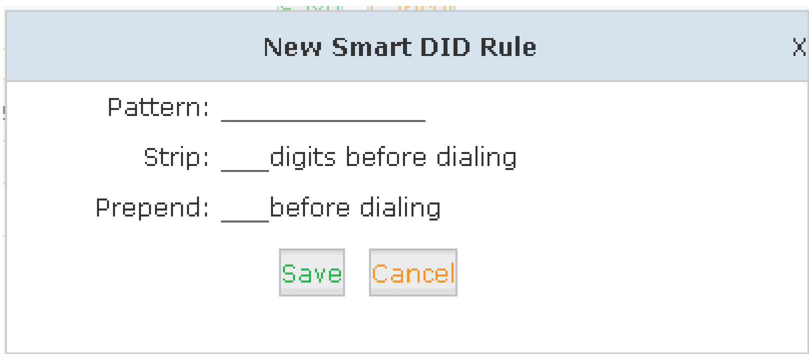
Smart DID: After extension user makes an outbound call, the call is ringing back to Planet IP PBX, and directed to the one who made the last call. Please configure it as shown below:

Click【Advanced】->【Smart DID】:



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

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

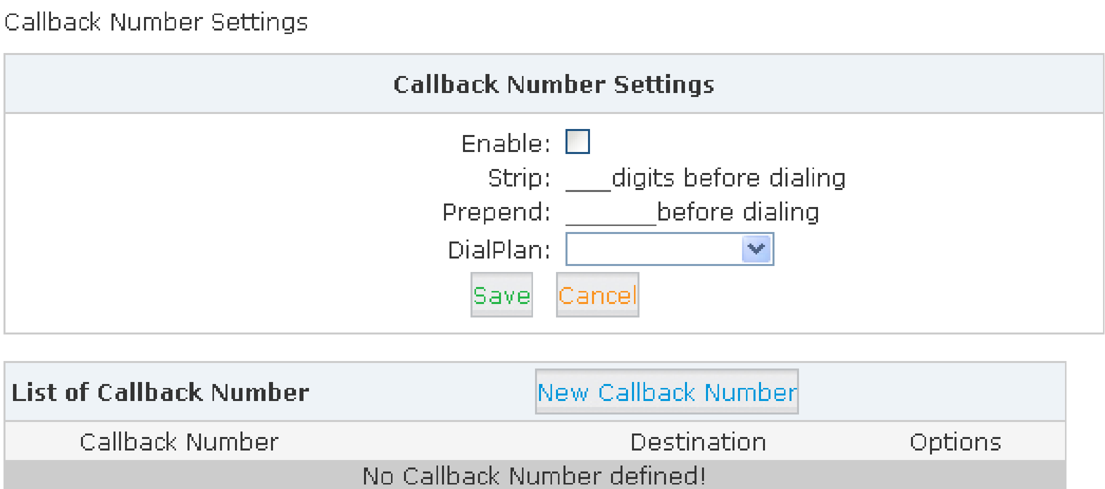


Input the pattern and define how many digits need to be striped or prepend, and then click “Save”--“Activate”.

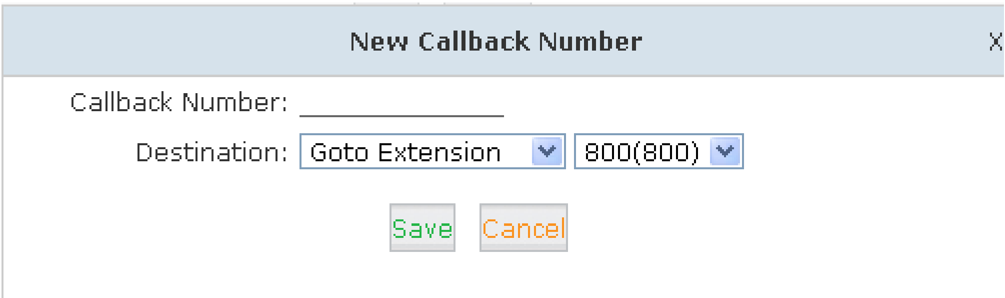
4.12 Call Back

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

Click【Advanced】->【Callback】:



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

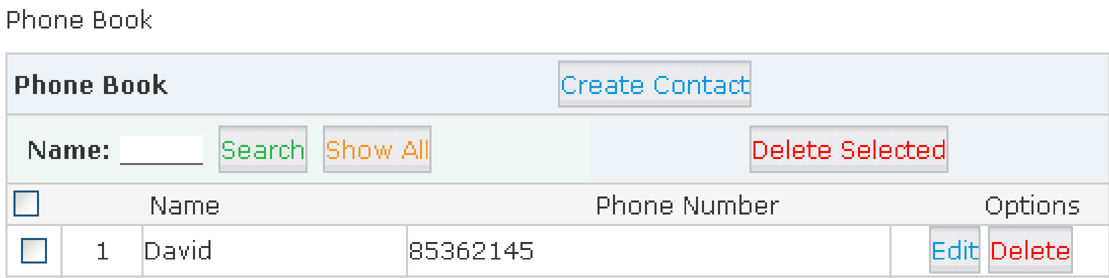


Input callback number and define the destination.

4.13 Phone Book

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

Click【Advanced】->【Phone Book】:



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

Click【Create Contact】to see the following diagram:

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

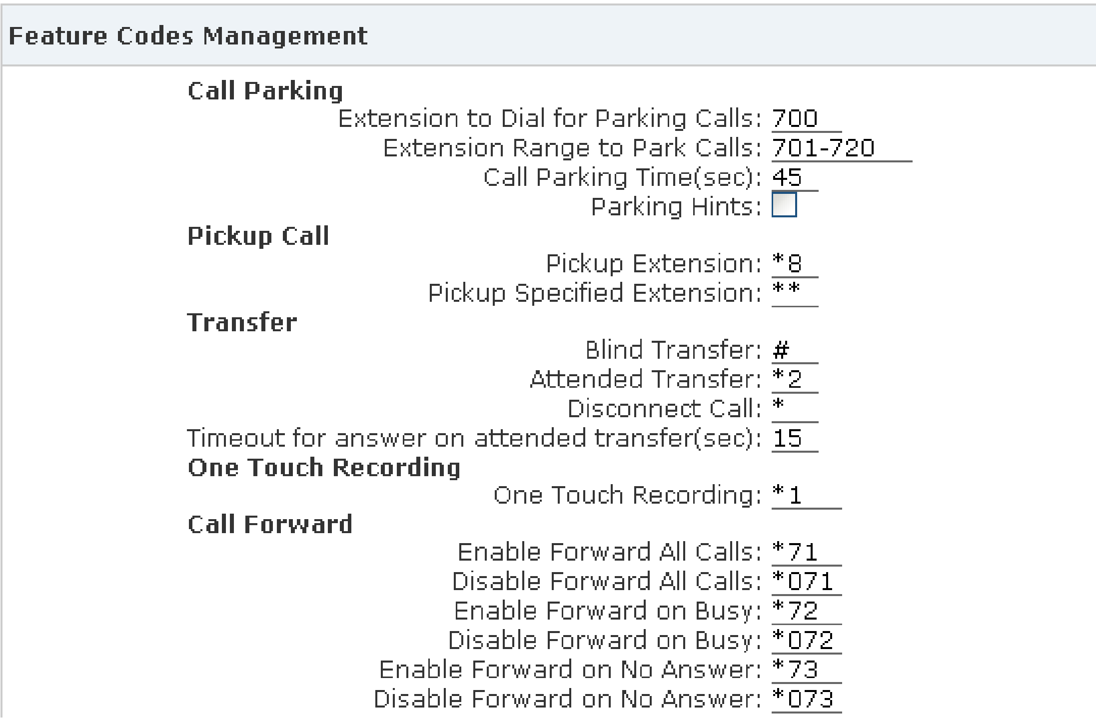
Phone book is for the incoming call to use; if the incoming caller ID matches the number in Phone book, it will display the name defined in Phone book.

For example, Name: David Number: 123456789.

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

4.14 Feature Codes

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



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

Blind Transfer	Allow unattended or blind transfers. It works like this: While on a conversation with A, you dial the blind transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number (B's number) and A is put through to B immediately. Your line is off. The caller ID displayed to B is exactly the same as the caller ID presented to you.
Attended Transfer	Allow attended transfer or supervised transfer. It works like this: While on conversation with A, you dial the Attended Transfer key sequence. The system says "Transfer" then gives you a dial tone, while A is on hold. You dial the transferee number (B's number) and talk with B to introduce the call, then you can hang up and A will be connected with B. In case B does not want to answer the call, he/she simply hangs up and you will be back to your original conversation.
Disconnect Call	Disconnect the current transfer call (for Attended transfer).
Timeout for answer on attended transfer（sec）	Set the timeout value
One Touch Recording	Configure the function key for One Touch Recording
Call Forward	Enable/Disable Call Forward and the settings of function keys for different forward modes.
Do Not Disturb	Enable/Disable “Do Not Disturb”
Spy	Configure the function keys for spy modes.
Blacklist	Add/Delete blacklisted number.
Voicemail	Configure the function keys for entering voicemail and check extension voicemail.
Invite Participant	In conference, the administrator can invite people into the conference by dialing “0”. After pressing “0”, you will get dial tone, and you can dial to invite people. After the call is connected, please press ** to direct the people into the conference, or *# to hang up the current call and return to the conference.
Create Conference	During the call, you can dial *0 to forward to the conference with the callee.
Return to conference with participant	In conference, the administrator can dial “0” to invite people into the conference. After pressing “0”, you will get dial tone, and you can dial to invite the participant; when the call is connected, dial “**” to return to the conference with invited participant.
Return to conference without participant	In conference, the administrator can dial “0” to invite people into the conference. After pressing “0”, you will get dial tone, and you can dial

	to invite the participant. When the call is connected, you can dial "*#" to hang up and return the conference yourself.
Pause Queue Member Extension	Pause the agent, and the agent cannot receive the call.
Unpause Queue Member Extension	Unpause the agent, and the agent can receive the call.
Others	Function key for Intercom/ Paging/ Directory

