

Internet Telephony PBX System IPX-2200/IPX-2500

User's Manual

Internet Telephony PBX



System

► IPX-2200

▶ IPX-2500



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

Energy Saving Note of the Device

This power required device does not support Standby mode operation. For energy saving, please remove the DC-plug or push the hardware Power Switch to OFF position to disconnect the device from the power circuit.



Without removing the DC-plug or switching off the device, the device will still consume power from the power circuit. In view of Saving the Energy and reducing the unnecessary power consumption, it is strongly suggested to switch off or remove the DC-plug from the device if this device is not intended to be active.

WEEE Warning



To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal

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Chapter 1. Introduction



Intuitive, Ease-of-Use IP PBX Machine Management

PLANET IPX-2200/IPX-2500 IP PBX telephony system is SIP-based for optimizing communications among the small and medium businesses. The IPX-2200 and IPX-2500 are able to accept 200/500 user registrations, and easy to manage a full voice over IP system with the convenience and cost advantages.

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

The IPX-2200/IPX-2500 integrates **up to 8 calls** via the IPX-21FO (4 FXO) and IPX-21GS (4 GSM) modules to form a feature-rich PBX system that supports seamless communications between the existing PSTN calls, analog, IP phones and SIP-based endpoints.

Replacing Old PBX Easily without New Wiring

Cost-effective, easy-to-install and simple-to-use, the IPX-2200/IPX-2500 converts standard telephones to IP-based networks. It enables the service providers and enterprises to offer users traditional and enhanced telephony communication services via the existing broadband connection to the Internet or corporation network.

With the IPX-2200/IPX-2500, home users and companies are able to save the installation cost and extend their past investments in telephones, conferences and speakerphones. The IPX-2200/IPX-2500 can be the bridge between traditional analog systems and IP network with an extremely affordable investment.

Distributed VoIP Network Infrastructure

For the new-generation communication age, the IPX-2200/IPX-2500 supports IPv6 and VPN (client/server) connection to provide users with more flexible and advantageous



communications products. With PLANET DDNS function, the IPX-2200/IPX-2500 also helps users to apply and remember the login information easier. Moreover, its multiple language feature helps user to quickly and friendly manage the system. The IPX-2200/IPX-2500 supports Lync server to which smart phone (using third-party app) and analog phone are connected via its communication with other devices of Lync server.

Standard Compliance

Compliant with the Session Initiation Protocol 2.0 (RFC 3261), the IPX-2200/IPX-2500 are able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

Green IP Office

Virtual fax functionality on IPX-2200/IPX-2500 system allow faxes to be sent and received without requiring a fax machine. This useful feature will allow businesses to demonstrate their green credentials while at the same time reduce fax related costs across the enterprise. Inbound faxes can be automatically received and converted to TIF files and saved in the IPX-2200/IPX-2500 system. It is also possible to configure the IPX-2200/IPX-2500 system to send the TIF files to a user's email box. Sending outbound faxes is as easy as uploading a file from the extension user web portal, thus creating a paperless or green office.

Full Security with VPN Support

The IPX-2200/IPX-2500 VPN securely and cost-effectively connects geographically disparate offices of an organization, creating one cohesive virtual network. The IPX-2200/IPX-2500 VPN technology is also used by ordinary Internet users to connect to proxy servers for the purpose of protecting one's identity. They include VPN server and client function that can support users full security login.





1.1 Features

System Highlights

- 60 concurrent calls and up to 200 registers (For IPX-2200)
- 100 concurrent calls and up to 500 registers (For IPX-2500)
- HD voice codec G.722 for perfect voice quality
- Virtual Fax for green office
- Voicemail to Email for not missing any important message
- Paging and intercom function strengthens work efficiency.
- Built-in SIP Proxy Server following RFC 3261
- Multiple Languages of GUI for international business
- Web-based Control Panel for easy configuration and management of the system.
- Hardware Echo Cancellation module for great and smooth communication.
- Strong security features protect your system from hacking.
- Supports maximum 8 ports for FXO/GSM (on 2 slots)
- Records voice and voicemail to external USB disk
- Supports Lync server

Codec and Protocol

- SIP 2.0 (RFC 3261), IAX2 compliant
- Audio Codec: G.722/G.711-Ulaw/G.711-Alaw/G.726/G.729/GSM/SPEEX
- Video Codec: H.261/H.263/H.263+/H.264
- DTMF: RFC 2833, SIP info, in-band

Network and Security Features

- DDNS Client (PLANET DDNS, Dyndns.org, No-ip.com, zoneedit.com, freedns.afraid.org, www.oray.com, 3322.org)
- DHCP Server/SNMP v1/v2
- IEEE 802.1Q of VLAN
- IPv4/IPv6, SIP over IPv6
- Manual Configuration of Static Route Table
- Troubleshooting (Ping, Traceroute)
- VPN Server (L2TP/PPTP/OpenVPN/IPSec, up to 20 connections for VPN clients)
- VPN Client (L2TP/PPTP/OpenVPN/N2N/IPSec)
- Refuse SIP Register DoS
- Refuse Abort Invite Dos



- Refuse SSH Login DoS
- Firewall/SRTP
- Enhances HTTPS connection

PBX Features

- Auto-Provision (PLANET/Cisco IP Phone)
- Black List
- BLF (Busy Lamp Field), Speed Dial
- CDR (Call Detailed Record) (20000 records)
- Conference Room (36 rooms)
- Call Queue Record, Ring Group Record
- DoD (Direct Outward Dialing) and DID (Direct Inward Dialing) numbers
- DISA (Direct Inward System Access)
- DND (Do Not Disturb)
- Feature Codes, Flash Operation Panel
- Flexible Dial Plan, Follow Me
- IVR (Interactive Voice Responses)
- LDAP Server for phonebook
- Multi-language System Prompt
- Multiple Languages of GUI
- One Number Stations
- Phone Book/PIN Set
- Phonebook/LDAP (5000 contacts)
- Record Files Download
- Ring Group, SIP Trunk
- Skype for SIP/Smart DID/System Log/System Backup
- T.38 fax (pass-through)/time-based rule
- Virtual Fax/Voicemail & Voicemail to Email
- WebRTC

Call Features

- Attend Transfer, Call Waiting
- Call Back, Call Forward, Call Group
- Call Hold, Call Paging and Intercom
- Call Park, Call Pickup, Callback
- Call Center Queues (36)
- Call Record, Call Route, Blind Transfer



- Caller ID, Dial by Name
- Customized IVR, On-hold Music, Transfer
- Three-way Conferencing, Video Call



1.2 Package Contents

Thank you for purchasing PLANET Internet Telephony PBX system, IPX-2200 and IPX-2500. Open the box of the Internet Telephony PBX system and carefully unpack it. The box should contain the following items:

- Internet Telephony PBX system unit x 1
- Quick Installation Guide x 1
- User's Manual CD x 1
- Power Cord x 1
- RJ45 x 1
- Bracket x 2

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

1.2.1 Physical Specifications of IPX-2200

Dimensions

Dimensions (W x D x H)	343 x 154 x 35 mm
Weight	1.4 kg (gross weight), 1.8 kg (with package)

Front Panel

() <u>PLANET</u>	• PWR	• 515	WAN	LAN				 ■ EXCR 85M ■ EXS 4 ÷ Kinging 	Internet Telephony PBX System
IPX-2200									

Rear Panel



LED definitions

Front Panel LED	Status	Description						
PWR	Steady Green	PBX Power ON						
FWR	Off	PBX Power OFF						
	Blinking Green	System is working						
SYS	On	System doesn't boot						
	Off	System failure						



Front Panel LED	Status	Description				
	Blinking Green	Data transfer				
WAN	On	PBX network connection is established				
	Off	Waiting for network connection				
	Blinking Green	Data transfer				
LAN	On	PBX network connection is established				
	Off	Waiting for network connection				
	Steady Red	Ready/Standby				
FXO	Flashing	Ringing				
	Off	Module not available				

Physical interfaces description

1	Power Switch	Switch the power on or off							
2	Power Cord	AC 100~240V, 50/60Hz, 1.5A max							
3	WAN/LAN	The WAN/LAN port supports auto negotiating Fast Ethernet 10/100BASE-TX networks. The WAN port allows your IP PBX to be connected to an Internet Access device, e.g., router, cable modem or ADSL modem through a Cat5 twisted-pair Ethernet cable.							
4	HDMI Port	For video output (factory use)							
5	USB	For external store device to store voice and voicemail							
6	Audio In/Out	For external paging							
7	Module Slot 1/Slot 2	 2 external slots with compliant FXO/FXS/GSM module -FXO module is connected to PBX or CO line with RJ11 analog line. FXO port is connected to the extension port of a PBX or directly connected to a PSTN line of carrier -GSM module is connected to Global System for Mobile Communications (GSM) with SIM card 							



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



1.2.2 Physical Specifications of IPX-2500

Dimensions

Dimensions (W x D x H)	343 x 154 x 35 mm
Net Weight	1.4 kg (gross weight), 1.8 kg (with package)

Front Panel

PLANET	PWR SYS	WAJ	● N LAN	•	— SLOT	r1 ● 3	•	• •	SLOT 2	● IXO/GSM ● IXS 4 ★ Kinging	Internet Telephony PBX System
IPX-2500											

Rear Panel



LED definitions

Front Panel LED	Status	Description						
	Steady Green	PBX Power ON						
PWR	Off	PBX Power OFF						
	Blinking Green	System is working						
SYS	On	System doesn't boot						
	Off	System failure						
	Blinking Green	Data transfer						
WAN	On	PBX network connection is established						
	Off	Waiting for network connection						
	Blinking Green	Data transfer						
LAN	On	PBX network connection is established						
	Off	Waiting for network connection						
	Steady Red	Ready/Standby						
FXO	Flashing	Ringing						
	Off	Module not available						
	Steady Red	Ready/Standby (SIM card inserted)						
GSM	Flashing	Ringing						
	Off	No SIM card inserted						
	Steady Green	Ready/Standby						
FXS	Flashing	Ringing						
	Off	Module not available						



Physical interfaces description

1	Power Switch	Switch the power on or off
2	Power Cord	AC 100~240V, 50/60Hz, 1.5A max
3	WAN/LAN	The WAN/LAN port support auto negotiating Fast Ethernet 10/100BASE-TX networks. The WAN port allows your IP PBX to be connected to an Internet Access device, e.g., router, cable modem or ADSL modem through a Cat5 twisted-pair Ethernet cable
4	HDMI Port	For video output (factory use)
5	USB	For external store device to store voice and voicemail
6	Audio In/Out	For external paging
7	Module Slot 1/Slot 2	 2 external slots with compliant FXO/FXS/GSM module -FXO module is connected to PBX or CO line with RJ11 analog line. FXO port is connected to the extension port of a PBX or directly connected to a PSTN line of carrier -GSM module is connected to Global System for Mobile Communications (GSM) with SIM card



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



1.3 Specifications

	IPX-2200	IPX-2500			
Droduct					
Product	Internet Telephony PBX system	Internet Telephony PBX system			
	(200 SIP Users registrations)	(500 SIP Users registrations)			
Hardware Specification					
WAN	1 x 100BASE-TX RJ45 for WAN, conn router	ecting to broadband modem or a WAN			
LAN	1 x 100BASE-TX RJ45 for LAN, conne	ecting to a LAN switch			
HDMI Port	For video output (factory use)				
USB	For external store device to store voice	e and voicemail			
Audio In/Out	For external paging				
2 Slots	Supports maximum 8 ports (FXO/GSN	l)			
USB	Store data for external disk				
LED Indications	PWR: 1, LNK/Off SYS: 1, LNK/Off WAN: 1, LNK/Off LAN: 1, LNK/Off SLOT: 2, FXO/GSM (Red), FXS (Green)				
Dimensions (W x D x H)	343 x 154 x 35 mm				
Power Requirements	100 - 240 VAC	100V - 240 VAC			
EMC/EMI	CE, FCC Class B, RoHS				
Protocols and Standard					
Standard	SIP 2.0 (RFC3261), IAX2				
Protocols	RFC 793 TCP RFC 826 ARP RFC 1034, 1035 DNS 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 over IP	T.38 Fax (pass-through) Image: Constraint of the second	on fax machine, SIP provider and			
Voice Processing	DTMF detection and generation In-band and RFC 2833, SIP info				

Internet Telephony PBX System IPX-2200/IPX-2500



Protocols	SIP 2.0 (RFC-3261), TCP/IP, UDP/RTP/RTCP, HTTP/HTTPS, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE					
System Capacity						
System Capacity	 60 concurrent call legs Up to 200 IP phone registers/extensions Recording and Voicemail (GSM/default): 1500 hours Wav: 150 hours 	 100 concurrent call legs Up to 500 IP phone registers/extensions Recording and Voicemail (GSM/default): 75000 hours Wav: 7500 hours 				
Network and Configuration						
Access Mode	Static IP, PPPoE, DHCP					
Environment						
Operating Environment	0~40 degrees C 5~95% humidity					



Chapter 2. Installation Procedure

2.1 Web Login

- Step 1. Connect a computer to a LAN port on the IPX-2200 or IPX-2500. Your PC must be set up to the same domain of 192.168.0.X as that of the IPX-2200 or IPX-2500.
- **Step 2.** Start a web browser. To use the user interface, you need a PC with Internet Explorer (version 8 and higher), Firefox, or Safari (for Mac).
- Step 3. Enter the default IP address of the IPX-2200 or IPX-2500: https://192.168.0.1 in the URL address box.
- **Step 4.** Enter the default user name <u>admin</u> and the default password <u>admin</u>, and then click Login to enter Web-based user interface.