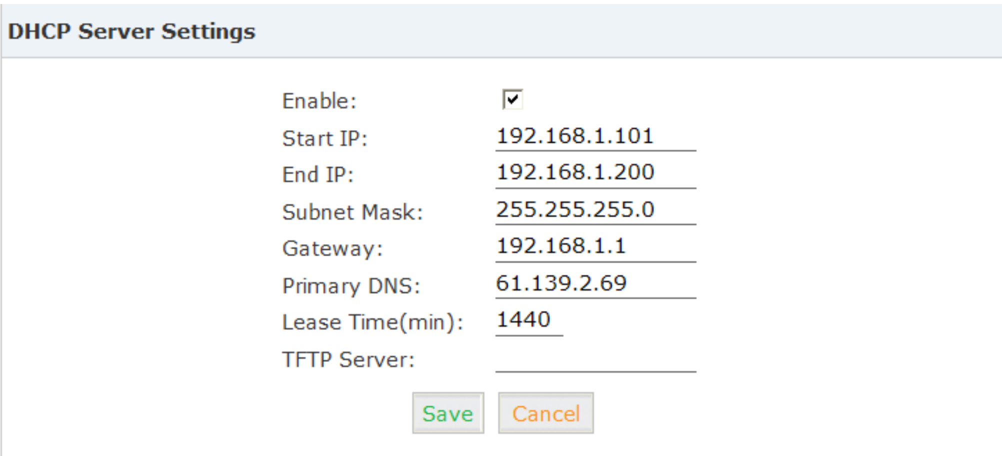
4.15 IP Phone Provision

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

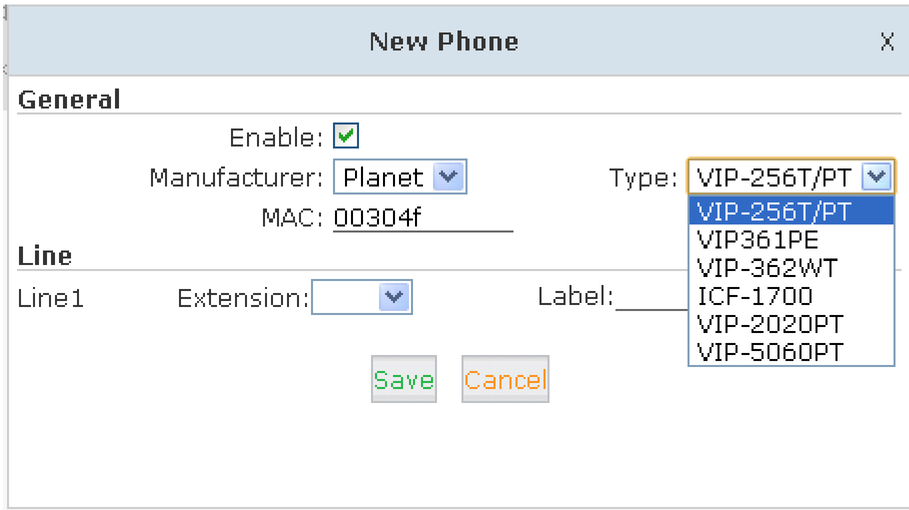
There are two operation methods to fulfill this function. Please see details as shown below:

Enable DHCP service

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



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



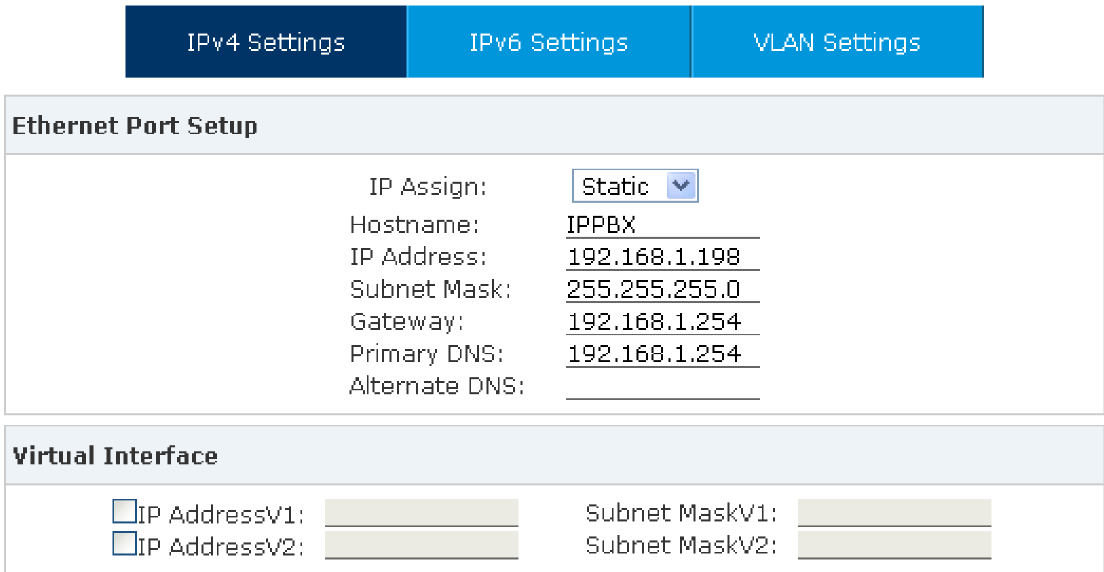
Enable Phone Provisioning in【Basic】, select the IP Phone manufacture, input MAC of the phone, and select the extension for provisioning.

Chapter 5 Network Settings

5.1 Network

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

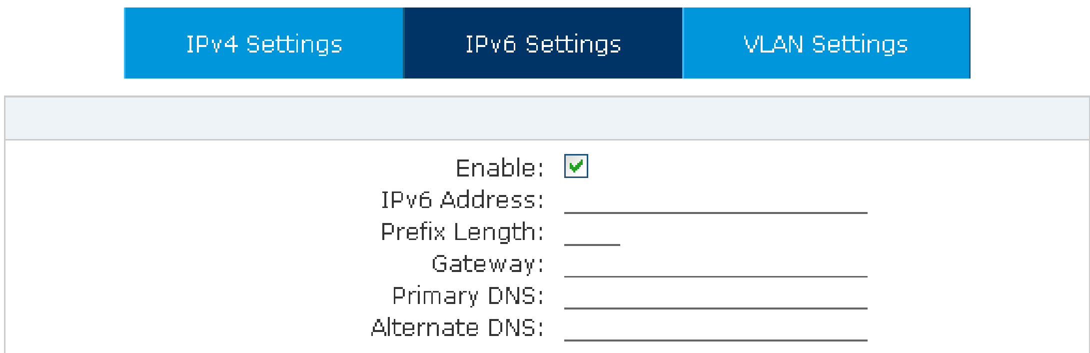
Click【Network Settings】->【Network】->【IPv4 Settings】



Reference

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

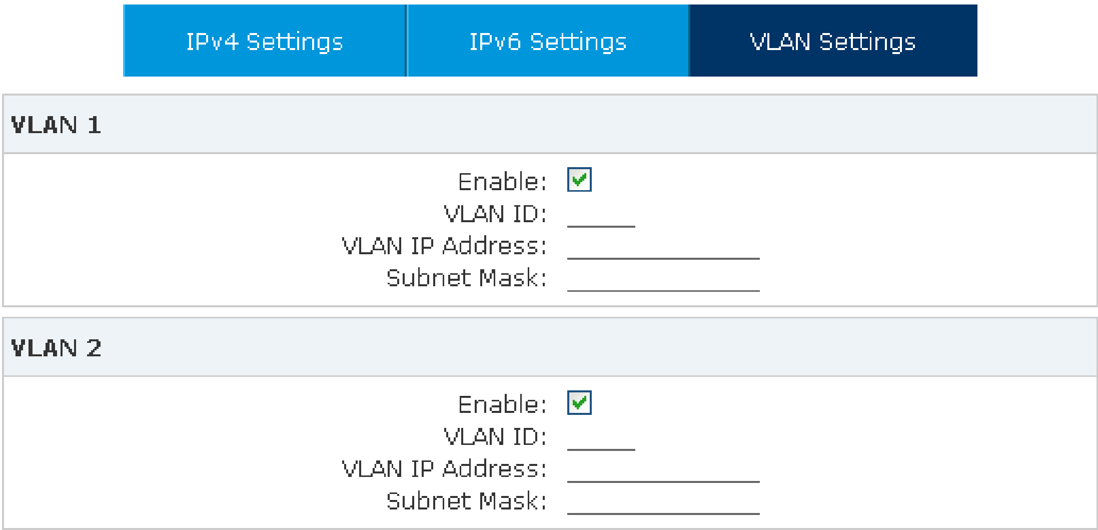
Click【Network Settings】->【Network】->【IPv6 Settings】



IPv6 Reference:

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

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

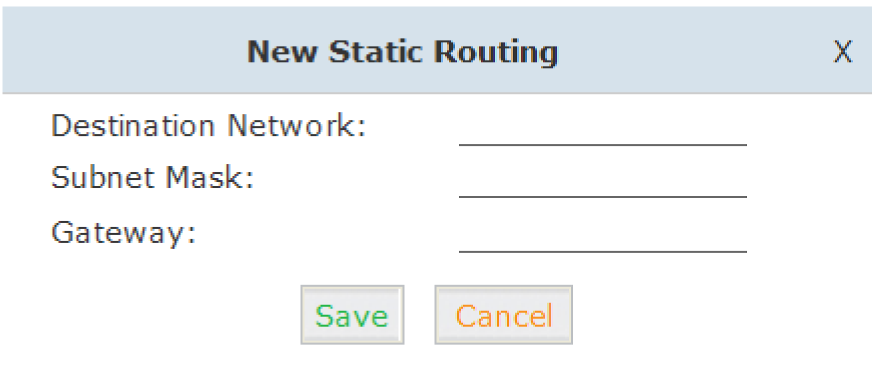


VLAN Reference:

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

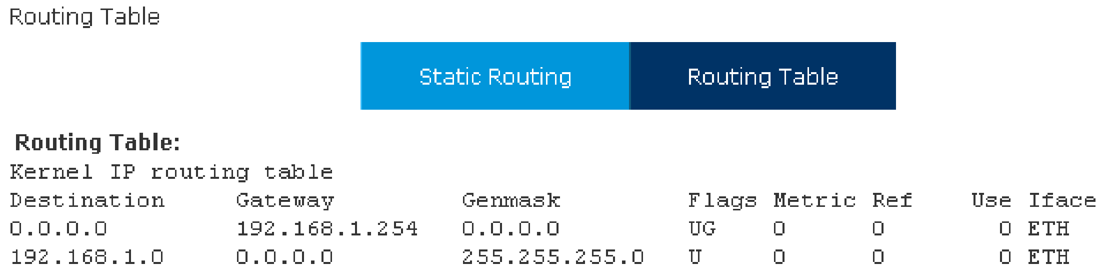
5.2 Static Routing

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



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

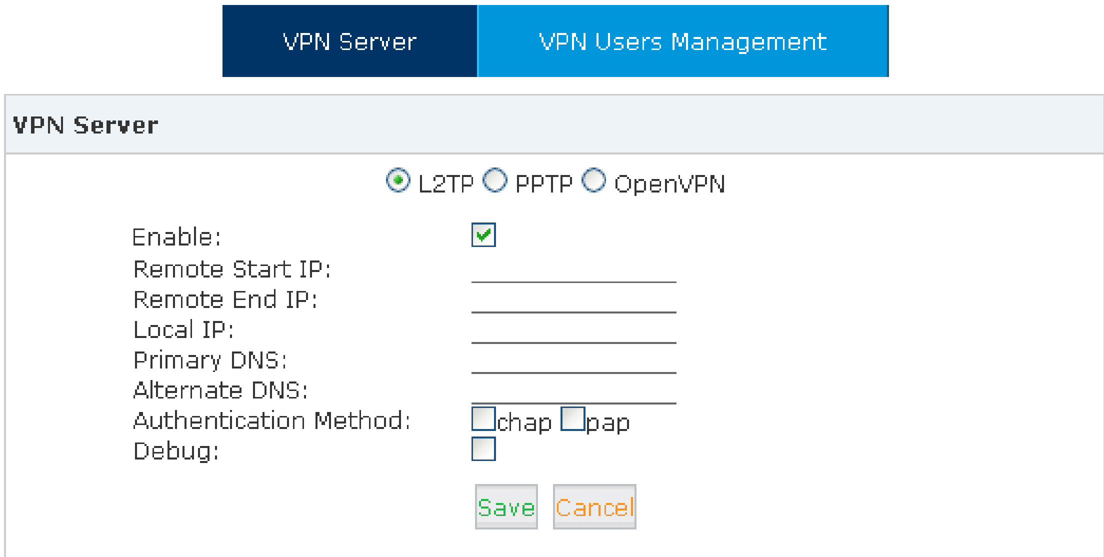
Click【Network Settings】->【Static Routing】->【Routing Table】, and the current routing information will be displayed below:



5.3 VPN Server

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

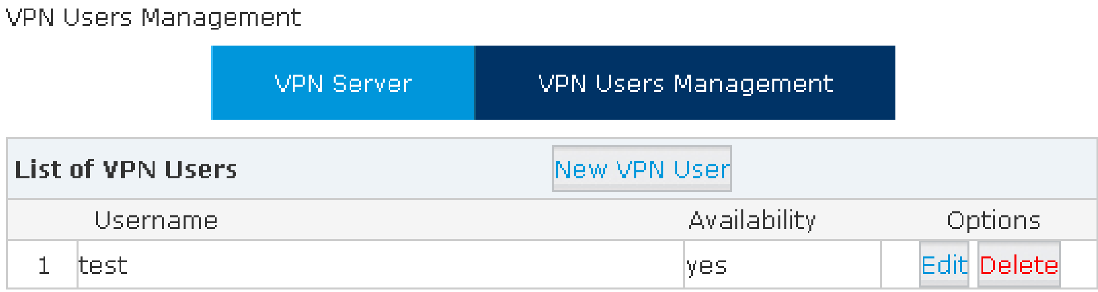
Click【Network Settings】->【VPN Server】:



Reference:

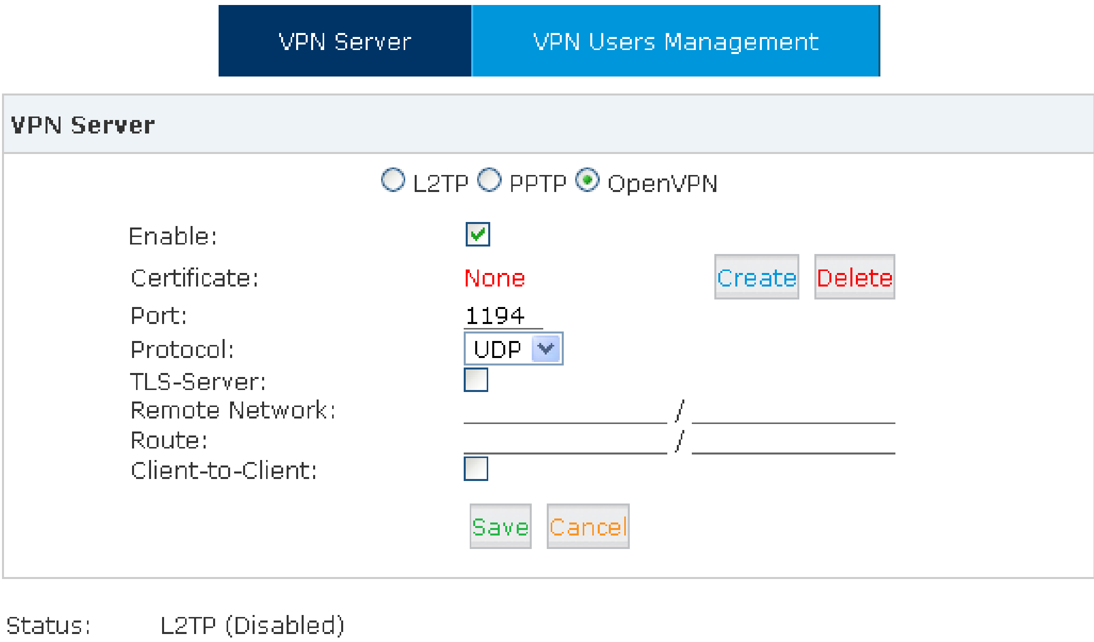
Item	Explanation
VPN Server Mode	Three kinds of VPN servers -- L2TP, PPTP and OpenVPN -- supported（Only one mode can be enabled simultaneously).
Enable	Enable/Disable VPN Server

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



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

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

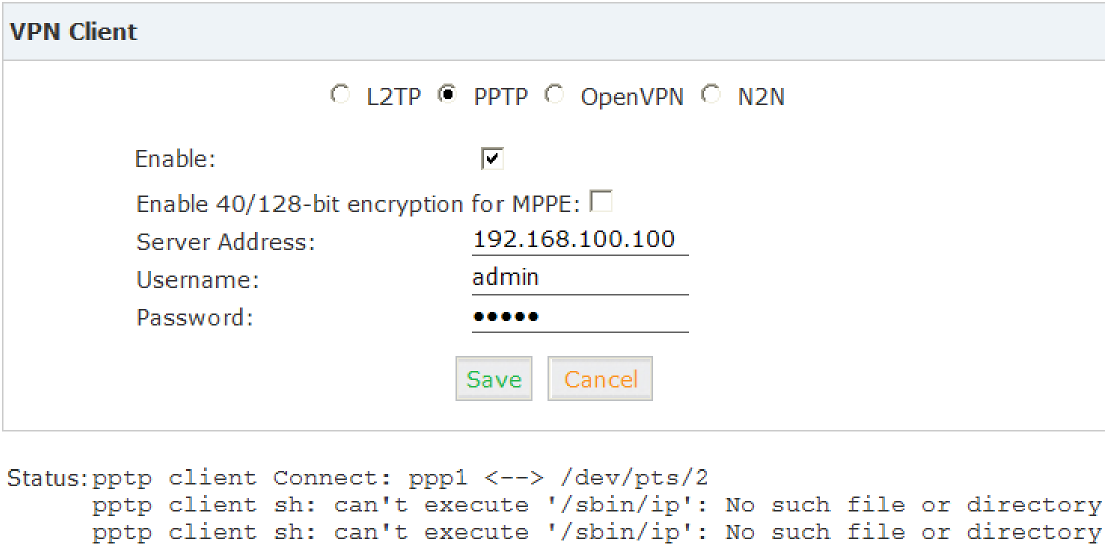


This page is used for management of OpenVPN certificate file.

5.4 VPN Client

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

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

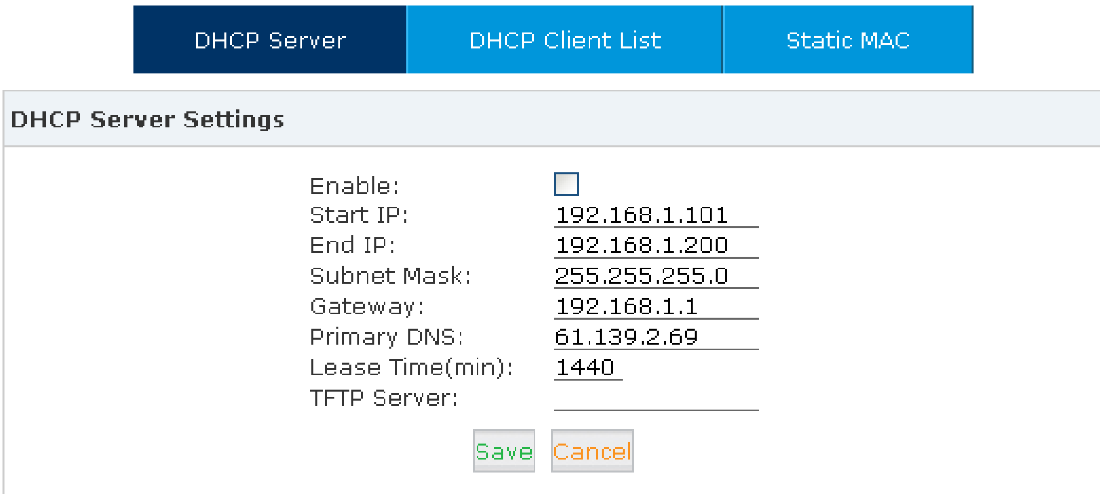


Reference:

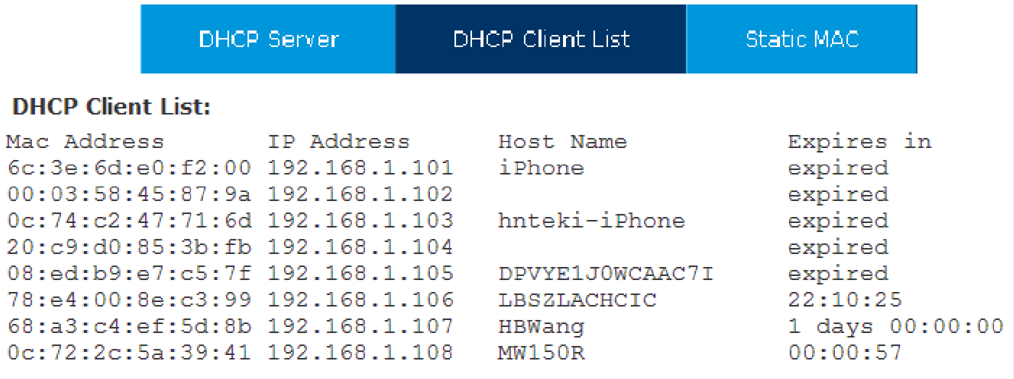
Item	Explanation
VPN Client	Four kinds of VPN Clients supported: L2TP, PPTP, OpenVPN and N2N（Only one mode can be enabled simultaneously）
Enable	Enable/Disable VPN Client

5.5 DHCP server

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



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

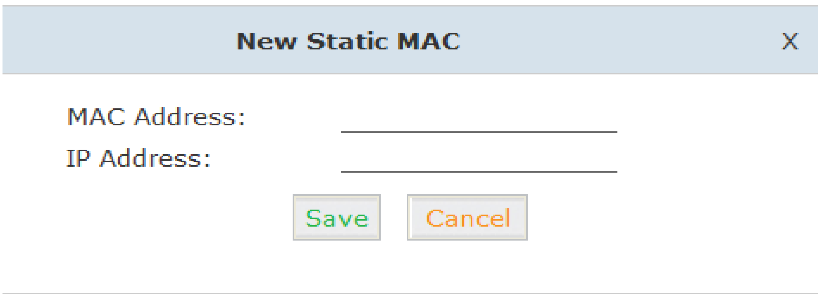


Mac Address	IP Address	Host Name	Expires in
6c:3e:6d:e0:f2:00	192.168.1.101	iPhone	expired
00:03:58:45:87:9a	192.168.1.102		expired
0c:74:c2:47:71:6d	192.168.1.103	hnteki-iPhone	expired
20:c9:d0:85:3b:fb	192.168.1.104		expired
08:ed:b9:e7:c5:7f	192.168.1.105	DPVYE1J0WCAAC7I	expired
78:e4:00:8e:c3:99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

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

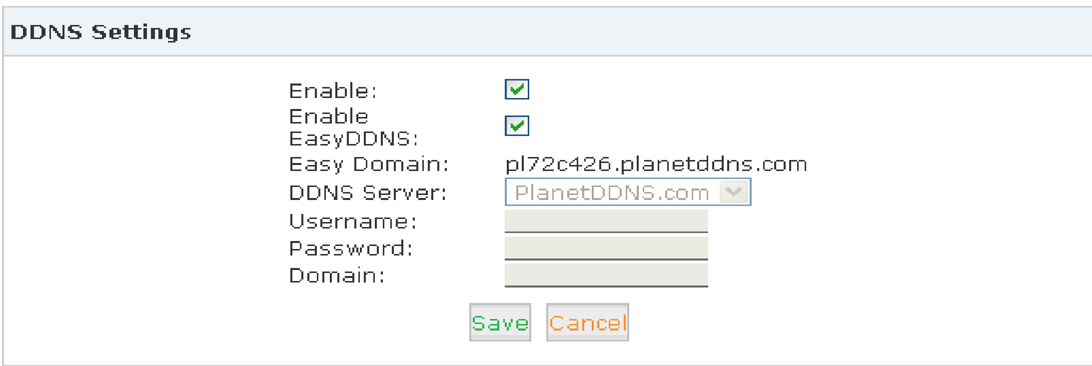
When DHCP Server distributes address, the Client’s MAC address is associated with the IP address, and then the device will get the same IP address every time.