(Default IP)

Default LAN IP: https://192.168.0.1

Default WAN IP: https://172.16.0.1

Default User Name: admin

Default Password: admin

PLANET Retworking & Communication	Inte	ernet Tel	ephon	y PBX Sys	em
Usernam	ie: [
Passwor	d:				
Languag	e:	English	1		
				Login	

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



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 \rightarrow Network



Network

IPv4 Set	tings IPv6 S		Settings VLAI		N Settings		
WAN Port Setup							
	IP Assig						
	I	P Address:	192.168.1.19	97			
	Sub	onet Mask:	255.255.255	.0			
		Gateway:	192.168.1.2	54			
	Pri	mary DNS:	8.8.8.8				
	Altern	ative DNS:	168.95.1.1				
LAN Port Setup							
IP Address:	192.168.0).1	Subn	et Mask:	255.255.255.0		
IP AddressV1:			Subnet	MaskV1:			
IP AddressV2:			Subnet	MaskV2:			



Step 2. Edit your WAN port IP information.

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

Network

IPv4 Set	ttings	IPv6 S	Settings	VLA	N Settings	
WAN Port Setup						
	Sul	IP Assig P Address: bnet Mask: Gateway: imary DNS: ative DNS:	19 Static DHCP 25 PPPoE 192.168.1.2	7 0 54		
LAN Port Setup						
IP Address: IP AddressV1: IP AddressV2:	192.168.0).1	Subnet	net Mask: MaskV1: MaskV2:	255.255.255.0	

Figure 2-4. Selection of IP Connection Type





Chapter 3. Basic Configuration

3.1 Preparation Before Operation

What kind of IP phone can be used with the IP PBX IPX-2200 and IPX-2500?

• Our IPX-2200 and IPX-2500 is based on SIP 2.0 (RFC 3261); any IP phone model based on the same protocol can work with the IPX-2200 and IPX-2500.

3.2 Before Making a Call

3.2.1 System Information

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

	Internet Telephony PBX System
Username	e;
Password	E
Language	e: English 💌
	Login



1. To login to the IPX-2200 or IPX-2500, your PC must use the same domain as the LAN IP address of the IPX-2200 or IPX-2500.

For security reason, please modify the user name and password after you login.
 You can modify it on this page: "System"---"Management"



3. Every time after saving the change, please press "Activate Changes" to make modification effective.

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

• Home	Home 🌣		···.				Logout Move the mouse over a field to see tooltips
▶ Operator			System Info				toonips
Basic	Network						
Inbound Control	WAN		IP: 192	.168.1.80	MAC: 00:30:4	:06:04:CE	
Advanced	LAN		IP: 192	.168.0.80	MAC: 00:30:4	FD:04:CE	
Network Settings	Storage						
Security	Disk Ext Disk		Total: Total:	13G N/A	Used: Used:	2.5G N/A	
Report	Slot Info		rotal.	19//2	Used.	190/3	
System	SLOT 1		SLOT				
	1 2 N/A N/A	3 4 N/A N/A	A N/A	2 N/A	3 N/A	4 N/A	
			Device Info				
	Model No.:	IPX-2200	System Ve	ersion:	2.1.4		
	Current Time:07/15/1	6 09:17			Run Time:7 d	ays, 16:39	

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



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
E	5 Advanced	Configuration of extension's default information,
5		Conference Call, Call Transfer, Function Key, etc.
6	Network	Configuration of Routing, Network, VPN, DHCP and other
0	Settings	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

▶ Home	Operator 🌣 Current Active		Exten	sions		
 Operator 	Current Active	: U 😑 Idle	🥚 Ringing 🛛 😑 InUs	e 🤨 Hold (UnAvailable	
Basic	800	801	802	0	03	804
Inbound Control	800 800 800 800 800 800 800 800 800 800	P) 🔮 801(SIP) 802(S	IP) 💙 8	03(SIP)	804(SIP)
Advanced	805 805(SII	e) ● 806 806(SIP)		08 08(SIP)	809 809(SIP)
Network Settings			VoIP 1			
Security	Status	Trunk Name	Type Username		:name/IP/Port	Reachability
Report			No VoIP Tru	nk defined.		
System			You can <mark>click here</mark>	to create Trunk.		
			FXO/GS	M Ports		
	Sta	tus Si	ignal Strength Ty	pe Poi	t	BLF Label
	Discor	inected		FXO	1	Channel1
	Discor	inected		FXO	2	Channel2
	(0K		FXS	3	
	(0K		FXS	4	
	Discor	inected		FXO	5	Channel5
	Discor	inected		FXO	6	Channel6
	Discor	inected		FXO	7	Channel7



Display all the Extension, VoIP Trunk and Slot information.

About extension:

1	۲	Idle
2	۲	Ringing
3	٠	In use
4	0	Hold
5	۲	Unavailable

3.2.3 Basic Configuration

Add new extensions

→ Home	Exter	nsio	ns							Move the mouse over a field to see tooltips
Operator	i i			Extensions		Upload	Download Ext	ensions		toonipa
Basic										
	Exte	ensi	on:	Search	Show /	All				
Trunks										
Outbound Routes	N	iew I	User B	atch Add Edi	t Selec	ted	Delete Selec	ted Delete All		
Inbound Control	Exte	ensio	ons							
Advanced			Name	Extension	Port	Protocol	DialPlan	Outbound CID	Options	_
Network Settings		1	800	800		SIP	DialPlan1		Edit	
Security		2	801	801		SIP	DialPlan1		Edit	
Report			802	802		SIP	DialPlan1		Edit	
System			803	803		SIP	DialPlan1		Edit	
System			804	804		SIP	DialPlan1		Edit	
			805	805		SIP	DialPlan1		Edit	
			806 807	806 807		SIP	DialPlan1 DialPlan1		Edit	
			808	807		SIP	DialPlan1 DialPlan1		Edit	
			809	809	-	SIP	DialPlan1		Edit	
				000		0	Brain fairt			

You can add more extensions one by one by clicking the "New User" button or bulk add extensions by clicking the "Batch Add" button.

	Batch Add	х
Extension Start: 810 DialPlan: DialPlan1	Extension End: 829 ▼ Password:(Random)	
	Save Cancel	



Field description

Item	Explanation
Extension	These two fields define the new extension range to be generated.
Start/Extension End	
Dial Plan	Select a same dial plan for these new extensions.
Password	Can be different random passwords consisting of numbers, letters
	and special characters (suggested) by checking the "Random"
	checkbox. Or you can specify the same password for all new
	extensions.

Other Extension Ranges

In Planet IP PBX system, we limited the user extension range within 800 and 899. If you want more extensions or you want the extensions in other ranges you need to change the extension range before you can add new extensions.

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

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

▶ Home	General			
 Operator 	General	Analog Settings	SIP Settings	IAX2 Settings
Basic				
Inbound Control				^
Advanced	Default Settings for Nev	w User		
Options	SIP: 🔽	IAX2: 🗌 Web	Manager: 🗌 🛛 Call V	Waiting: 🔽
 Virtual Fax 	Agent: 🗖 🛛 Void	cemail: 🔽 🛛 Del	lete VMail: 🔲 👘 VM Pas	ssword: 1234
 Voicemail 		nsport: UDP 🔻	SRTP:	
SMTP Settings	Audio Codecs	□ G.722 🗹 G.729 □ G.7		
Conferences		0.722 10 0.723 10 0.7		
 Music Settings 	Extension Preferences			
▶ DISA		User Extensions 8	00 _ 899	
Follow Me		Conference Extensions 9		-
Call Forward		IVR Extensions 6	10 _ 629	
One Number Stations		Queue Extensions 6	30 - 639	
 Paging and Intercom 		Ring Group Extensions 6	40 _ 659	_
Web Extensions	F	Paging Group Extensions 6		
PIN Sets		Web Extensions 6	80 - 699	-
Call Recording		Res	set	
Smart DID				
Callback		Save	Cancel	



Configure Extensions

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

[Basic] ---- [Extensions]

Ext	ensi	on:	Search	Show A	II				
N	lew l	Jser Bat	ch Add Edit	Select	ed	Delete Select	ted	Delete All	
xte	ensio	ons							
		Name	Extension	Port	Protocol	DialPlan	Outb	ound CID	Options
	1	800	800		SIP	DialPlan1			Edit
	2	801	801		SIP	DialPlan1			Edit
	3	802	802		SIP	DialPlan1			Edit
	4	803	803		SIP	DialPlan1			Edit
	5	804	804		SIP	DialPlan1			Edit
	6	805	805		SIP	DialPlan1			Edit
	7	806	806		SIP	DialPlan1			Edit
	8	807	807		SIP	DialPlan1			Edit
	9	808	808		SIP	DialPlan1			Edit
	10	809	809		SIP	DialPlan1			Edit

Total:10 30 ▼ Per Page Pages: << 1 ▼ >>

By default, 10 existing extension numbers have already been given. They are from 800 to 809.



Click [New User] to see the extension configuration interface as shown below:

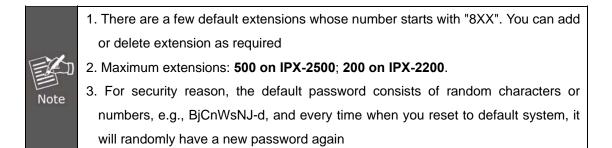
	Edit	
General		
SIP:	IAX2:	
Name: 800	Extension:	800
Password: BjCnWsNJ-d	Outbound CID:	
DialPlan: DialPlan1 -	Analog Phone:	None 👻
Voicemail		
Enable: 🔽	Password:	1234
Delete 🔲 VMail:	Email(Fax/Voicemail)	:
Other Options		
Web Manager:AgentAllow Being Spied:PickupMobility Extension:Mobility	p Group: 1	g: 🗸
VoIP Settings		
NAT: Transport	UDP 👻	SRTP:
DTMF Mode: RFC2833 -	Permit IP:	
Video Options		
Video Call: H.26	L H.263 H.263+ H	1.264
Audio Codecs		
g726 gsm speex • ««	alaw Jlaw 129	
Disallowed	Allowed	
	Save Cancel	

Extension Settings

Item	Explanation
SIP/IAX2	Choose extension protocol.
Name	Extension Name (English Character Only), e.g. Tom.
Extension	Extension Number connected to the phone, e.g. 888.
Password	Random password. (6-16 digits, e.g.123456)
Outbound CID	Override the caller ID when dialing out with a trunk.
Dial Plan	Please choose the Dial Plan which is defined in the menu "Outbound
Diai Pian	Routes".
Analog Phone	Please select the related FXS port for your analog phone.



Item	Explanation
Voicemail	Select this option to open the voicemail account
VM Password	Set password for Voicemail, e.g. "1234"
Delete VMail	Check this option to delete voicemail from system after it's sent to
	mail box.
Email	Extension user's mail box, which is used for receiving fax or
(Fax/Voicemail)	voicemail (you need to open the function to fax to email/voicemail),
	e.g. Tom@gmail.com
Web Manager	It's allowed to login Extension Management Panel to manage
	extension like voicemail, call recording, call transfer, etc. when you
	select this option.
Agent	Check this option to set this extension user as agent.
Call Waiting	Enable call waiting
Allowing Being Spied	Check this option to allow being spied.
NAT	Check this option if extension user or the phone is located after the
	NAT (Network Address Translation) available gateway.
Pickup Group	Select the Pickup Group which the extension user belongs to.
Mobility Extension	After checking this option, you must set mobility extension number.
	User can make calls to the IP PBX server with this mobility number,
	and have all rights of this extension, e.g. Outbound Call, Internal Call,
	Listen to the voicemail.
Transport	Select the Transport Protocol: UDP, TCP, TLS
SRTP	Enable SRTP
DTMF Mode	Default DTMF is rfc2833. It can be changed if necessary.
Video Call	Check to enable video call for this extension. And select the audio
	codecs you need to use.
Permit IP	Set computer permitted IP to visit this IP PBX, e.g.,192.168.1.77 or
	192.168.10.0/255.255.255.0. Computer with other IPs is not allowed
	to visit this IP PBX.
Audio Codec	Select what audio codec you need to use.