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



5.6 DDNS Settings

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

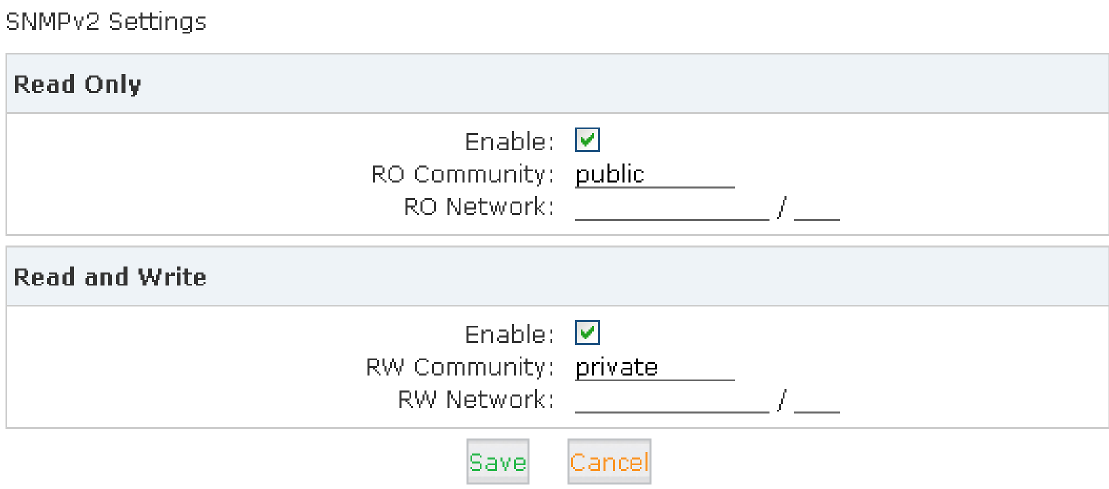


Planet supports DDNS provided by Planet DDNS / Dyndns.org / No-ip.com / zoneedit.com.

5.7 SNMPv2 Settings

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

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

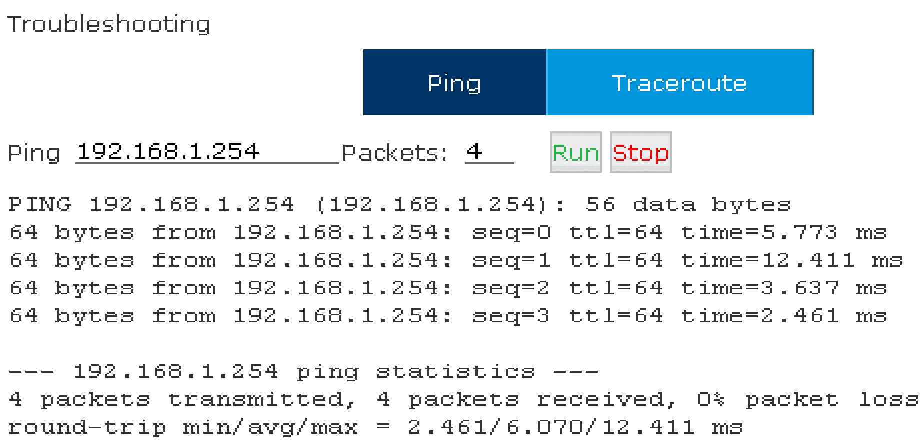


Reference

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

5.8 Troubleshooting

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

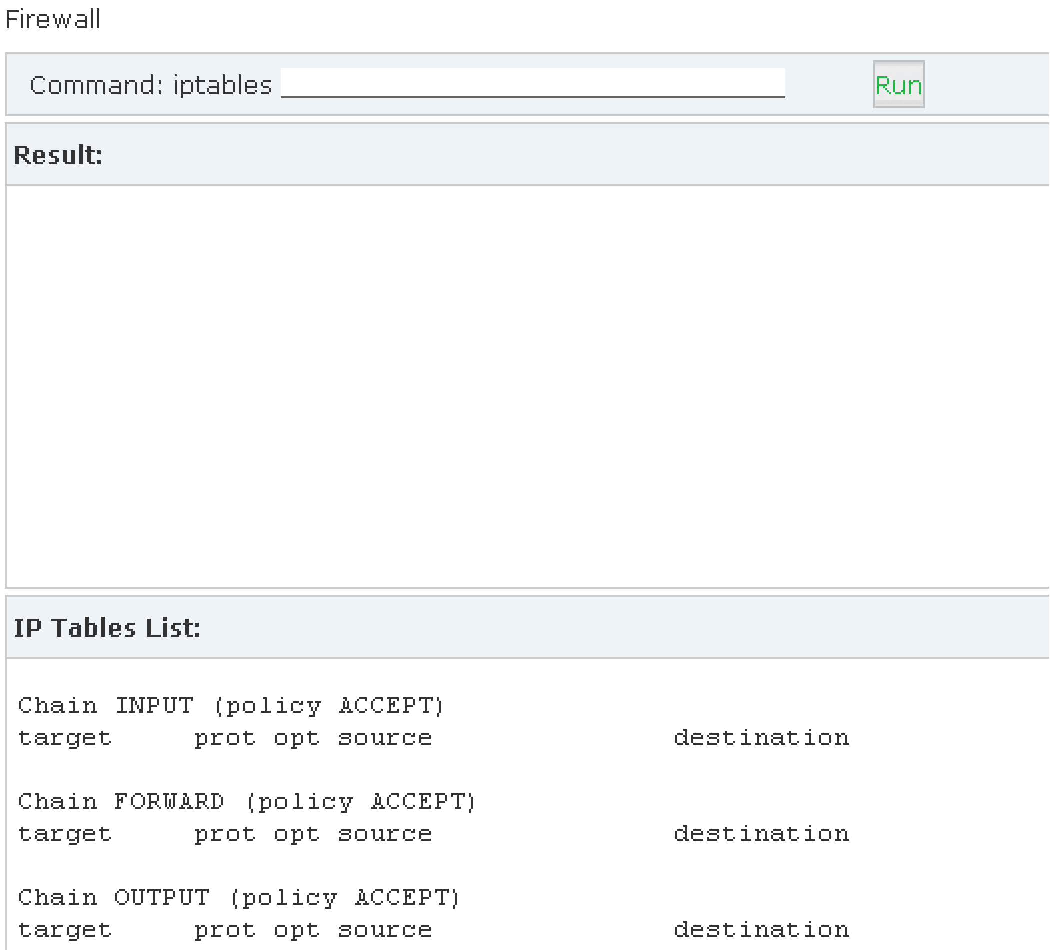


Chapter 6 Security

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

6.1 Network And Country

Click【Security】->【Firewall】

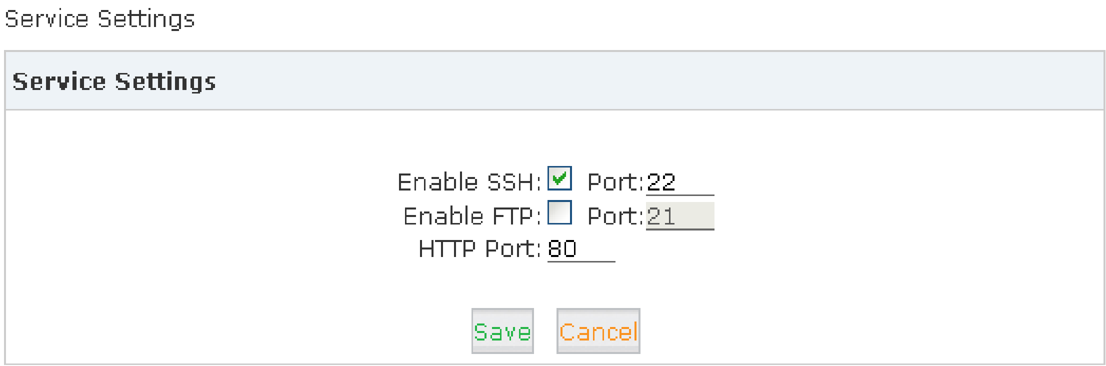


Iptables Command	Explanation
Check iptables list	iptables -L -n
Clear iptables list	iptables -F
Deny an IP(192.168.0.3	iptables -A INPUT -s 192.168.0.3 -j DROP
Deny every IP to access 80 port	iptables -A INPUT -p tcp --dport 80 -j DROP
Deny IP（192.168.0.3） to access port 80	iptables -A INPUT -s 192.168.0.3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

6.2 Service

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

Click【Security】->【Service】:



Enable SSH to login background management system through SSH.

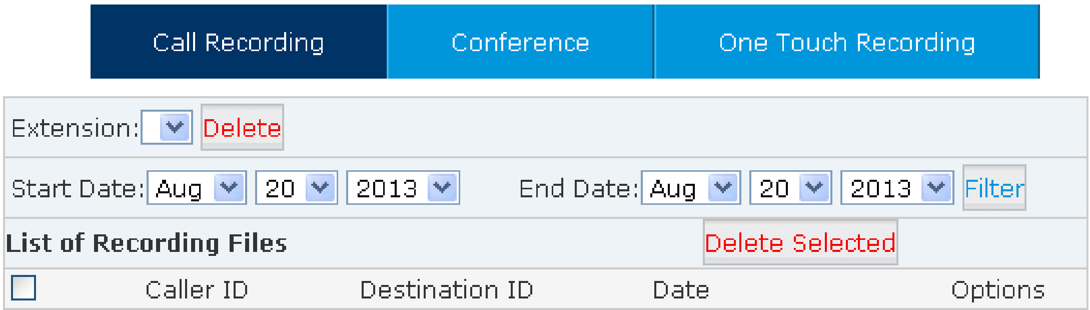
Enable FTP to allow uploading files to system through FTP.

Chapter 7 Report

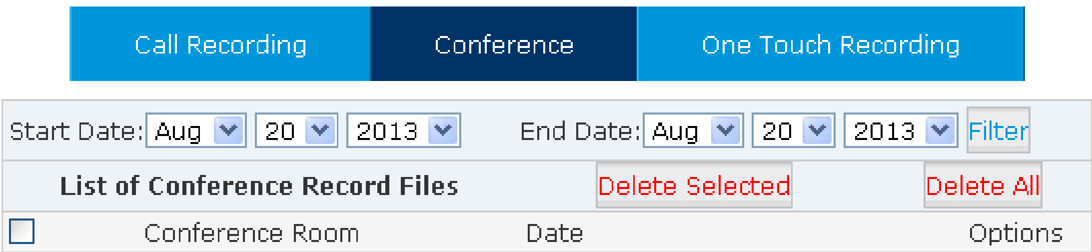
7.1 Record List

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

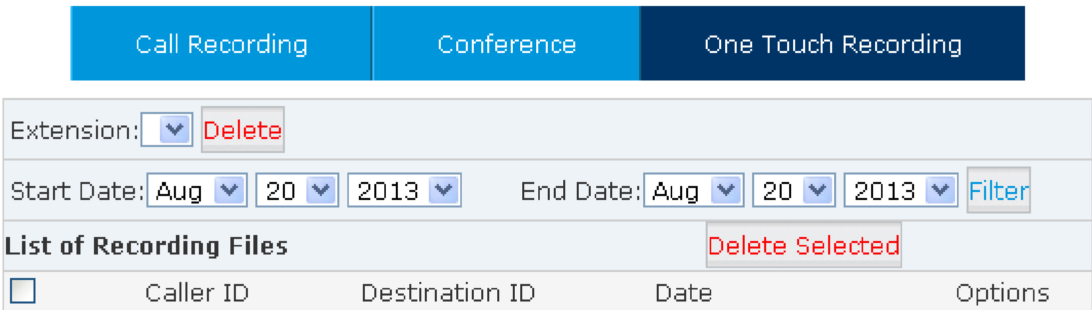
【Record List】:



【Conference】:



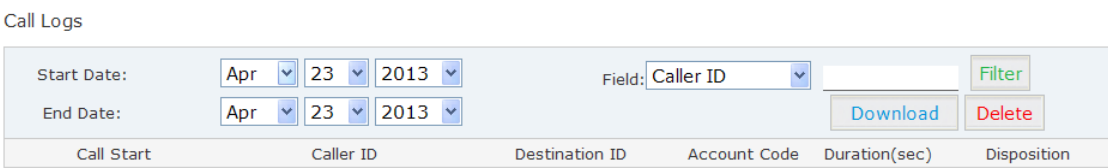
【One Touch Recording】



7.2 Call logs

Check call logs by caller ID or callee ID.

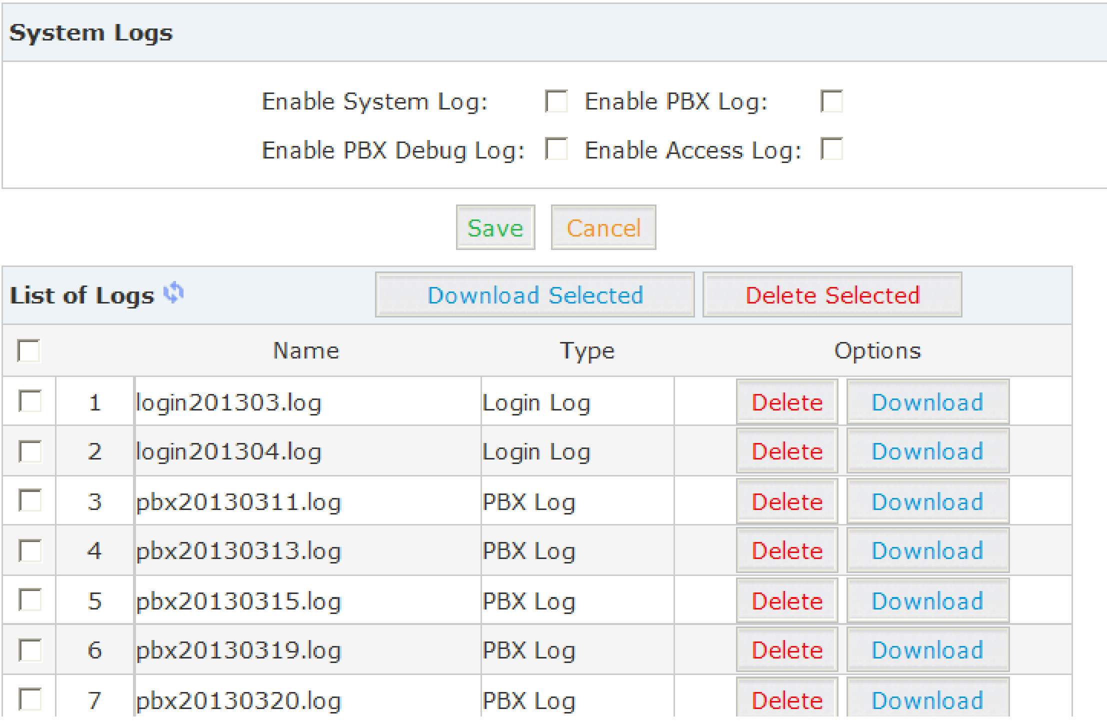
Click【Report】->【Call Logs】:



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

7.3 System logs

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

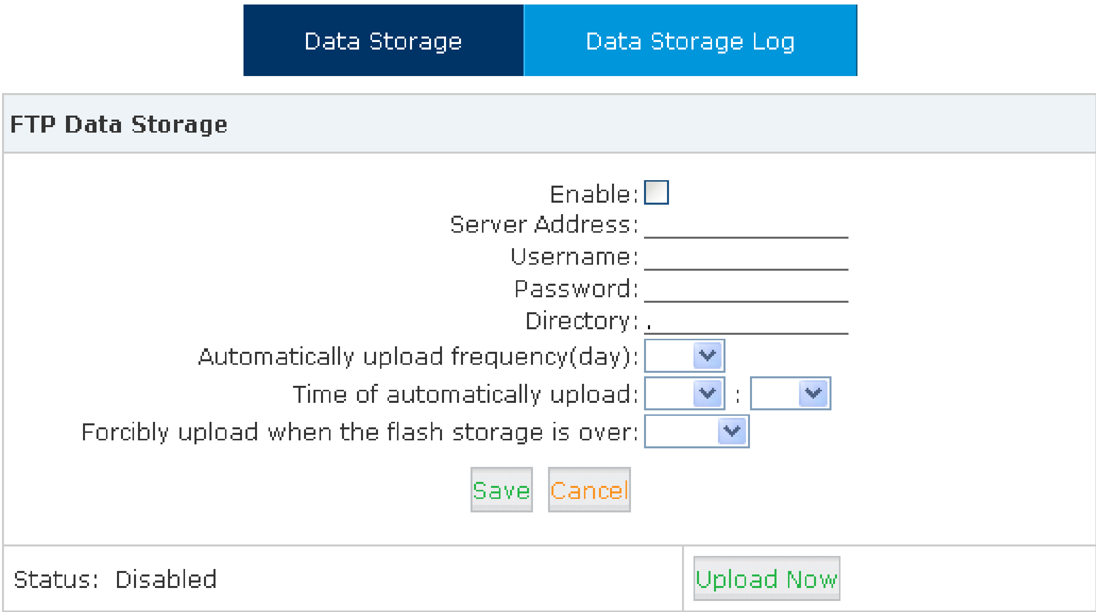


		Name	Type	Options	
	1	login201303.log	Login Log	Delete	Download
	2	login201304.log	Login Log	Delete	Download
	3	pbx20130311.log	PBX Log	Delete	Download
	4	pbx20130313.log	PBX Log	Delete	Download
	5	pbx20130315.log	PBX Log	Delete	Download
	6	pbx20130319.log	PBX Log	Delete	Download
	7	pbx20130320.log	PBX Log	Delete	Download

7.4 Data Storage

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

Click【System】->【Data Storage】:



Reference

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

Check from【Data Storage Log】:

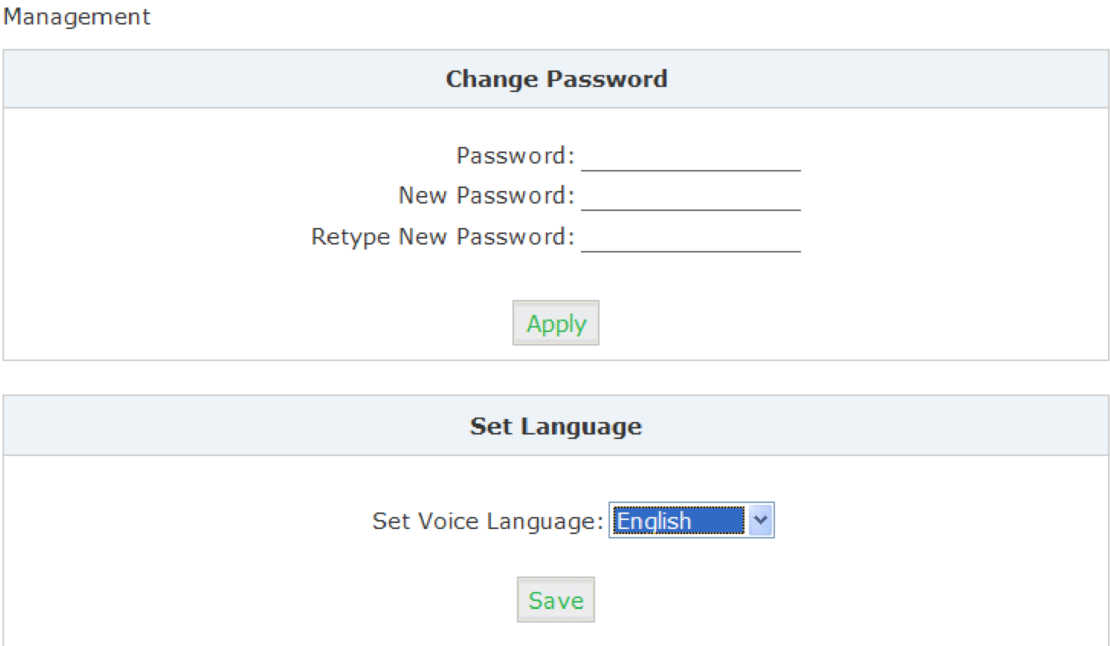
Click【Refresh】to refresh data storage log.

Click【clear】to clear data storage log.

7.5 Management

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

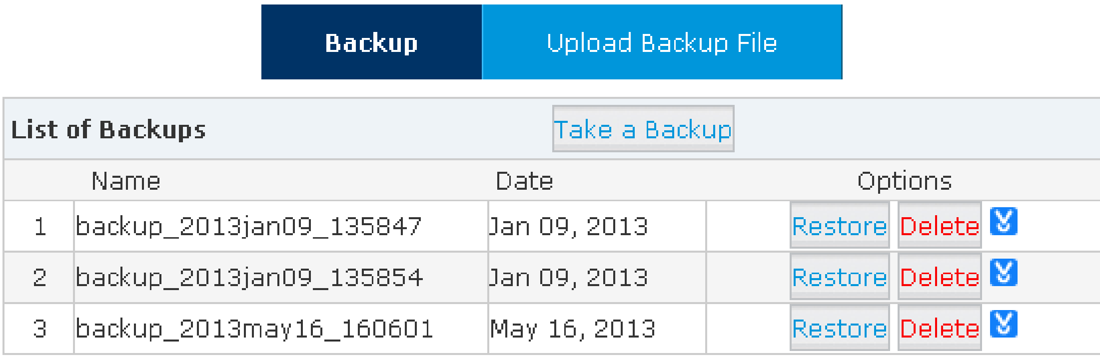
Click 【System】->【Management】:



【Set Language】Choose the voice language you want

7.6 Backup

Click【System】->【Backup】

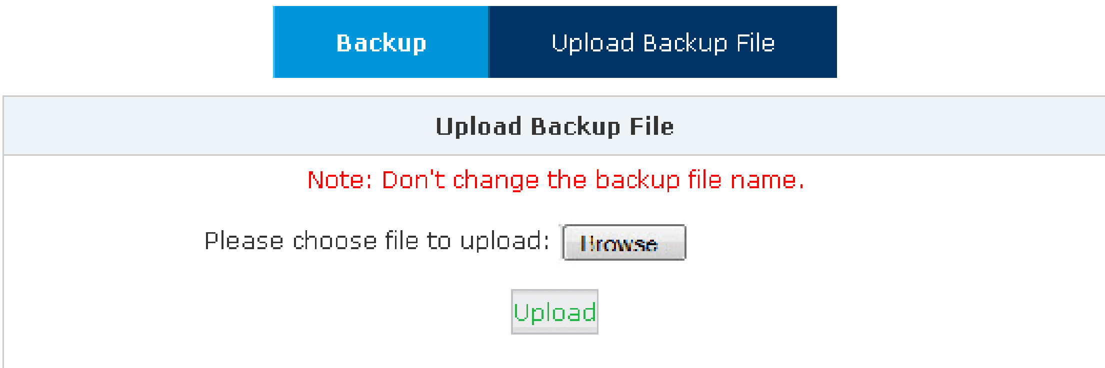


Reference:

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

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

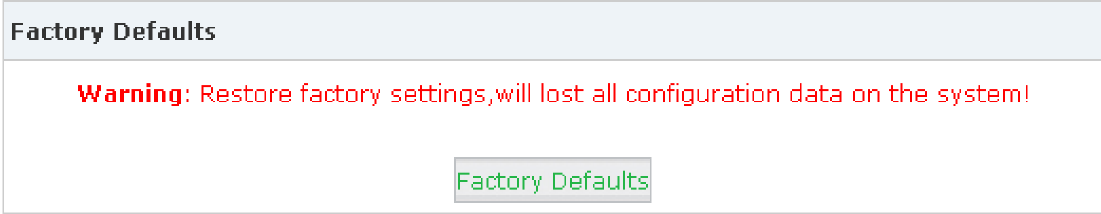
Click【Upload Backup File】to upload the backup file here.