Upload/Download Extensions

Click [Upload/Download Extensions] to add extensions as shown below:

Upload/Download Extensions

	Extensions	Upload/Download Extension	ons
Upload Extensions			
	Please choose file to t	upload:	Browse
		Upload	
Download Extensio	ns Template		
	E	xtensions Template	
	Right Click her	re to Save as Template File (.csv))
	Right Click he	ere to Save as Template File (.txt)	
Download Extensio	ns(.csv)		
	D	ownload Extensions	

- Upload Extensions: Here you can upload .csv or .txt file to generate extensions.
- Download Extensions Template: Here you can download a template file in .csv or .txt format. Inside there are examples given, you can follow the examples to add your desired new extensions in the same format, and the new file can be used to upload to IP PBX system to generate new extensions.
- Download Extensions (.csv): Here you can download the existing extensions in the system for backup. The downloaded CSV file can be used for extension list recovery.



3.3 Outbound Call

3.3.1 Trunks

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

PLANE Hetworking & Community	T Inter		es Cancelled! hony PBX Syst	em (19X-2100)
▶ Home	VoIP Trunks			
 Operator 		VoIP Trunks	FXO/GSM Trunks	
Basic	and the second second			
Extensions	List of Trunks		New VoIP Trunk	
• Trunks	Provider Nam	e Type Hostnam	e/IP Username	Options
• Outbound Routes				
Inbound Control	No VoIP Trunk define	ed		
Advanced	Please click on 'New to add a Trunk	VoIP Trunk' button		
Network Settings				
Security				
Report				
System				

Planet IP PBX supports 3 kinds of trunks: VoIP Trunks, FXO Trunks and GSM Trunk.

VoIP Trunks

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

Ed	it SIP trunk trunk_2
Description:	VoIP
Peer Mode:	
Host:	192.168.1.21 :5060
Maximum Channels*:	0
Prefix:	
Outbound CID:	
🗆 Without Authentica	tion
Username: planettest	
Authuser: planet	
Password: •••••	
Advanced Options	
Fromdomain: 192.16	i8.1.21 Insecure: port, invite
Fromuser:	Qualify(sec): 🔽 2
DID Number:	Transport: UDP 💌
DTMF Mode: RFC28	333 💌 NAT: 🗖 SRTP: 🗖
Auto Fax Detection: 🗖	
Context: Default	🖌 Language: Default
Audio Codecs	
🗹 ulaw 🗹 alaw 🗌 G.1	722 🗖 G.729 🗖 G.726 💌 GSM 🗖 Speex
Video Codes	_
🗆 Н.261 🗆 Н.263 🗔 I	H.263+ 🗆 H.264



Planet IP PBX can register as a SIP user agent to a SIP proxy (provider). If you have subscribed VoIP service from ITSP, then with the account details given by them you can setup a VoIP trunk on Planet IP PBX system for the user extensions to share this trunk to make outbound phone calls.

Item	Explanation	
Description	Define the VoIP (figure or character).	
Protocol	Select protocol for outbound route, SIP or IAX2.	
Host	Set host address (provided by VoIP Provider).	
Maximum Channels	Set maximum channels for simultaneous call. (Only for outbound call;	
	"0" = no limitation).	
Prefix	The prefix will be added in front of your dialed number automatically	
	when the trunk is in use.	
Caller ID	This Caller ID will be displayed when user makes an outbound call.	
	Note: This function must be supported by local provider.	
Without	If you don't need the Authentication when connecting the IP PBX,	
Authentication	please check this option.	
User Name	User Name provided by VoIP Provider.	
Authuser	The optional authorization user for the SIP server	
Password	Password provided by VoIP Provider.	
Advanced Options	Advanced options for this trunk, e.g., codec, dial plan, etc.	
Domain	The domain is where you register your username.	
Insecure	Default value is "port, invite"; "port" Allow matching of peer by IP	
	address without matching port number; "invite" Do not require	
	authentication of incoming INVITEs.	
From User	Fromuser = yourusername; Many SIP providers requires this.	
Qualify (sec)	Asterisk sends a SIP OPTIONS command regularly to check that the	
	device is still online. Default value is 2 (sec).	
DID number	Self defined, it can be used to set up number DID.	
Transport	Default transport type for SIP messages	
DTMF Mode	Used to tell the system how to detect the DTMF (Dual Tone Multi	
	Frequency) key press. Choices are inband, rfc2833, or info. By	
	default, we use RFC2833.	
NAT	With this option enabled Asterisk may override the address/port	
	information specified in the SIP/SDP messages, and use the	
	information (sender address) supplied by the network stack instead.	



Item	Explanation
Context	Custom dial plan for this trunk, by default it's using the "default" dial
	plan. Configure only if this trunk is for branch office integration, so the
	calls coming from the other side can dial out from this IPPBX trunk
	directly. DO NOT change it unless you know how exactly this option
	works.
Language	You can choose a language here; the system will indicate the
	incoming calls from this trunk with the voice prompts you selected.
Audio Codecs	Select the audio codec/codecs the provider can support.
Video Codecs	If the ITSP supports video call, you can enable compatible video
	codecs here for video phone calls.



Except the configuration options related to the service provider and your account details, please do not change the trunk advanced parameters if you are not familiar with. After the SIP trunk is successfully added you can see it's listed here on this page

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

2) FXO/GSM Trunk

Click [FXO/GSM Trunk] -> [New FXO/GSM Trunk] :

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

FXO.	/FXS	/GSM	Ports

Status	Signal Strength	Туре	Port	BLF Label
Connected		FXO	1	Channel1
Connected		FXO	2	Channel2
Connected		FXO	3	Channel3
Connected		FXO	4	Channel4
Disconnected		FXO	5	Channel5
Connected		FXO	6	Channel6
Connected		FXO	7	Channel7
Connected		FXO	8	Channel8

To be able to use these lines connected on FXO ports to make phone calls, you have to use them to create trunk/trunks first. Navigate to web menu *Basic->Trunks->FXO/GSM Trunks*.



Click the "New FXO/GSM Trunk" button and you'll see available port numbers that can be used.

	Edit	Х
Description : Lines : Prefix :	FXO FXO: 1 2 3 4 25 6 7 28 Advanced Options	
Call Method:	Order Cycle 🔻	
Busy Detection:	Yes V Busy Count: 3	
Input Volume:	40% ▼ Output Volume: 40% ▼	
Call Progress:	No ▼ Progress Zone: US ▼	
Busy Pattern:	Language: Default	•
Answer on Polari		
Hangup on Polari Auto Fax Detection		
	Save Cancel	

Item	Explanation
Description	Define the description for this trunk (figure or character).
Lines	Available FXO and GSM ports.
Prefix	The numbers you dialed will first be manipulated by the dial rules,
	while the manipulated numbers reached the trunk before finally
	sending out to this prefix, which will be added to the numbers and
	then send out through this trunk. Usually you don't need this prefix.
	Please leave this field blank.
Call Method	If in this trunk you have more than 1 FXO/GSM port selected, then
	this parameter defines how to use these ports for outbound phone
	calls.
Busy Detection	Enable busy tone detection; it is also possible to specify how many
	busy tones to wait for before hanging up.
Busy Count	Specify how many busy tones to wait for before hanging up,
	configurable only if Busy Detection is enabled.
Input Volume	The volume of the calls from FXO channel/channels which have
	been received.
Output Volume	The volume of the calls from FXO channel/channels which have
	been made.
Call Progress	If turned on, call progress attempts to determine answer, busy, and
	ringing on phone lines. This feature is HIGHLY EXPERIMENTAL and
	can easily detect false answers so don't count on it being very
	accurate.



Item	Explanation	
Progress Zone	Progress zone also affects the pattern used for busy detection, only	
	effective when Call Progress is turned on.	
Busy Pattern	If busy detect is enabled, it is also possible to specify the cadence of	
	your busy signal.	
Language	You can choose a language here; the system will indicate the	
	incoming calls from this trunk with the voice prompts you selected.	
Answer on Polarity	For FXO (FXS signal) ports watch for a polarity reversal to mark	
Switch	when an outgoing call is answered by the remote party.	
Hang up on Polarity	In some countries, a polarity reversal is used to signal disconnect of	
Switch	a phone line. If the hang up polarity switch option is selected, the call	
	will be considered "hung up" on a polarity reversal.	

3) GSM Trunk

If you have ordered GSM modules for your IP PBX, the user extensions will be able to make and receive phone calls from the mobile network. You have to insert the SIM cards into the SIM slots of the GSM modules (Called IPX-21GS) and then install the modules to the IP PBX module slots. Antennas should be properly installed and placed in the open space for better signal reception. After this, power on the IP PBX and you'll be able to configure GSM trunks the same as you configure FXO trunks.

GSM Specifications

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

3.3.2 Outbound Routes

Outbound Routes enable you to tell Planet IP PBX which Trunks (phone lines) to use when people dial external telephone numbers. A simple installation will direct Planet IP PBX to send all calls to a single trunk. However, a complex setup could have an outbound route for emergency calls, another outbound route for local calls, another for long distance calls, and perhaps even another for international calls.

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



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

▶ Home	DialPlans	Move the mouse over a field to see tooltips
 Operator 	DialPlans DialRules	
Basic		
	List of DialPlans New DialPlan	
→ Trunks	Default DialPlan Name Rules Options	
Outbound Routes	Extensions, Spy, Conference, Ring I DialPlan1 Groups, IVR, Call Queues, Paging and Edit Delete	
Inbound Control	Intercom, Directory, DISA	
Advanced		
Network Settings		
Security		
Report		
System		

Dial rules

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

New DialRule	
Rule Name: Domestic	
PIN Set: 🖉 forDialRule 🔻 Record in CDR: 🗹	
Call Duration Limit: seconds Time Rule: Place this call through:	
fxo(FXO/GSM)	
Available Trunks Selected Trunks	
Custom Pattern: 9XXX. Z Any digit from 1 to 9 N Any digit from 2 to 9 X Any digit from 0 to 9 . Any number of additional digits belete 1 digits prefix from the front and auto-add digit before ialing	
Save Cancel	

Item	Explanation	
Rule Name	A name for this dial rule	
PIN set	A collection of PIN codes for granting outbound phone calls.	
Record in CDR	Record the PIN codes used for outbound phone calls along with the	
	user extension number and the dialed numbers.	
Call Duration Limit	Specify how long the calls can be made using this dial rule.	
Time Rule	Set a time condition when this dial rule can be used.	



Item	Explanation			
Available Trunks	All existing trunks in the IPPBX system.			
Selected Trunks	Trunk/Trunks can be used by this dial rule.			
Custom Pattern	Dial patterns act like a filter for matching numbers dialed with trunks.			
	The various patterns you can enter are similar to Asterisk's definition			
	of them:			
	X — Refers to any digit between 0 and 9			
	N — Refers to any digit between 2 and 9			
	Z — Any digit that is not zero. (e.g. 1 to 9)			
	. — Wildcard. Match any number of anything. Must match			
	something.			
Delete digits	The first blank is to strip some digit/digits before dialing out. Here you			
prefix from the front	need to fill in a count of number. The second blank is to prepend			
and auto-add	some digit/digits before dialing out. Here you need to fill in the exact			
digit	number to be added in front of the dialed number. For example a			
before dialing	user dialed 912345678 using the dial rule introduced above, the			
	prefix 9 at the first digit will be removed, and 00 will be added, so			
	eventually the user will call the number 0012345678.			

Dial plans

DialPlans

			DialPlans	DialRules	
List of	Dia	IPlans		New DialPlan	
Default DialPlan Name Rules Optio				Options	
V	1	DialPlan1	VoIP, Ring Groups, Call Queues, Paging and Intercom, IVR, Conferences, Extensions, DISA, Directory, Spy		Edit Delete

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



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

Edit	X
DialPlan Name: <u>DialPlan1</u> —Include External Calling Rules—	 Include Internal Calling Rules Ring Groups Call Queues Paging and Intercom IVR Conferences Extensions DISA Directory Spy
	Conferences Extensions DISA Directory

The calling rules in the left column are for external calls and calling rules in the right column are for internal calling. If you want to restrict some uses from calling out through some trunk lines or you don't want them to be able to call some internal destinations, you can create new dial plan by clicking the "New DialPlan" button.

New DialPlan	x
DialPlan Name: DialPlan2 Include External Calling Rules International Domestic Call Queues Paging and Intercom VIVR Conferences Extensions DISA Directory Spy	
Save Cancel	

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



3.4 Inbound Call

3.4.1 Inbound Routes

When a call is made from outside, you want to forward this call to an extension or IVR. This Chapter will introduce you how to deal with the inbound calls. The Inbound Control section is where you define how IP PBX system handles incoming calls. Typically, you determine the phone number that outside callers have called (DID Number) and then indicate which extension, Ring Group, Voicemail, or other destination to which the call should be directed.

→ Home	General			
 Operator 	General	Port DIDs	Number DIDs	DOD Settings
Basic				
Inbound Control	From FXO/GSM Cha	nnels		
Inbound Routes				
→ IVR	Distinctive Ring T	one:		
IVR Prompts	Destination:	Goto Time R	ule 👻 Time Rule Tir	meRule 👻
→ Call Queues				
Ring Groups				
• Black List	From VoIP Channels	;		
• Do Not Disturb				
• Time Based Rules	Distinctive Ring T	one:		
Advanced	Destination:	Goto Extens	ion 👻	-
Network Settings				
Security		Sa	ve Cancel	
Report				
System				

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] :

	New Po	rt DID	×
Port: Destination:	Goto Extension	Label:	
	Save	Cancel	

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

Number DIDs

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

Click [Number DID] -> [New Number DID] :

	New Number DID	×
DID Number: Destination:	Goto Extension 💌 800(800) 💌	
	Save Cancel	

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



DOD Settings

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

	New DOD	×
DOD Number: Destination:	Goto Extension 💌 800(800) 💌	
	Save Cancel	

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

3.4.2 IVR

IVR will improve office efficiency based on your requirement.

Please configure on this page [Inbound Control] -> [IVR] :

• Home	IVR					
 Operator 	List	of IVRs		New IVR		
Basic		Extension	Name	Dial other Extensions	Op	tions
Inbound Control	1	610	working time	Yes	Edit	Delete
 Inbound Routes 	2	611	closed time	No	Edit	Delete
• IVR						
• IVR Prompts						
Call Queues						
• Ring Groups						
• Black List						
• Do Not Disturb						
• Time Based Rules						



Click [New IVR] to create a new IVR:

	New IVR	Х
IVR	Settings	
Nar	ne: office-hours Extension: 612	
Weld	come Message	
Pleas	e Select: office_hours	
Repe	at Loops: 1 🔹	
	out: <u>0</u>	
Dial	other Extensions: (<u>Custom</u>)	
Кеур	ress Events	
Key	Action	
0	Goto Extension V 401(401)	*
1	Goto Ring Group 🔻 sales 🔻	
2	Goto Ring Group marketing	
3	Disabled •	
4	Disabled •	
5	Disabled 🔹	
6	Disabled 🔹	
7	Disabled 🔹	
8	Disabled 🔹	
9	Disabled 🔹	
*	Disabled 🔹	
#	Disabled 🔹	
t	Goto Extension • 401(401) •	
i	Goto Extension • 401(401) •	+

Item	Explanation		
Name	Set a name for the IVR		
Extension	Extension number for the IVR, by calling this number can		
	access the IVR menu.		
Please Select	Select a voice prompt for this IVR menu.		
Custom Prompts	Click this button to navigate to Inbound Control->IVR Prompts		
	page for new voice prompts.		
Repeat Loops	Define how many times to play the IVR menu to the caller.		
Timeout	Timeout for key pressing of each IVR loop.		
Dial Other Extensions	If enabled, the caller can dial extension number directly on IVR.		
Custom	By clicking "Custom" you can set dial plan for this IVR menu.		
	The callers on IVR would be able to dial some other		
	destinations the dial plan allows.(Not recommended)		
Key Press Events	Define which destination to go by pressing a key on the phone		
	keypad. If the undefined keys is pressed, then it will be handled		
	by the "i" parameter; "i" means invalid. And "t" stands for		



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

3.4.3 IVR Prompts

To configure IVR menu on IP PBX system you'll first need to record the IVR prompts. The IVR prompts will indicate the callers how to place their calls

▶ Home	IVR F	romp	its 🦃							
• Operator				IVR Promp	ts	Upload I\	/R Prom	ots		
Basic										
Inbound Control	List	of Pr	ompts 🌵			New Voice	Delete	Select	ted	
Inbound Routes			Name				Opt	tions		
→ IVR		1	closed.gs	m		Recor	d Again	Play	Delete	8
IVR Prompts		2	welcome.	.gsm			d Again			
Call Queues										
▶ Ring Groups										
Black List										
• Do Not Disturb										
• Time Based Rules										

Click [IVR Prompts] ---- [New Voice] to create new IVR prompt:

New Voice	
File Name:	office_hours
Format:	WAV (16-bit)
Extension used for recording	800
Record Ca	ancel

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



If you want to hear the prompt, please click [Play] :

Play record voice					
Extension used for playing: 800 🗸					
Play Cancel					

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

Upload IVR prompt

→ Home	Upload IVR Prompts				
 Operator 		IVR Prompts	Upload IVR Prompts		
Basic					
Inbound Control		Upload	d IVR Prompts		
Inbound Routes	Note: The sou		/av(16bit/8000Hz/Mono), gsm	, ulaw or alaw!	
• IVR	The size is limited in 15MB!				
IVR Prompts	Pleas	se choose file to uploa	d: Brow	se	
Call Queues					
• Ring Groups	Upload				
• Black List					
• Do Not Disturb					
• Time Based Rules					



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

3.4.4 Call Queue

A call queue places incoming calls in line to be answered while extension users are busy with other calls. The queued calls are distributed to the next available extension user in the order received. After they have been created, they can be assigned to specific extensions and configured to feature greetings, messages, and hold music.



There are 3 existing call queues. All you have to do is click the "Edit" button to configure them.

If you want more call queues, you can click "New Call Queue" to add more queues.

	New
Call Queue Reference:	
Queue Number: 633 Ring Strategy: Random •	Label:
	ave any users defined as agents! <mark>here t</mark> o manage users.
Queue Options:	Announcements:
Agent TimeOut(sec): <u>15</u> Auto Pause: Wrap-Up-Time(sec): <u>10</u>	Announcements: Caller Position Announcements Frequency(sec): 30 Announce Hold Time: No
Agent TimeOut(sec): 15 Auto Pause:	Caller Position Announcements Frequency(sec): 30 Announce Hold Time: No Periodic Announcements Repeat Frequency(sec): 0
Auto Pause: Wrap-Up-Time(sec): 10 Max Wait Time(sec): Max Callers: 8	Caller Position Announcements Frequency(sec): 30 Announce Hold Time: No Periodic Announcements

Here we can see in the "Agents" field there're no available agents to be assigned to the call queues. Click "click here" you'll be redirected to the extension page to determine which extensions will be employed as call queue agents.

Tick the checkbox of the extension numbers which will be employed as call queue agents, then click the "Edit Selected" button and tick the "Agent" option in "Other Options" section.

Other Options			
Web Manager:		✓Agent:	
Pickup Group:	1		



Save and go back to *Inbound Control->Call Queues* page again and configure the existing call queues and add new call queues with available agents.

	Edit
Call Queue Reference:	
Queue Number: 630 Ring Strategy: Random • Agents:	Label: support
Queue Options:	Announcements:
Agent TimeOut(sec): 15 Auto Pause: Wrap-Up-Time(sec): 10	Announcements: Caller Position Announcements Frequency(sec): 30 Announce Hold Time: Yes
Agent TimeOut(sec): 15 Auto Pause:	Caller Position Announcements Frequency(sec): <u>30</u>
Agent TimeOut(sec): 15 Auto Pause: Wrap-Up-Time(sec): 10 Max Wait Time(sec): Max Callers: 8 Join Empty:	Caller Position Announcements Frequency(sec): 30 Announce Hold Time: Yes ▼ Periodic Announcements Repeat Frequency(sec): 0
Auto Pause: Wrap-Up-Time(sec): <u>10</u> Max Wait Time(sec): Max Callers: <u>8</u>	Caller Position Announcements Frequency(sec): 30 Announce Hold Time: Yes Periodic Announcements

Item	Explanation
Queue Number	Define an extension number to identify the queue.
Label	Define the label for the queue. A user can be agent of multiple
	queues, by giving a label for the call queue; if an incoming call
	is distributed to an agent the label will be displayed on the
	phone screen along with the caller ID. So a call queue agent
	can tell from which call queue the call is coming from.
RingAll	Ring all available agents until one answers (default).
RoundRobin	Starting with the first agent, ring the extension of each agent in
	turn until the call is answered.
LeastRecent	Ring the extension of the Agent who has least recently received
	a call
FewestCalls	Ring the extension of the Agent who has taken the fewest
	number of calls.
Random	Ring the extension of a random Agent.
RRmemory	RoundRobin with Memory, like RoundRobin above, except
	instead of the next call starting with the first agent, the system



Item	Explanation
	remembers which extension was called last and begins the
	round robin with the next agent.
Agent	Check each agent that is to be a member of this specific Call
	Center Queue.
Agent TimeOut (sec)	Specify the number of seconds to ring an agent's extension
	before sending the call to the next Agent (based on Ring
	Strategy)
Auto Pause	If an Agent's extension rings and the Agent fails to answer the
	call, automatically pause that agent so the stop receiving calls
	from the queue.
Wrap-Up-Time (sec)	This is the amount of time in seconds that an agent has to
	complete work on a call after the call is disconnected. (Default
	is 0, which means no wrap-up time.)
Max Wait Time (sec)	Calls that have been waiting in the queue for this number of
	seconds will be sent to the "If not answered" destination.
Max Callers	Max number of the callers who are allowed to wait in the queue.
	(Default is 0, which means no limitation.) With this number of
	callers in the queue already, subsequent callers will be sent to
	the "If not answered" destination.
Join Empty	Allow callers to enter the Queue when no Agents are available.
	If this option is not defined, callers will not be able to enter
	Queues with no available agents - callers will be sent to the "If
	not answered" destination.
Leave When Empty	If this option is selected and calls are still in the queue when the
	last agent logs out, the remaining callers in the Queue will be
	transferred to "If not answered" destination. This option cannot
	be used with Join Empty simultaneously.
Auto Fill	Callers will be distributed to Agent automatically.
Report Hold Time	Report the hold time of the next caller for Agent when the Agent
	is answering the call.
Frequency(sec)	Repeat frequency to announce the hold time for callers in the
	Queue.("0" means no announcement).
Announce Hold Time	Announce the hold time. Announce (yes), do not announce (no)
	or announce once (once), it will not be announced when the
	hold time is less than 1 minute.
Repeat Frequency(sec)	Interval time to play the voice menu for callers.("0" mean not to



Item	Explanation
	play).
Announcement Prompt	Select a prompt as the Announcements Prompt from the IVR
	Prompts.

3.4.5 Ring Groups

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

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

lame: sales	Strategy:	RingAll 🔻	
403(SIP) 403	*		
404(SIP) 404		402(SIP) 402	1
405(SIP) 405	-	407(SIP) 407	
406(SIP) 406		408(SIP) 408 409(SIP) 409	
		410(SIP) 409	
		411(SIP) 411	
	*	412(CID) 412	-
Ring Group Mem	bers	Available Cha	nnels
	Label:		
Extensi	on for this ring g	roup: 640	
Ring (each/all)	for lasting time	(sec): 20	
f not answered			
Goto Extension			
Goto Voicemail			
Goto Ring Group			
Goto IVR			
Hangup			

Item	Explanation	
Name	Define a name for the Ring Group.	
Strategy	Define how to ring the group members; select "RingAll" will ring	
	all the member extensions at the same time, select "Ring In	
	Order" will ring the member extensions one by one.	
Ring Group Members	The extensions selected to be the members of the ring group.	
Available Channels	All available extensions/channels can be added to the ring	
	group.	
Label	The extensions can be members of multiple ring groups, by	



Item	Explanation	
	giving each ring group a different label, if an incoming call rings	
	a ring group the label will be displayed on the phone screen	
	along with the caller ID. So a ring group member can tell from	
	which ring group the call is coming in.	
Extension for this ring	By calling this extension can reach the ring group members	
group		
Ring(each/all) for lasting	Ring duration of the group members.	
time(sec)		
If not answered	Setup a destination to redirect the incoming calls to, if no one	
	answers.	

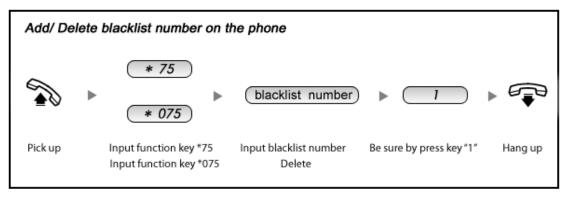
3.4.6 Black List

Before call spy can work, you have to make sure the extensions to be spied on have the "Allow Being Spied" option enabled on extension settings page.

If some numbers need to be blocked, you can use this functionality. Please configure it here: Click [Inbound Control] -> [Blacklist] -> [New Blacklist]

	New Bla	cklist	х
Blacklist	Number:		
[Save	Cancel	

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:





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.	

Reference Parameters and Explanation of the Blacklist:

3.4.7 Do Not Disturb

Do Not Disturb

Enable Do Not Disturb: *74 Disable Do Not Disturb: *074

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

3.4.8 Time-based Rules

For the companies and shops, they all have their own business hours and non-business hours. Routing the incoming calls by proper time conditions is much more reasonable.