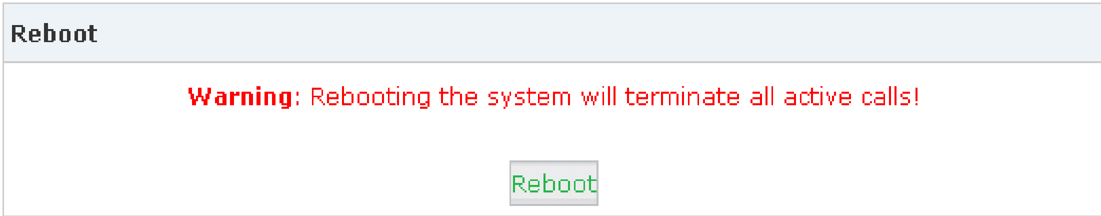


Click【browse】to select the local backup file, and click【Upload】to upload the backup file to system.

7.7 Reset & Reboot

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





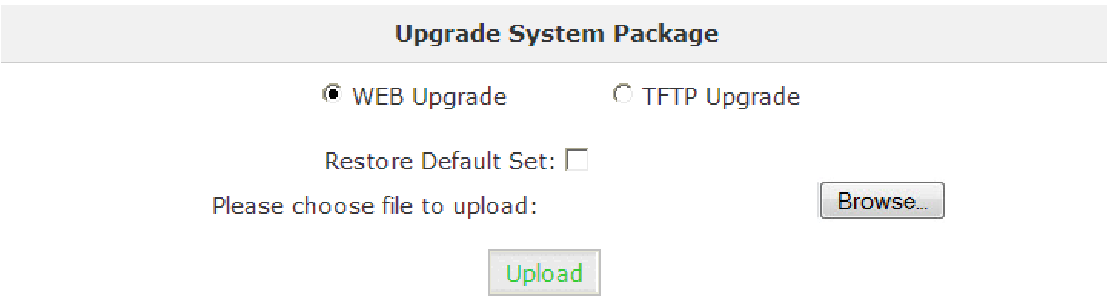
Click【Factory Default】to reset the system to factory default.

Click【Reboot】to reboot the system.

7.8 Upgrade

7.8.1 WEB Upgrade

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

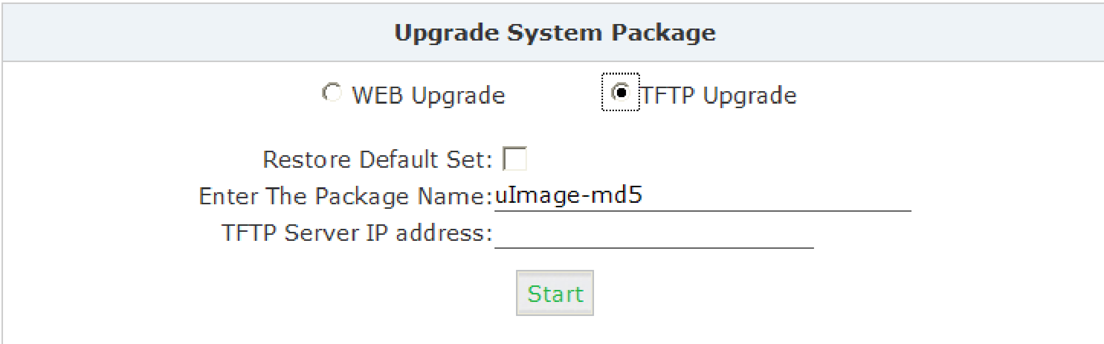


Click【Browse】to select the firmware file, and then click【Upload】to upload the selected firmware to system and finish the upgrading automatically.

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

7.8.2 TFTP Upgrade

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



Reference:

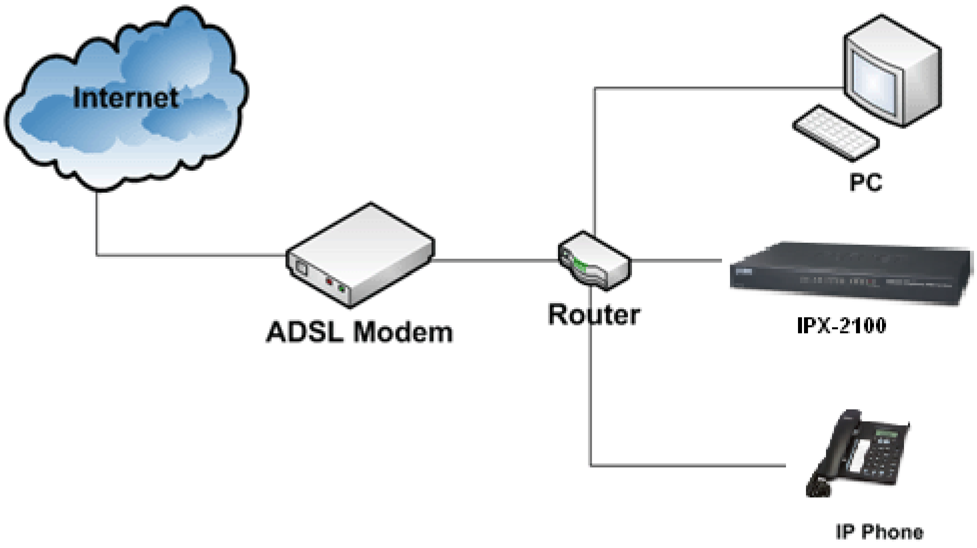
Item	Explanation
Restore Default Set	System will restore to factory defaults after checking this option.
Enter The Package Name	Enter the package name for upgrading.
TFTP Server IP address	Enter your TFTP server IP address.

Chapter 8 Operating Instructions

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

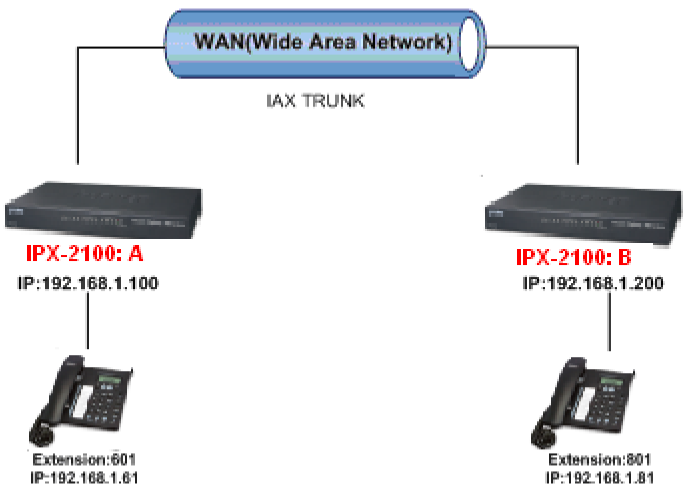
8.1 How to connect the IPX-2100 IP PBX to the Internet

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



8.2 How to combine two IPX-2100 IP PBX in a different network

Normally, two sets of the IPX-2100 are located in different places with different IP addresses for Internet access.



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

Take the following instructions as an example:

Register IPX-2100-B IP to a trunk of IPX-2100-A with authentication.

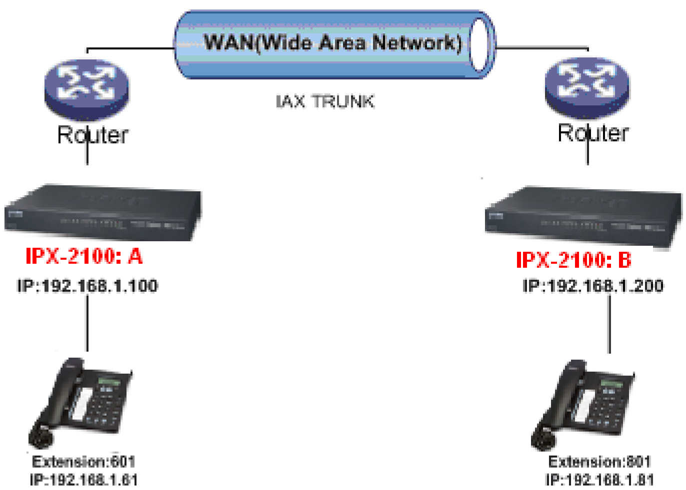
Configuration Rule:

1. IP Phone registers on IPX-2100-A as extension 601.
1. Another IP Phone registers on U50-B as extension 801.
2. IPX-2100-A IP:192.168.1.100.
3. IPX-2100-B IP:192.168.1.200.
4. Extension format of IPX-2100-A: 6XX.
5. Extension format of IPX-2100-B: 8XX
6. Create an extension 888 with password 123456 on IPX-2100-B.
7. All extensions on IPX-2100-A can call extensions on IPX-2100-B with format 8XX.
8. All extensions on IPX-2100-B can call extensions on IPX-2100-A with format 6XX.

For detailed steps, please take chapter 8.2 as reference.

Two sets of IPX-2100 **behind router**

Sometimes the IPX-2100 doesn't have a public IP address, and you have to configure port mapping for your router.

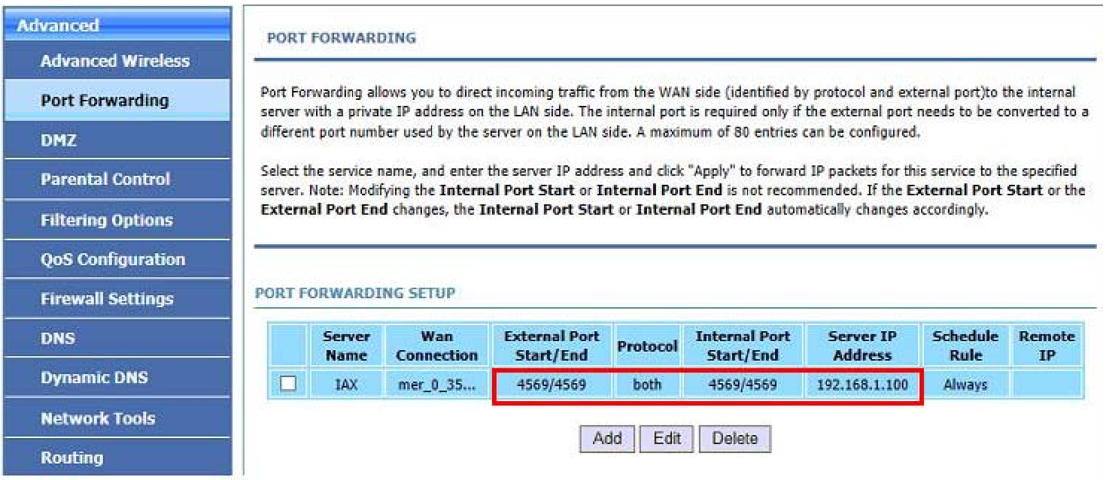


Step1: Configure the mapping rule of IPX-2100-A on the router.