Please set from this page: [Time-based Rule] --- [New Time Rule] :

Edit	х
Rule Name: <u>TimeRule</u>	
Time & Date Conditions	
Start Time: 09 💙 : 00 💙 End Time: 18 💙 : 00 🌱 Start Day: Mon 🌱 End Day: Sun 🌱 Start Date: 01 🌱 End Date: 31 🜱 Start Month: Jan 🌱 End Month: Dec 🌱	
Destination	
if time matches: IVR working time if time unmatches: IVR closed time Save Cancel	

New Time Rule:

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



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

3.5 Advanced

3.5.1 Options

General

General	Analog Settings	SIP Settings	IAX2 Settings
---------	-----------------	--------------	---------------

3.5.1.1 General

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

Local Extension Settings		
	Operator Extension: Global Ring Time Set(sec): 30 Enable Transfer: ♥	
	Enable Attended Transfer Caller ID: Enable Music On Ringback:	
	Auto-Answer: 🗹 Fax Detect Time: 1 🔻 Web Dial Auto-Answer: 🗐	
	Record Format: GSM Call Forward CID:	
	P-Preferred-Identity: 🔲	

Item	Explanation	
Operator Extension	Choose an extension to be operator extension. While an	
	incoming call had been directed to voicemail, by pressing '0' the	
	caller can get to operator extension.	
Global Ring Time	If not specifically configured, the incoming call will ring the	
Set(sec)	extension for the time given here.	
Enable Transfer	If enabled, the extension users will be able to do call transfer.	
Enable Attended Transfer	Normally if you use feature code *2 to transfer a call to another	
Caller ID	extension, the extension user only sees your extension number	
	as caller ID but not the actual caller ID, by enabling this option	
	the real caller will be passed to the user extension.	



Item	Explanation	
Enable Music On	If enabled this option, callers will hear music instead of ringback	
Ringback	tone while calling other extensions.	
Auto-Answer	Auto answer enables the IPPBX to automatically answer the	
	inbound calls from analog ports.	
Fax Detect Time	If auto answer enabled, you are able to configure the fax auto	
	detection time here.	
Web Dial Auto-Answer	Enable/disable auto answer of the extension numbers while	
	dialing from Web GUI.	
Record Format	Choose GSM or WAV as the call recording format.	
Call Forward CID	Allow passing the real caller ID to the forwarded number.	
P-Preferred-Identity	The P-Preferred-Identity header is used among trusted SIP	
	entities (typically intermediaries) to carry the identity of the user	
	sending a SIP message as it was verified by authentication.	

Default Settings for New User

Default Settings for New User				
Audio Codec	Fransport: UDP ▼ S	Web Manager: Delete VMail: SRTP: G.726 GSM Spe		

In this section the options are for new extensions. If you have one of the options enabled, then the newly created extensions will all have this option enabled.

Extension Preferences

Extension Preferences			
User Extensions 800	to 899		
Conference Extensions 900	to 909		
IVR Extensions 610	to 629		
Queue Extensions 630	to 639		
Ring Group Extensions 640	to 659		
Paging Group Extensions 660	to 679		
Web Extensions 680	to 699		
Reset			

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



according to your real applications.

3.5.1.2 Analog Settings

Analog Settings are used for configuring the IP PBX system working seamlessly with your telephone lines from the telecom.

Caller ID Detect

• Home	Analog Settings
 Operator 	General Analog Settings SIP Settings IAX2 Settings
Basic	
Inbound Control	Caller ID Detect
Advanced	Caller ID Detection: 🔽
Options	Caller ID Signaling: Bell-US -
 Virtual Fax 	Caller ID Start: Ring -
 Voicemail 	CID Buffer Length: 2500 v
 SMTP Settings 	Ring Debounce: 64 🔻
Conferences	DTMF Hits Begin: 2 💌
 Music Settings 	DTMF Misses End: 3 💌
• DISA	Detect Caller ID After: 2 💌
Follow Me	

These options are used to teach the IP PBX system how to detect caller identity (caller ID) from the PSTN lines on FXO ports.

Item	Explanation	
Caller ID Detection	Enable/Disable Caller ID Detection	
Caller ID Signaling	The signaling type applied on the PSTN lines to pass caller ID.	
	Bell-US—Also known as BellcoreFSK. Used in the Canada,	
	China, Hong Kong and US.	
	DTMF—Dual Tone Multi-Frequency. Used in Denmark, Finland	
	and Sweden.	
	V23—Mostly used in UK.	
	V23-Japan—Mostly used in Japan.	
Caller ID Start	When the caller ID starts.	
	Ring—Caller ID starts when a ring received.	
	Polarity—Caller ID starts when polarity reversal starts.	
	Polarity (India)—Can be used in India.	
	Before Ring—Caller ID starts before a ring received	
CID Buffer Length	The buffer length can be used to store caller ID info.	



General

General	
Opermode: FCC	
Tone Zone: China 🔹	
Ring Timeout(s): 8	
Relax DTMF: 🔲	
Send Caller ID After: 1 🔻	
Echo Cancel: 🗹	
Echo Training: <u>no</u> (yes/no/number)	

Item	Explanation	
Opermode	Set the Opermode for FXO Ports	
ToneZone	Select the tone zone of your country.	
Ring Timeout(s)	FXO (FXS signaled) devices must have a timeout to determine	
	if there was a hangup before the line was answered. This value	
	can be tweaked to shorten how long it takes before DAHDI	
	considers a non-ringing line to have hung up.	
Relax DTMF	Relax DTMF	
Send Caller ID After	Some countries (UK) have ring tones with different ring tones	
	(ring-ring), which means the caller ID needs to be set later on,	
	and not just after the first ring, as per the default (1).	
Echo Cancel	Enable/Disable software Echo Cancel algorithm.	
Echo Training	Enabling echo training will cause the PBX system to mute the	
	channel, send an impulse, and use the impulse response to	
	pre-train the echo canceller so it can start out with a much	
	closer idea of the actual echo. Value may be "yes", "no", or a	
	number of milliseconds to delay before training (default = 400).	
	This option does not apply to hardware echo cancellers.	

3.5.1.3 SIP Settings

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



General	Analog Settings	SIP	Settings	IAX2 Settings
General				
		UDP Port:	5060	
	🗹 Enable	TCP Port:	5060	
	🗖 Enable	TLS Port:	5061	
	Start	RTP Port:	10001	
	End RTP Port:		10500	
	DTMF Mode:		Auto 💙	
Allow Guest:				
Max Registration/Subscription Time(sec):		3600		
Min F	Min Registration/Subscription Time(sec):		60	
Default Incoming/Outgoing Registration Time(sec):		60		

Item	Explanation	
UDP Port to bind to	SIP standard port is 5060	
TCP Port	Default TCP port is 5060	
TLS Port	Default TLS port is 5061	
Start RTP Port	RTP port range	
End RTP Port	RTP port range	
DTMF Mode	Set default DTMF mode for sending DTMF, support auto,	
	RFC2833, inband, info. Default: RFC 2833	
Allow Guest	This setting determines if anonymous callers are	
	permitted to place calls to the IP PBX system. For	
	security precautions please do not enable this option.	
Max Registration/Subscription	Maximum duration (in seconds) of incoming	
Time	registrations/subscriptions is 3600 seconds by default	
Min Registration/Subscription	Minimum duration (in seconds) of	
Time	registrations/subscriptions is 60 seconds by default	
Default Incoming/Outgoing	Default duration (in seconds) of incoming/outgoing	
Registration Time	registration	



NAT Support

External IP:	210.61.134.91
External Host:	210.61.134.91
External Refresh(sec):	10
Local Network Address:	192.168.1.0/255.25
Local Network Address:	
Local Network Address:	

Item	Explanation
External IP	Address that we're going to put in outbound SIP
	messages if we're behind a NAT
External Host	Alternatively, you can specify an external host, and
	Asterisk will perform DNS queries periodically. Not
	recommended for production environments! Use external
	IP instead
External Refresh	How often to refresh external host if used. You may
	specify a local network in the field below
Local Network Address	192.168.1.0/255.255.255.0' : All RFC 1918 addresses are
	local networks, '10.0.0.0/255.0.0.0' : Also RFC1918,
	'172.16.0.0/12' : Another RFC1918 with CIDR notation,
	'169.254.0.0/255.255.0.0' : Zero conf local network

T.38 Fax Passthrough Support

T.38 Fax (UDPTL) Passthrough: 📃

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



Type of Service			
TOS for Signaling packets:	CS3	•	
TOS for RTP audio packets:	ef	•	
TOS for RTP video packets:	AF41	•	
COS Priority for Signaling packets:	3 🔻		
COS Priority for RTP audio packets:	5 🔻		
COS Priority for RTP video packets:	4 🔻		
DNS SRV Look Up:			
Relax DTMF:	v		
RTP TimeOut(sec):			
RTP Hold TimeOut(sec):			
Add 'user=phone' to URI:			
User Agent:	VOIP		
Premature Media:			
Progress Inband:	Never	•	

Item	Explanation
TOS for Signaling packets	Sets Type of Service for SIP packets
TOS for RTP audio packets	Sets Type of Service for RTP audio packets
TOS for RTP video packets	Sets Type of Service for RTP video packets
COS Priority for Signaling	Sets 802.1p priority for SIP packets.
packets	
COS Priority for RTP audio	Sets 802.1p priority for RTP audio packets.
packets	
COS Priority for RTP video	Sets 802.1p priority for RTP video packets.
packets	
DNS SRV Look Up	Enable DNS SRV lookups on outbound calls.
Relax DTMF	Relax DTMF handling.
RTP TimeOut(sec)	Terminate call if there is 60 seconds of no RTP or RTCP
	activity on the audio channel when we're not on hold. This
	feature enables the ability to hangup a call in the case of
	a phone disappearing from the network, for instance if the
	phone loses power.
RTP Hold TimeOut(sec)	Terminate call if 300 seconds of no RTP or RTCP activity
	on the audio channel when on hold.
Add 'user=phone' to URI	Enable this option if the SIP provider requires



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

Register TimeOut(sec):

Register Attempts:

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



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

3.5.1.4 IAX2 Settings

• Home	IAX2	Settings			
Operator		General	Analog Settings	SIP Settings	IAX2 Settings
Basic					
Inbound Control	Gen	eral			
Advanced				UDP Port: 4569	
Options					•
• Virtual Fax			Max Registration/Subscriptio	on Time(sec): 1200	
 Voicemail 			Min Registration/Subscriptio		
SMTP Settings					
Conferences					
 Music Settings 	—		Save	Cancel	

Item	Explanation
UDP Port	IAX2 signaling and media port, default is 4569.



Item	Explanation
Bandwidth	Specify bandwidth of low, medium, or high to control
	which codecs are used in general.
Max Registration/Subscription	Maximum amounts of time that IAX peers can request as
Time (sec)	a registration expiration interval (in seconds).
Min Registration/Subscription	Minimum amounts of time that IAX peers can request as
Time (sec)	a registration expiration interval (in seconds).

3.5.2 Virtual Fax

Virtual Fax

Virtual Fax		
	Enable:	
	Country Code:	886
	Area Code:	2
	Outbound CID:	22199518
	Label:	Planet
	Fax Seat:	4 🗸
	DialPlan:	DialPlan1 -
		Save Cancel

Item	Explanation
Enable	Enable the following settings for outbound fax.
Country Code	Enter your country code here. (Optional).
Area Code	Enter your Area Code here. (Optional)
Outbound CID	Only works if the outbound fax goes out through VoIP
	trunks. The other side receives your fax with this number.
Label	Some custom information to be printed to the header of
	the fax pages.
Fax sent	Defines how many users can send fax at the same time.
DialPlan	A proper dial plan to send faxes.



3.5.3 Voicemail

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

General

	General	Email Settings
/oiceMail Reference		
Max Greetin Dial "0" for	g Time(sec): Operator:	<u>30</u> ▼
Voice Message Options		
Message Fo	rmat:	WAV (16-bit) 🗸
Maximum Me	essages:	100 👻
Max Messag	je Time(min):	2 👻
Min Messag	e Time(sec):	2 🗸
Playback Options	. ,	
. ayback options		
		essage CallerID
		essage Duration
	💷 Play Er	nvelope

	Allow	Users	to	Review
--	-------	-------	----	--------

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



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

3.5.4 SMTP Setting

SMTP Settings

SMTP Settings:	
SMTP Server: Port: 25 SSL/TLS: Venable SMTP Authentication Username: Password: Send Test	_
Save 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.
	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	If your SMTP server needs Authentication, please enable SMTP
Authentication	Authentication, and configure the following information.
User Name	Input user name of your email box.
Password	Input password of your email box.

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

Send Test	Х
Email Address:	
Send Cancel	



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

3.5.5 Conference

Home

If you want to create a conference room for some extension users or with external lines, you can input conference room number 900, input conference room password 1234 (Admin's password is 2345), then enter the conference room. IPX-2200 and IPX-2500 support 3 conference rooms. Please configure it on this page [Conference] :

Basic Default Extension Guest Password Administrator Password Options Options Options Virtual Fax Virtual Fax Vicemail SMTP Settings Conferences Music Settings Onlow Me Call Forward Onlow Number Stations PIN Sets Call Recording Smart DID Call Recording Smart DID Call Recording Feature Codes 	Home							
Inbound Control Advanced Advanced 2 901 1234 2345 Edit Delete > Options 2 901 1234 2345 Edit Delete > Options 3 902 1234 2345 Edit Delete > Virtual Fax . . Voicemail .<	▶ Operator	Conferences		New Conference				
Advanced 1 900 1234 2345 Edit Delete • Options · · 3 902 1234 2345 Edit Delete • Voicemail · · 3 902 1234 2345 Edit Delete • Voicemail · <td< th=""><th>Basic</th><th>Default</th><th></th><th>Extension</th><th>Guest Password</th><th>Administrator Password</th><th>C</th><th>Options</th></td<>	Basic	Default		Extension	Guest Password	Administrator Password	C	Options
 Options Vitual Fax Voicemail SMTP Settings Conferences Music Settings One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Inbound Control	V	1	900	1234	2345	Edit	Delete
 Virtual Fax Voicemail SMTP Settings Conferences Music Settings DISA Follow Me Call Forward One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Advanced		2	901	1234	2345	Edit	Delete
 Voicemail SMTP Settings Conferences Music Settings DISA Follow Me Call Forward One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Options		3	902	1234	2345	Edit	Delete
 SMTP Settings Conferences Music Settings DISA Follow Me Call Forward One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Virtual Fax				1			
 Conferences Music Settings DISA Follow Me Call Forward One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	 Voicemail 							
 Music Settings DISA Follow Me Call Forward One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	SMTP Settings							
 DISA Follow Me Call Forward One Number Stations Paging and Intercom Web Extensions YEN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Conferences							
 Follow Me Call Forward One Number Stations Paging and Intercom Web Extensions Web Extensions Orall Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	 Music Settings 							
 Call Forward One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	→ DISA							
 One Number Stations Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Follow Me							
 Paging and Intercom Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Call Forward							
 Web Extensions PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	• One Number Stations							
 PIN Sets Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	Paging and Intercom							
 Call Recording Smart DID Callback Phone Book LDAP Server Feature Codes 	• Web Extensions							
 Smart DID Callback Phone Book LDAP Server Feature Codes 	• PIN Sets							
 Callback Phone Book LDAP Server Feature Codes 	Call Recording							
Phone Book LDAP Server Feature Codes	• Smart DID							
LDAP Server Feature Codes	Callback							
Feature Codes	• Phone Book							
	LDAP Server							
Phone Provisioning	Feature Codes							
- Hone Hovisioning	Phone Provisioning							

Conferences



	Edit	х
Conference Number		
Room Exten	sion:	900
Conference Password		
Guest Pass	word:	1234
Administrate	or Password:	2345
Conference Options		
Conference DialPlan	Internal 👻	
	Play hold music	for first caller
	Enable caller me	enu
	Announce caller	s
	Record conferen	ice
	Quiet Mode	
	Close the confer	rence when last administrator exits
	Leader Wait	
	Save C	ancel

Item	Explanation
Room Extension	By calling this extension number to enter the conference room
Guest Password	If the callers use this password to enter the conference then
	they are ordinary participants
Administrator Password	If the callers use this password to enter the conference then
	they are administrators, they have advanced conference menu
	for example inviting people to participate the conference.
Conference DialPlan	Conference admin can use this dial plan to invite other
	participants.
Play hold music for first	Play the hold music for the first participant in the conference
caller	until another participant enters in this conference.
Enable caller menu	Check this option to allow the conference admin to access the
	conference menu by pressing "*" on the phone.
Announce Callers	Announce all the participants in the room that new participant is
	coming in.
Record Conference	Record this conference. (Recording format is wav.) The
	recorded conference can be searched from Report->Record
	List->Conference page.



Item	Explanation		
Quiet Mode	If check this option, system will not give any announcement		
	when the participants enter or leave the conference		
Close the conference	If checked this option, the conference will be terminated when		
when last administrator	the last administrator exits		
exits			
Leader Wait	Wait until the conference leader(administrator) enters the		
	conference before starting the conference		

Please check the following diagram to learn:

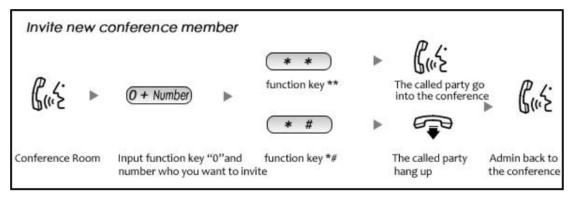
Go to conference:

Ready to	go into	conference				
\$	►	900	•	1234	•	Guiz
Pick up		Input conference room number: 900		Input conference room password: 1234		Go into the conference room

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

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

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





3.5.6 Music Settings

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

Click [Music Settings] to display the dialog shown below:

Music Settings:

Music Settings							
	Music Settings	Music Management					
Music On Hold Ref	Music On Hold Reference						
	Music:	Music 1 👻					
Music On Ringbac	k Reference						
	Music:	Music 2 👻					
Music On Queue R	eference						
	Music:	Music 3 👻					

Please define different music files for different music folders.

Music Management:

Music Management				
	Music Settings	Music Managem	ent	
Music Management				
	Select Music Directory: Files:	Music 1 🔻	Load Delete	
Upload Music File				
	Select Music Directory Music 1 -			

Select Music Directo Note: The sound file must be mp3, wa The size is	-	/Mono), gsm, ulaw or alaw	!
Please choose file to upload	:	Browse	
Up	load		



Item	Explanation	
Select Music Directory	Load music in the music file.	
File	Display music name under the music file. You can delete it.	
Select Music Directory	Select the file where you want to save your uploaded music.	
Please choose file to upload	Select the music you want to upload. Note: music file must be	
	MP3, WAV (16bit/8000Hz/Mono), GSM, ulaw or alaw, and less	
	than 15MB.	



The sound file must be MP3, wav (16bit, 8000Hz, mono), gsm, ulaw and alaw audio file format. The size is limited to **15MB.**

3.5.7 DISA

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

New DISA	
Name:	
PIN Set:	✓ Without PIN
Record in CDR:	
Response Timeout(sec): 10	
Digit Timeout(sec): 5	
Extension for this DISA(Optional)	:
Allow Outbound Route	1

Select DialPlan DialPlan1 -

Item	Explanation	
Name	Define a name for DISA.	
PIN Set	A set of PIN codes to authorize the callers using the system features and facilities.	
Without PIN	If enabled, the callers will not be required to enter any PIN code to be able to use the system features can facilities (Not recommended).	
Record in CDR	The PIN code that has been used will be stored into call logs	



Item	Explanation
	which can be traced on <i>Report->Call Logs</i> page.
Response Timeout (sec)	The maximum time for waiting before hanging up if the dialed
	number is incomplete or invalid. Default is 10 seconds
Digit Timeout (sec)	The maximum interval time between digits when typing extension
	number is 5 seconds by default.
Extension for this DISA	If you want to access DISA by dialing an extension, you can
(optional)	define an extension number for this DISA.
Select Dial Plan	Select a dial plan for this DISA so the callers will be able to make
	outbound phone calls using the trunks on the IP PBX system.

3.5.8 Follow Me

The Follow Me feature allows you to create a more specialized method of routing calls that are sent to a specific extension. Using this module, you can cause a call to an extension to ring several other extensions, or even external phone numbers. So the inbound calls can ring all the numbers which can possibly find you.

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

	New Follow Me	x	
	Extension: 800 (800)		
Follow Me List:	802,20 91558888878,20 922199518,20		
	Save Cancel		

Item	Explanation	
Extension	Select the extension number which will be configured with follow	
	me.	
Ring lasting for <u>20</u> seconds	Define how long to ring the extension before the call is forwarded	
	out. By default 20 seconds.	
Follow Me List	The list of numbers to forward the calls to. Each line is written	
	with the format "number, time", "number" is one of the number to	
	forward the calls to, "time" defines how long to ring this number,	
	they are separated with a comma without space. The order of	



	ringing the	ese numbers are the order	you writing in this column	
Follow Me Options	i			
	Follow Me	Follow Me Options		
Follow Me Options				
Playback the incoming	i status message pric	or to starting the follow-me step)(sec).	
Record the caller's name so it can be announced to the callee on each step.				
Playback the unreachable status message if we've run out of all steps or the callee was set not to be reachable.				
Always take the call				
		Save		

3.5.9 Call Forward

3.5.9.1 Configure From the Web

This feature allows calls to an extension to be automatically forwarded to a specific internal extension or external phone number. Before configuring call forward you can enable the IP PBX system to play a voice prompts before the call was forwarded out. This voice prompts can be recorded or uploaded from the *Inbound Control->IVR Prompts* page. Once the voice prompts file is ready you can navigate to web menu *Advanced->Call Forward*. And enable the system to play back the voice prompts before the incoming call was forward out.

Forward Prompt		
	Enable: 🗷 Please Select:	welcome 🔹
	Save	Cancel



After the voice prompt is set, you can click the "New Forward" button to set call forward for an extension.

New Forward	x
Extension: 800800	
Always 922199518	
Busy	
No Answer	
Ring lasting for seconds	
Save Cancel	

Item	Explanation
Always	Unconditionally forward the incoming calls.
Busy	Forward the incoming calls only if the extension is busy.
No Answer	Forward the incoming call only if the extension didn't answer.
Ring lasting for seconds	Only is call forward on "No Answer" this option is available to be configured. It defines how long to ring the extension before forwarding.



1. If you forward a call to an external phone number please make sure to add a prefix in front of the number if your system requires prefix to dial out.

2. The forward condition "Always" is mutually exclusive to "Busy" and "No Answer".

3.5.9.2 Configure From the Phone

Navigate to web menu Advanced->Feature Codes.

You'll see feature codes for call forward as follows:

Call Forward

Enable Forward All Calls: *71 Disable Forward All Calls: *071 Enable Forward on Busy: *72 Disable Forward on Busy: *072 Enable Forward on No Answer: *73 Disable Forward on No Answer: *073

With these feature codes, you can activate or deactivate call forward directly from your phones without the need to configure on the Web GUI. For example, the IP PBX requires prefix 9 to call outbound, and the number you want to forward the calls to is 86547096. Activate always call forward: Dial *71986547096, press 1 to confirm. Deactivate always call forward: Dial *071.



Activate call forward on busy: Dial *72986547096, press 1 to confirm.

Deactivate call forward on busy: Dial *072.

Activate call forward no answer: Dial *73986547096, press 1 to confirm.

Deactivate call forward no answer: Dial *073.

3.5.9.3 Call Transfer

Call Transfer is used to transfer a call in progress to some other destination. There are two types of call transfer.

- Attended call transfer Where the call is placed on hold, a call is placed to another party, and a conversation can take place privately before the caller on hold is connected to the new destination. It is also called "Supervised Call Transfer".
- Blind call transfer Where the call is transferred to the other destination with no intervention (the other destination could ring out and not be answered for instance).

Navigate to web menu *Advanced->Feature Codes*. You'll see the feature code for call transfer as below:

Transfer

Blind Transfer: # Attended Transfer: *2 Disconnect Call: *

Timeout for answer on attended transfer(sec): 15

Item	Explanation
Blind Transfer	In a live call, extension user can press # key and the IP PBX
	system prompts "Transfer", then you enter the number to be
	transferred to. This call will be transferred instantly and the user
	can hang up. If the transferred number didn't answer this call it
	will ring back to the extension user.
Attended Transfer	In a live call, extension user can press *2 and the IPPBX system
	prompts "Transfer", then you enter the number to be transferred
	to. After he/she answered your call, you can introduce this call
	and hang up, and then the call is transferred.
Disconnect Call	In an attended transfer if the other side doesn't want to take the
	call to be transferred, you can press * to disconnect with him/her
	and get back to the caller.
Timeout for answer on	In an attended transfer if the third party rings for 15 seconds
attended transfer(sec)	without answering, the extension user will go back to the caller
	and the transfer will be terminated.



3.5.10 One Number Stations

One number stations is an innovative IPPBX feature provided by Planet only. With one number stations feature, you can have the same extension number in several different locations. One number stations feature can put several extension numbers in the same "group", a main number can be selected from the members, when there's an incoming call to the main number it will ring all the member extensions including the main number. Any extension call other extensions will display only the main number.

Navigate to web menu *Advanced->One Number Stations*. Click "New One Number Stations" button to create a one number stations group.