The IPX-2100-B is connected behind the router, and registers on IPX-2100-A through internet. You need to configure the port mapping of IAX2 port(4569) on the router. Then, all data received from eth0 port of router(192.168.1.100:4569) will be sent to IPX-2100-A

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

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



Step2: IPX-2100 Configuration

Configure the trunk and dial plan on IPX-2100-B, and register IPX-2100-B IP to IPX-2100-A. The configuration is the same as the above, but you have to replace the public IP address with the internal IP:192.168.1.21.

Step3: Configure port mapping rule of IPX-2100-B on the router

Configure port mapping of IPX-2100-B on the router according to Step1.

Step4: Connect two sets of the IPX-2100 and make the call

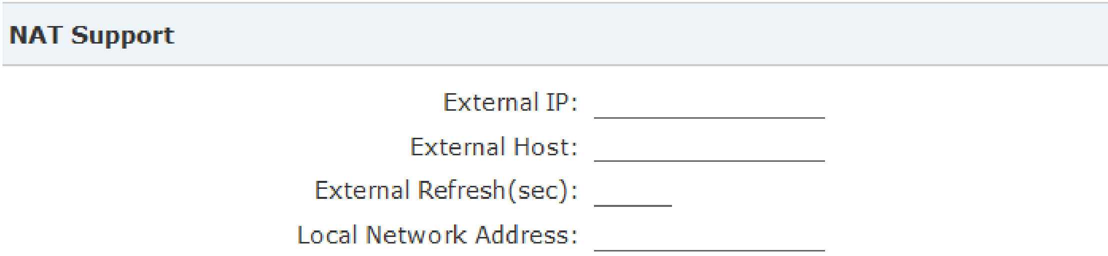
Create extension 601 on IPX-2100-A, extension 801 on IPX-2100-B, and create the correct outbound rule.

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

8.3 How to resolve the problem about hearing one side only

If the IPX-2100 is behind router, to resolve the problem, please set up IP address as shown below:

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



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

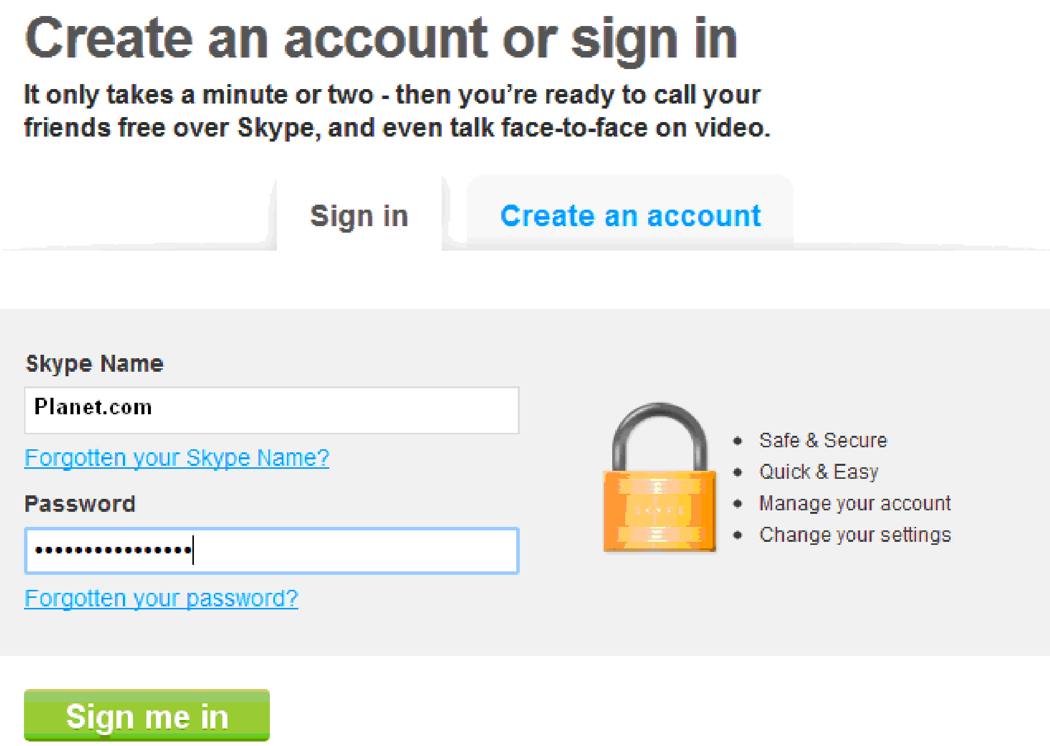
8.4 How to use Skype account in IPX-2100

[Answer] :

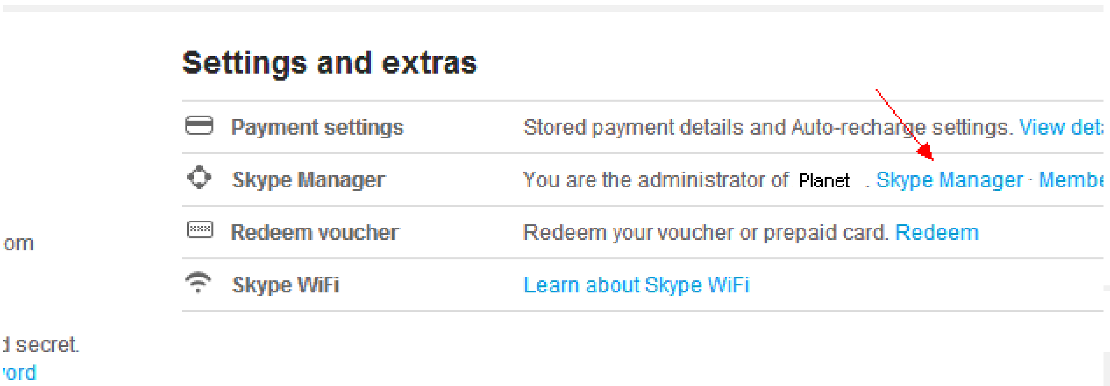
Notice: The fee of your business account is much more than **€50** when you use the account for the first time.

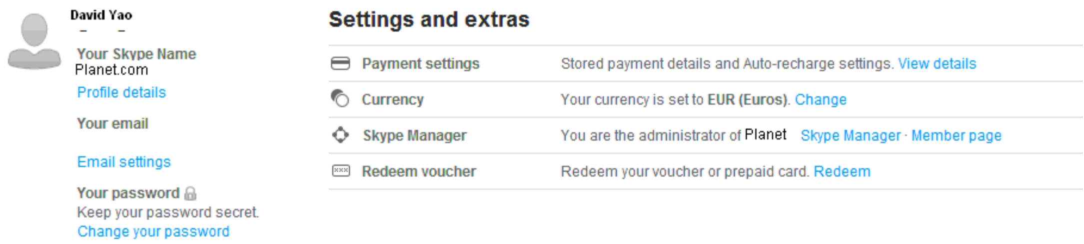
1. https://login.skype.com

Sign in with the business account.

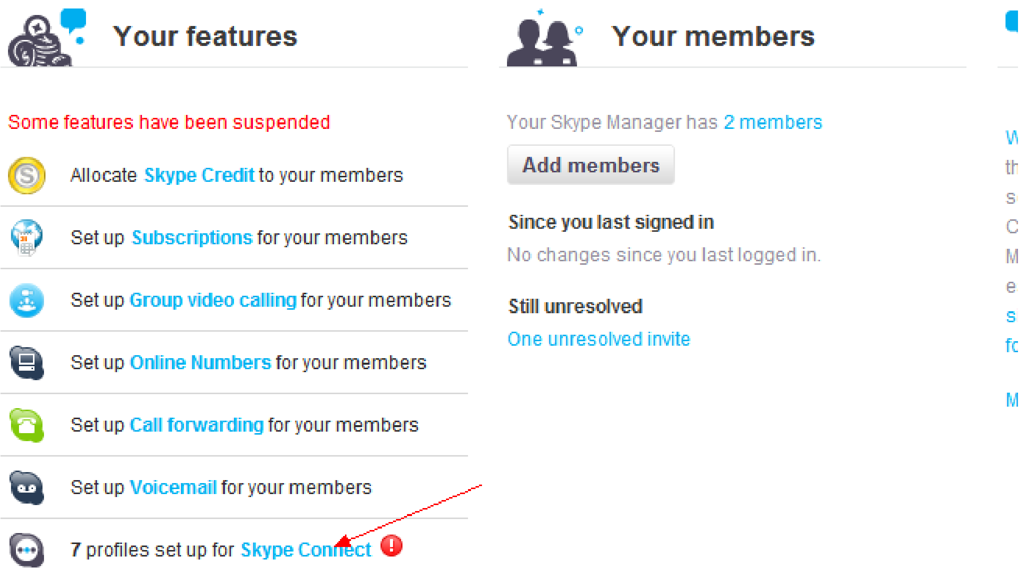


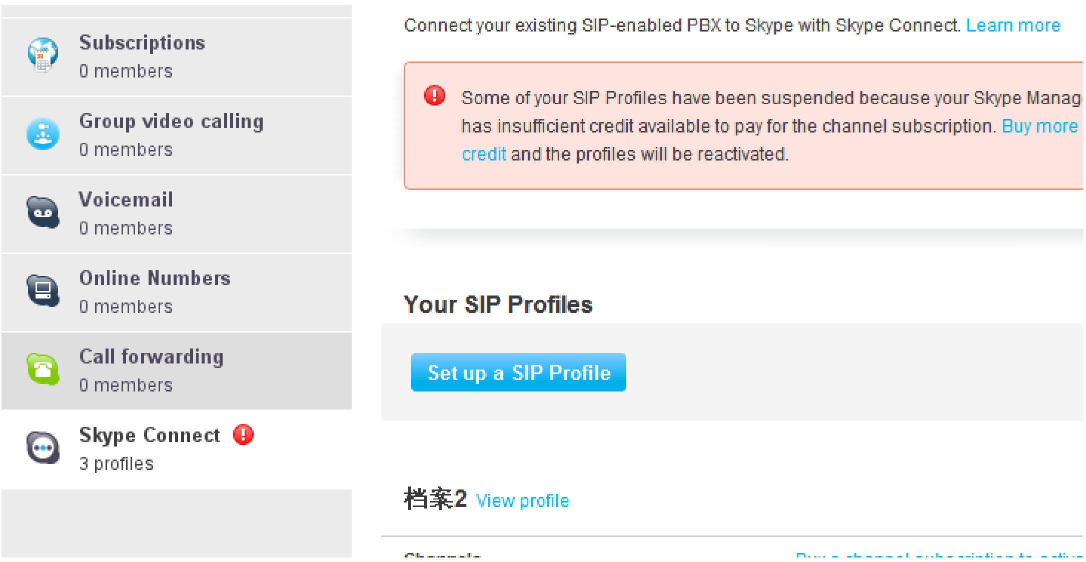
2. When you have signed in, at the end of this page, you will find the **Skype Manager**, Please click it.



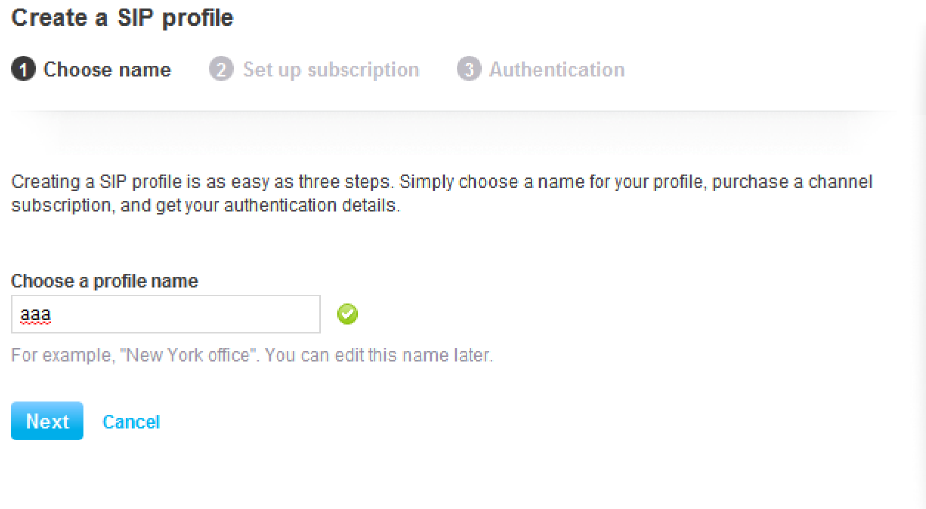


3. Please click the **Skype connect**

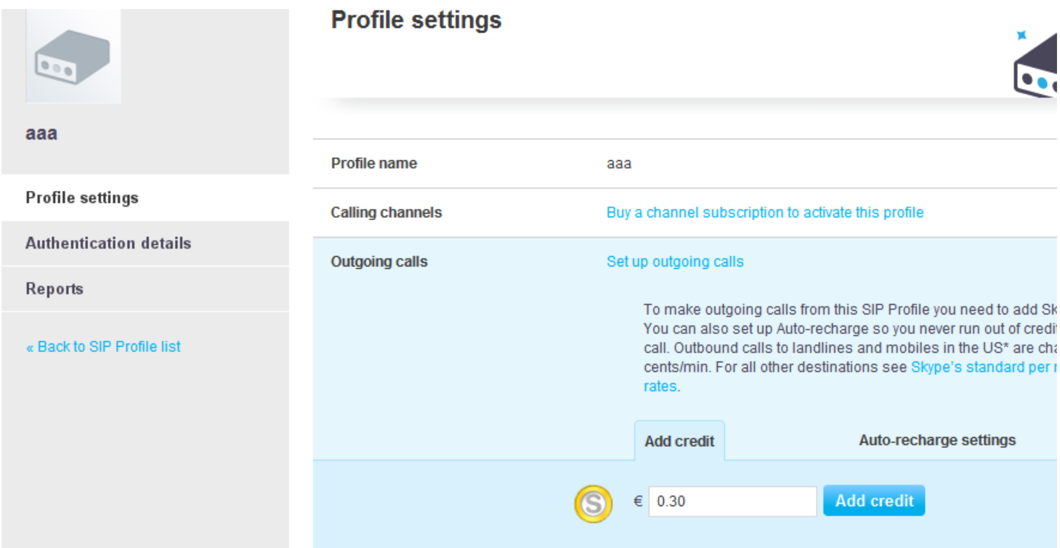




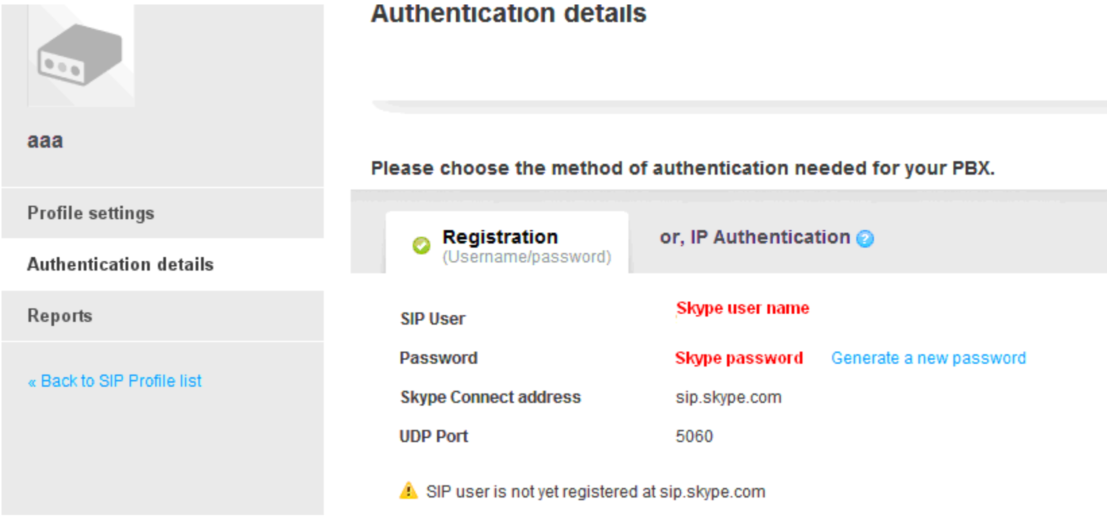
4. **Create a SIP profile**



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



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



5. Settings on IPPBX

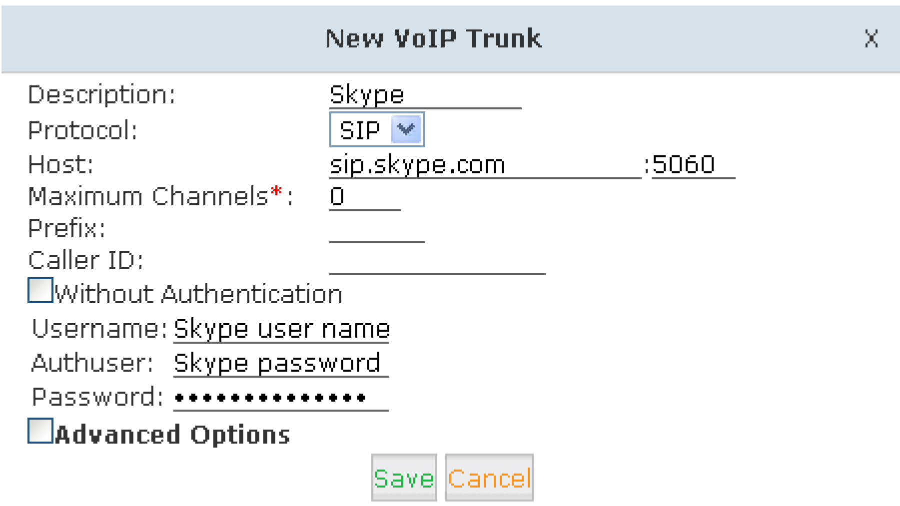
5.1. Build one sip trunk with Skype for sip account

Provider Type: Custom Trunk

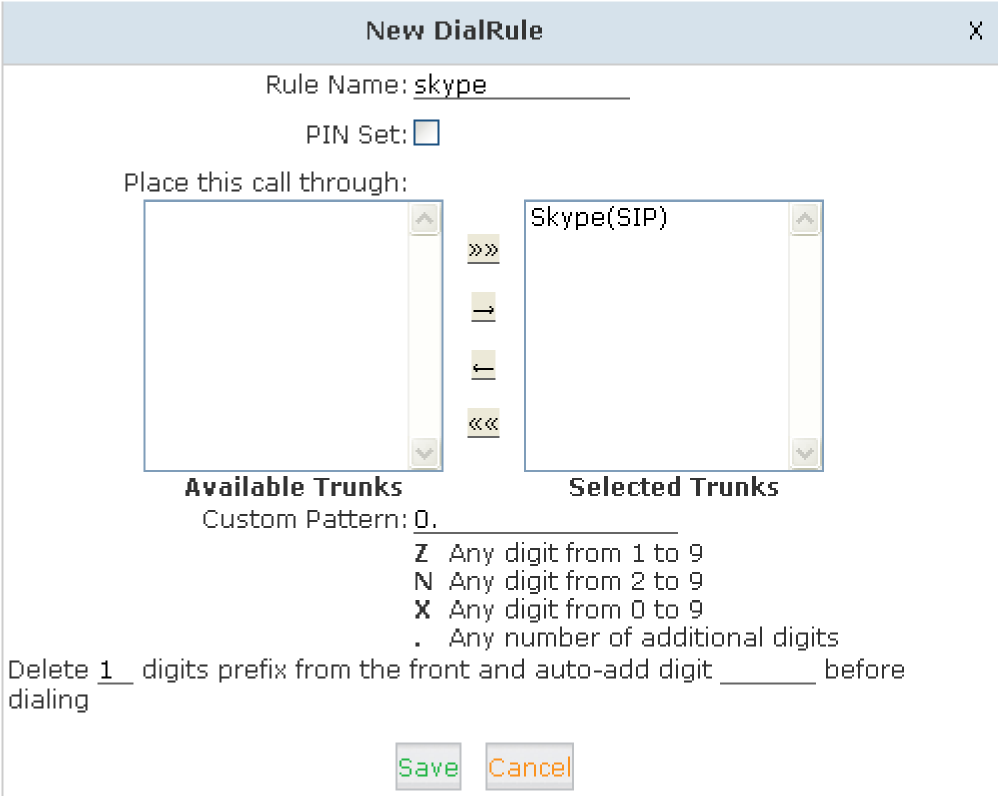
Host: sip.skybe.com

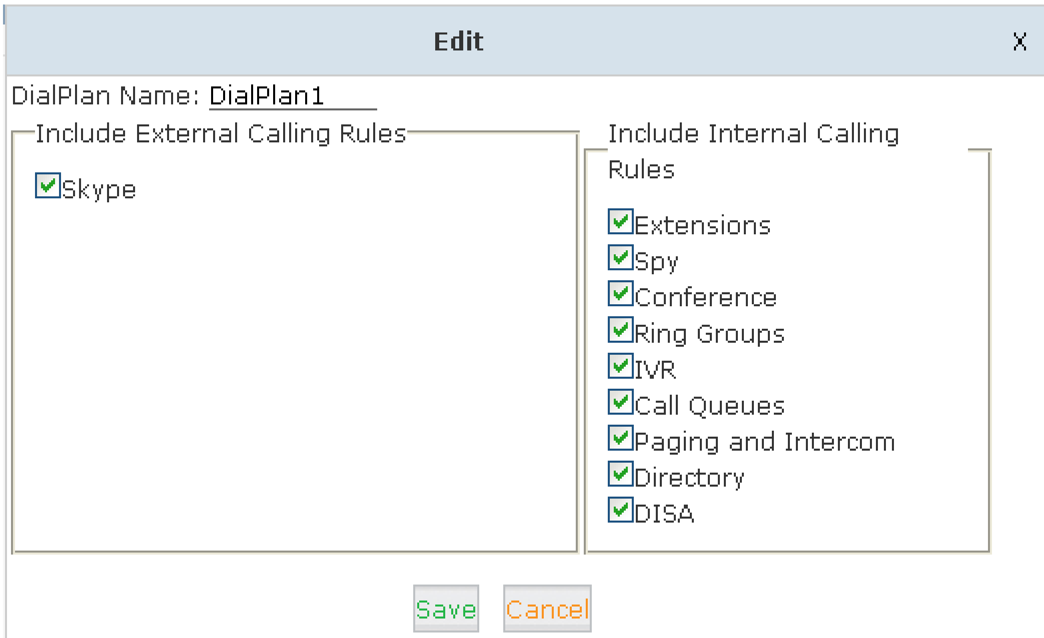
User name: the user name you defined in Authentication detail

Password: the password you defined in Authentication detail



5.2. Set one outbound rule





5.3. Make an outbound call

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

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

5.4. Set inbound rule