M	lew One Numb	er Stations	х
407 408 409 ONS Group M	↓ ↓ 4embers	403 404 405 406 410 411 412 413 Extensions	* *
	Main Extensio Ring lasting fo		

Select the extensions from the "Extensions" column to the "ONS Group Members" column. In the "Main Extension" dropdown list select an extension to be the main extension number. And click on "Save" you'll have a new one number stations group.

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

Switch Station: <u>*1</u>	Save Cancel	
---------------------------	-------------	--

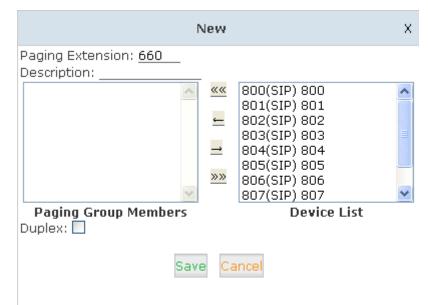
This feature code is used to switch extension during a phone call. For example, an inbound call called extension 407, the one number stations member 408 answered this call, you can press *1 from extension 407 or 409 to switch this live call to 407 or 409, 408 will be disconnected.



3.5.11 Paging and Intercom

The Paging and Intercom feature allows you to use your phone system as an intercom system, provided that your endpoints (phone devices) support this functionality. The Paging and Intercom feature allows you to define a number (just like an extension or Ring Group number) that will simultaneously page a group of devices. For example, in a small office, you might define a paging group that allows any user to dial 699, allowing them to page the entire office. You can also use the feature code *50/*51 to page/intercom a single extension, by dialing *50/*51 followed by the extension number.

Click [Advanced] -> [Paging and Intercom] -> [New Paging Group] :



Item	Explanation	
Paging Extension	The extension number for this paging group, by calling this extension	
	number you can reach the group members.	
Description	Provide a descriptive title for this Page Group.	
Paging Group Members	Selected device(s) on this page	
Device List	Select Device(s) to page.	
	If enabled the group members can talk to the caller.	
	By calling the paging extension number, all the group member phones	
Duplex	will auto answer in speaker mode (requires the IP phones support auto	
	answer feature), the caller can now make a brief announcement to the	
	group members.	



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



3.5.12 Web Extensions

Web Extensions is simply understanding of WebRTC. You can use your web browser to register an extension number to the IP PBX system without any plugins.

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

To register the first Web extensions, please follow the steps below:

Step 1:

Create a Web Extension

To create a web extension, navigate to web menu *Advanced->Web Extensions*. Click on the "New User" button to add a new web extension.

			New	
General				
Name:	680		Extension:	680
Password:	123456		Outbound CID:	
DialPlan:	Extensions	~	Transport:	WSS 💌
		Save	Cancel	

Item	Explanation
Name	User name of this web extension.
Extension	Extension number of this web extension.
Password	Password for registration of this web extension.
Outbound CID	Only works if the call was placed out through VoIP trunks.
DialPlan	Defines which type of numbers the web extension can dial.
Transport	WS or WSS
WS	WS (WebSocket) Protocol is an independent TCP-based protocol
	providing full-duplex communication channels over a single TCP
	connection. The WebSocket protocol was standardized by the IETF as
	RFC 6455 in 2011, and the WebSocket API in Web IDL is being
	standardized by the W3C.
WSS	WSS (WebSockets over SSL/TLS), like HTTPS, WSS is encrypted and
	we strongly recommend the secure wss:// protocol over the insecure
	ws:// transport. A variety of attacks against WebSockets are almost
	impossible if the transport is secured.



Step 2:

Upgrade Web extension patch

As you can see, web extensions use different protocols for signaling and media (WS/WSS) and they are not ordinary SIP/IAX2 extension that can use IP phones or softphones to register so must be treated differently.

Step 3:

Register a Web Extension

After completing the upgrade process you can access the WebRTC extension register interface. Open your web browser and enter URL <u>https://192.168.1.197/webrtc</u> (192.168.1.197 should be your IP PBX IP address) you will see the web extension register interface. Please complete the register credentials as shown below:

l ebf	ohone
Name	i.e. Homer Simpson
680	1
SIP URI	i.e. sip:homer@your-domain.com
680@192.168.1.197	1
SIP password	
SIT pasaword	1
WS URI	i.e. wss://your-domain.com:8089/ws
wss://192.168.1.197:8089/	WS
	advanced settings

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

3.5.13 PIN Set

Pin sets can be used to secure your IP PBX system phone services. For example outbound dial rules and DISA.



.Click [Advanced] --- [PIN Sets], Click on the "New Pin Set" button to create a collection of PIN codes.

New PIN Set	х
PIN Set Name: forDialRule PIN List: 54573 07259 50377 73269 Save Cancel	

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

3.5.14 Call Recording

IPPBX system has built-in ability to record calls. No additional software is required for recording calls. When IP PBX system records a call, both sides of the call are recorded and written out to a file for playback on a computer. Call recording can be used to ensure call quality, or to keep calls for later review. The IP PBX provides the ability to record all of the calls, or to selectively record calls.

Click [Advanced] -> [Call Recording] -> [New Call Recording] :

New Call Recording		
Extension: 800 (800) 801 (801) 802 (802) 803 (803) 804 (804) 805 (805) 806 (806) 807 (807) 808 (808) 809 (809)		
Call Recording Time		
Always Recording:		
Start Time:		
Call Recording Settings		
Inbound Record: Outbound Record:		
Save Cancel		



Reference:

Item	Explanation
Extension	Select the extensions which you want all their calls to be
	recorded.
Always Recording	If enabled all calls of the above selected extension will be
	recorded no matter when the calls have been made and
	received.
Start Time, End Time, Start	If Always Recording is unnecessary you can specify which time
Day, End Day	durations in a week to record all calls of the above selected
	extensions.
Inbound Record	Enable to record all inbound calls.
Outbound Record	Enable to record all outbound calls.



The recordings can be searched out on *Report->Record List->Call Recording* page.

3.5.15 Smart DID

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

Click [Advanced] -> [Smart DID] :

Smart DID

	Sma	rt DID	
	Enable Save	Cancel	
Smart DID Rules List	I	New Smart DID Rule	
Pattern	Strip	Prepend	Options
1 X.			Edit Delete

There's a default Smart DID rule, which enables all outbound calls to be monitored by Smart DID feature. If the call is not answered by the called party, then the called number will be stored into Asterisk database with the extension number which made this call. While the called party is calling back, the IP PBX system can automatically direct this call to the extension



number directly.

If you don't want all outbound calls to be monitored by Smart DID, you can modify the existing rule or click "New Smart DID Rule" to add you custom rule/rules. An example is shown below:

New Smart DID Rule
Pattern: 17951X.
Strip: 5 digits before dialing
Prepend: -886before dialing
Save Cancel

Item	Explanation
Pattern	Defines the number format which would be dialed.
Strip	Remove some digits from the front of the dialed number.
Prepend	Prepend some digits in front of the dialed number after being
	manipulated by the "Strip" option.

The numbers to be dialed will start with prefix 17951 and if they call back, the expected numbers will have +886 in front of them instead of the 5-digit prefix 17951. In such a situation, the outbound and inbound numbers are not the same, you'll need the "Strip" and "Prepend" options to manipulate the dialed numbers to make sure it can match the "same" number when it calls back. If the numbers to be called and the numbers to be received are the same, then you don't have to configure these 2 options. Or you can configure only one of these 2 options, it will all depend on the real applications.

For example the extension user 401 wants to call 86547096, and the carrier requires a prefix 17951 so the rate is much cheaper. The user would dial 1795186547096 to place this call. If the called party missed this call, IPPBX system will store this number +88686547096 with extension number 401 into its database. Later on, if the called party tried to call back, the IPPBX system gets +88686547096 as the caller ID and matches from it database, once successfully matched, this call will be automatically directed to extension 401.

The records of Smart DID functionality in the system database will be erased every day at midnight. Which means this is a dynamic effective feature.
 In the "Pattern" field, patterns that can be used are the same as the patterns used to manipulate dialed number in the dial rules.
 Due to the mechanism of how asterisk works. For now Smart DID only works



with VoIP trunks but not with FXO or GSM trunks.

3.5.16 Call Back

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

Callback Number Settings

Callback Nun	iber Settings
Prepend: DialPlan:	<pre> @ digits before dialing 0 before dialing DialPlan1 Cancel </pre>

Item	Explanation
Enable	Check the checkbox to enable call back feature.
Strip	The receive caller ID might have some additional digits in front of
	it and it's improper for you to call back directly, you can specify
	here to remove some digits before calling back.
Prepend	After the number had been manipulated by the "Strip" option, you
	can still add some extra digits in front of it before calling back.
DialPlan	Choose a proper dial plan to make sure the IPPBX system has
	the permissions for outbound calling.

Click (Advanced) -> (Callback) :

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

	New Callback Number
Callback Number:	13880424687
Destination:	Goto Extension 👻 800(800) 👻
	Save Cancel

Input callback number and define the destination.



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

Here in this case, if the caller 13880424687 calling in the IPPBX system, IPPBX will disconnect this call and make a call back to this number using extension 800.

3.5.17 Phone Book

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

Click [Advanced] -> [Phone Book] :

Phone Book

Phone Book		Import Export	Delete All Sync LDAP
The prefix of speed dial: *99 Save Cancel			
Field: Name 👻	Filter	Create Contact	Delete Selected
Name	Phone Number	Speed Dial	Options
🔲 1 Kent	85362145	01	Call Edit Delete

Item	Explanation	
Import	You can import contact list from .txt or .csv files.	
Export	Export the current contact list as .csv file.	
Delete All	Delete all contacts.	
Sync LDAP	Synchronize the contacts to the LDAP server.	
The prefix for speed dial	Using this feature code with the speed dial code of a contact you	
	can call the contact without knowing the exact number.	
Filter	Search contacts by contact name, phone number or speed dial	
	code.	
Create Contact	Create a new contact record.	
Delete Selected	Delete the selected contacts.	
Call	Assign an extension to call this contact.	
Edit	Edit the information of this contact.	
Delete	Delete this contact.	



Click [Create Contact] to see the following diagram:

	Create Contact	
Name:	Kent	
Phone Number:	85362145	
Speed Dial:	01	
Save	Cancel	

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

3.5.18 LDAP Server

3.5.18.1 LDAP Server Settings

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

LDAP Server

LDAP Server		
	Enable:	
	Username:	planettest
	Password:	•••••
	Domain:	Idapplanet.com
	Organization:	Planet Co,.LTD
	Port:	389



Item	Explanation
Enable	Enable/Disable LDAP Service.
Username	Define the username of the server administrator (e.g.: manager).
	This setting will be used on the IP Phone.
Password	Define the password of the server administrator. This setting will
	be used on the IP Phone.
Domain	Define a domain for the LDAP server (e.g.: Idapdomain.com).
	This setting will be used on the IP Phone.
Organization	Define an organization to describe the members recorded by
	LDAP (e.g.: planet.ltd). This setting will be used on the IP Phone.
Port	LDAP service port, default number 389.

3.5.18.2 Synchronize Contacts with LDAP Server

Navigate to web menu *Advanced->Phone Book*. Click on the "Sync LDAP" button to synchronize contacts with LDAP server.

Phone Book

Phone Book	Import Export Delete All Sync LDAP
The prefix of speed dial: <u>*99</u>	Save Cancel



3.5.19 Feature Codes

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

Feature Codes

eature Codes Managemen	t
Call Parking	
Ext	tension to Dial for Parking Calls: 700
	Extension Range to Park Calls: 701-720
	Call Parking Time(sec): 45
E	Enable Call Park BLF notification: 🗹
Pickup Call	
	Pickup Extension: *8
	Pickup Specified Extension: **
Transfer	
	Blind Transfer: #
	Attended Transfer: *2
	Disconnect Call: *
Timeout for ans	swer on attended transfer(sec): 15
One Touch Re	cording
	One Touch Recording: *1
Call Forward	
	Enable Forward All Calls: *71
	Disable Forward All Calls: *071
	Enable Forward on Busy: *72
	Disable Forward on Busy: *072
	Enable Forward on No Answer: *73
	Disable Forward on No Answer: *073
Do Not Distu	ırb
20110101010	Enable Do Not Disturb: *74
	Disable Do Not Disturb: *074
Spy	
	Normal Spy: *90
	Whisper Spy: *91
	Barge Spy: *92
Black List	<u> </u>

Item	Explanation
Extension to Dial for	Define an extension for parking calls.
Parking Calls	
Extension Range to Park	Define the extension range for parking calls. (e.g. 701-720)
Calls	
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.

Blacklist a number: <u>*75</u>



Item	Explanation	
Pickup Specified	Pick up the specified extension. Default: Dial**+extension number to	
Extension	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	Set the timeout value	
attended transfer (sec)		
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	In conference, the administrator can dial "0" to invite people into the	
participant	conference. After pressing "0", you will get dial tone, and you can dial	
	to invite the participant; when the call is connected, dial "**" to return	



Item	Explanation	
	to the conference with invited participant.	
Return to conference	In conference, the administrator can dial "0" to invite people into the	
without participant	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	Pause the agent, and the agent cannot receive the call.	
Extension		
Unpause Queue Member	Unpause the agent, and the agent can receive the call.	
Extension		
Others	Function key for Intercom/ Paging/ Directory	

3.5.20 Phone Provision

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

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

Enable DHCP service

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

DHCP Server Settings			
	Enable:	v	
	Start IP:	192.168.1.101	
	End IP:	192.168.1.200	
	Subnet Mask:	255.255.255.0	
	Gateway:	192.168.1.1	
	Primary DNS:	61.139.2.69	
	Lease Time(min):	1440	
	TFTP Server:		
	Save	Cancel	



Then Click [Advanced] -> [Phone Provisioning] -> [New Phone] :

	New Phon	ie	×
General			
	Enable: 🗹		
	Manufacturer: 🏾 Planet 💌	Type:	VIP-256T/PT 💌
	MAC: 00304f		VIP-256T/PT
Line			VIP361PE VIP-362WT
Line1	Extension: 🛛 💌	Label:	ICF-1700
	Save Ca	ncel	VIP-2020PT VIP-5060PT

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



Chapter 4. Network Settings

4.1 Network

IPPBX system supports static IP, DHCP and PPPoE as WAN connection options, and on LAN port only static IP is supported. If you are configuring WAN connection as static IP or DHCP, make sure WAN and LAN IP addresses are not in the same network.

4.1.1 IPv4 Settings

Click [Network Settings] -> [Network] -> [IPv4 Settings]

Network

IPv4	Settings	IPv6	Settings	VLA	AN Settings	
WAN Port Setup						
		IP Assig	n: Static 🗸			
	IF	Address:	192.168.1.19	97		
	Sub	net Mask:	255.255.255	.0		
		Gateway:	192.168.1.2	54		
	Prir	mary DNS:	8.8.8.8			
	Alterna	ative DNS:	168.95.1.1			
LAN Port Setup						
IP Addre	ess: 192.168	.0.1	Subne	t Mask:	255.255.255.0	
IP Address	V1:		Subnet N	/askV1:		
IP Address	V2:		Subnet N	1askV2:		
		Save	Cancel			

Reference

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

By default IP PBX has been preconfigured with static IP 172.16.0.1 and 192.168.0.1 on WAN and LAN interfaces. If you want to use a static IP, just configure here with the address, netmask, gateway and DNS given to be the ISP or the network admin.



And the LAN interface you can specify 2 additional virtual IP addresses. It can be used to access some other networks from the LAN port.

4.1.1.1 DHCP

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

	IPv4 Setti	ings	IPv6 s	Settings	VL4	AN Settings	
WAN Por	t Setup						
			IP Assig	n: DHCP 🗸			
		IP	Address:	192.168.1.1	97		
		Sub	net Mask:	255.255.255	5.0		
			Gateway:	192.168.1.2	54		
		Primary DNS:		8.8.8.8			
	Alternative DNS:		168.95.1.1				
LAN Port	Setup						
	IP Address: IP AddressV1: IP AddressV2:	192.168	.0.1	Subne Subnet M Subnet M	MaskV1:	255.255.255.0	

If DHCP is selected, WAN interface will not be configurable; it obtains all network parameters from the DHCP server. DHCP should be used cautiously. If all the IP extensions subscribe to the IPPBX system through WAN, you'd better make sure WAN gets a Static DHCP.



4.1.1.2 PPPoE

If PPPoE IPPBX will be connected to the network via ADSL modem by means of Point-to-Point Protocol over Ethernet (PPPoE) dial-up. In such a situation, extensions will subscribe to the IPPBX system through LAN, WAN port can be used for remote extensions.

	IPv4 Sett	ings	IPv6 s	Settings	VLA	N Settings	
WAN Por	t Setup						
			-	n: PPPoE - e: pppoe01			
			Password	d: •••••••			
		IP	Address:	192.168.1.19	97		
		Subr	net Mask:	255.255.255	i.0		
		(Gateway:	192.168.1.2	54		
		Prin	nary DNS:	8.8.8.8			
		Alterna	tive DNS:	168.95.1.1			
LAN Port	Setup						
	IP Address: IP AddressV1: IP AddressV2:	192.168.	0.1	Subne Subnet N Subnet N	MaskV1:	255.255.255.0	

If PPPoE is set, you just have to specify the username and password given by your ISP and the IPPBX system will dial-up to the ISP and you have Internet access on WAN. LAN port connects to your local network for internal IP extensions to register. If needed, you can change LAN IP to fit your local network.

4.1.2 IPv6 Settings

IPv6 (Internet Protocol Version 6) has been in development for nearly two decades. Now the next-generation protocol is ready to replace IPv4 and assume its place as the backbone of the Internet.

Today, major Internet service providers (ISPs), home networking equipment manufacturers, and web companies around the world are permanently enabling IPv6 for their products and services. Many organizations, institutions and universities have deployed their own networks on IPv6.

To be able to deliver VoIP calls over IPv6 (SIP over IPv6), you can configure IP PBX system



with IPv6 addresses to be able to deploy it in your IPv6 network infrastructure.

Click [Network Settings] -> [Network] -> [IPv6 Settings]

	IPv4 Settings	IPv6 Se	ttings	VLAN Settings
WAN Por	t Setup			
		Enable:	V	
	IPv6 Address: 2001:db8:4005:80a::200e			
	Prefix Length: 64			
Gateway: 2001:db8:4005:80a::1				4005:80a::1
Primary DNS: 2001:da8:8000:1			8000:1:202:120:2:	
	Alter	mative DNS:		
		Save	Cancel	

IPv6 Reference:

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

4.1.3 VLAN Settings

With a layer-3 switch you can configure VLAN on IP PBX system to divide the VoIP and data traffic. Voice VLAN can keep the phones working even when the data network is congested. You can see here on this page. You are able to configure 4 VLANs, 2 for each WAN or LAN port.

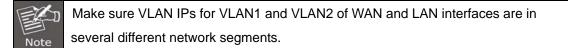


Click [Network Settings] -> [Network] -> [VLAN Settings] :

Network				
	IPv4 Settings	IPv6 Se	ettings	VLAN Settings
WAN VLAN	1			
		Enable: VLAN ID: IP Address: ubnet Mask:		
	2			
	Enable: VLAN ID: VLAN IP Address: Subnet Mask:			
LAN VLAN 1				
		Enable: VLAN ID: IP Address: ubnet Mask:		
LAN VLAN 2				
	Enable: VLAN ID: VLAN IP Address: Subnet Mask:			

VLAN Reference:

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



4.2 Static Routing

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



Click [Network Settings] -> [Static Routing] :

New Static Routing			
Destination Network:	222.209.4.1		
Subnet Mask:	255.255.255.255		
Gateway:	192.168.10.1		
Save	Cancel		

ltem	Explanation
Destination	Set the IP address of destination host or network address.
	E.g.222.209.4.1, 192.168.10.0.
Subnet Mask	Set subnet mask of the destination network.
Gateway	Define the gateway accessing the destination network.

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

Routing Table

	St	atic Routing	Routing	g Table			
Routing Table: Kernel IP routin Destination 0.0.0.0 192.168.1.0	ng table Gateway 192.168.1.254 0.0.0.0	Genmask 0.0.0.0 255.255.255.0	Flags UG U	Metric O O	Ref O O	0	Iface ETH ETH

4.3 VPN Server

VPN (Virtual Private Network) is mostly used for setting up long-distance and/or secured network connections. While it's been used on IP PBX, all the phone calls sending and receiving are encrypted so it secures your remote offices/extensions' phone services. Built-in VPN Server on Planet IP PBX series is an easy way to set up such secured connectivity between other Planet series IP PBXs or IP phones. You don't need to build a dedicated VPN server or buy a VPN router. This is also a workaround to avoid a firewall issue when configuring remote VoIP client as SIP protocol is notoriously to pass through a firewall due to its random numbers to establish connection.



The IP PBX supports four kinds of VPN variety: L2TP/PPTP/OpenVPN/IPSec. Click [Network]

Settings] -> [VPN Server] :

VPN Server

	VPN Server	VPN Users Management	:
VPN Server			
	© L2TP C ∣	PPTP C OpenVPN C IPSec	
Enable: Remote St Remote Er Local IP: Primary DN Alternative Authentica Debug: IPSec:	nd IP: NS: e DNS: ation Method:	Cancel	

Status: L2TP (Disabled)



4.3.1 L2TP VPN

VPN Server

VPN Server	VPN Users Management
VPN Server	
	🖲 PPTP 🔘 OpenVPN 🔘 IPSec
Enable:	
Remote Start IP:	192.168.10.2
Remote End IP:	192.168.10.10
Local IP:	192.168.10.1
Primary DNS:	8.8.8.8
Alternative DNS:	168.95.1.1
Authentication Method:	🗹 chap 🗹 pap
Debug:	
IPSec:	
IPSec Local IP:	192.168.1.197 👻
IPSec Password:	12345678
	Save Cancel

Reference:

Item	Explanation
Enable	Tick the checkbox to enable L2TP VPN server.
Tick the checkbox	L2TP VPN remote network IP range, between start IP and end IP
to enable L2TP	there must be less than 10 available IP addresses.
VPN server.	
Local IP	L2TP VPN local server IP address.
Primary DNS	Primary DNS for VPN connection.
Alternate DNS	Alternative DNS for VPN connection.
Authentication	Select the authentication method: chap or pap.
Method	pap: Password Authenticate Protocol PAP works like a standard
	login procedure; it uses static user name and password to
	authenticate the remote system.
	chap: Challenge Handshake Authentication Protocol
	CHAP takes a more sophisticated and secure approach to
	authentication by creating a unique challenge phrase (a randomly
	generated string) for each authentication.
Debug	Tick to enable debug for L2TP VPN connection, debug info will be



Item	Explanation
	written into system logs.
IPSec	Enable IPSec encryption for L2TP VPN server.
IPSec Local IP	IP PBX WAN IP which can access Internet.
IPSec Password	Define a password for IPSec VPN client to authenticate.

If the IP PBX system is behind NAT, you need to open ports 500, 4500 and 1701 on the router/firewall.

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

VPN Users Management

	VPN Server	VPN Users Management	
List of VPN Use	rs	New VPN User	
Username		Availability	Options
1 test		yes	Edit Delete

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

4.3.2 PPTP VPN

The Point-to-Point Tunneling Protocol (PPTP) uses a control channel over TCP and a GRE tunnel operating to encapsulate PPP packets. The intended use of this protocol is to provide security levels and remote access levels comparable with typical VPN products.



4.3.2.1 PPTP VPN Server

Navigate to web menu *Network Settings->VPN Server*. Check the radio button of PPTP to configure PPTP VPN server.

VPN Server

VPN Server	VPN Users Management
VPN Server	
C L2TP	◉ PPTP ◎ OpenVPN ◎ IPSec
Enable: Remote IP: Local IP: Primary DNS: Alternative DNS: Timeout(sec): Authentication Method: Enable mppe128:	▼ 192.168.100.2 192.168.100.1 8.8.8.8 168.95.1.1 20 Chap pap ♥mschap ♥mschap-v2 ▼
Debug:	Save Cancel

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



Item	Explanation		
	• mschap-v2: Microsoft Challenge Handshake Authentication		
	Protocol version 2 (MS-CHAP v2), it provides stronger security for		
	remote access connections.		
Enable mppe128	Microsoft Point-to-Point Encryption (MPPE) encrypts data in		
	Point-to-Point Protocol (PPP)-based dial-up connections or		
	Point-to-Point Tunneling Protocol (PPTP) virtual private network		
	(VPN) connections with 128-bit key.		
Debug	Tick to enable debug for PPTP VPN connection, debug info will be		
	written into system logs.		

For the VPN client to connect you'll need to create a VPN user account. Click the "VPN User Management" tab and click the "New VPN User" button to add a VPN user account.



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

4.3.3 OpenVPN

OpenVPN is an open-source software application that implements virtual private network (VPN) techniques for creating secure point-to-point or site-to-site connections in routed or bridged configurations and remote access facilities. It uses a custom security protocol [3] that utilizes SSL/TLS for key exchange. It is capable of traversing network address translators (NATs) and firewalls. It was written by James Yonan and is published under the GNU General Public License (GPL).

OpenVPN allows peers to authenticate each other using a pre-shared secret key, certificates, or username/password. When used in a multiclient-server configuration, it allows the server to release an authentication certificate for every client, using signature and Certificate authority. It uses the OpenSSL encryption library extensively, as well as the SSLv3/TLSv1 protocol, and contains many security and control features.



Check the radio button of OpenVPN to configure OpenVPN server.

VPN Server		
(🔍 L2TP 🔍 PPTP 🖲 OpenVPN	O IPSec
Enable: Stealth: Certificate: Port: Stealth Port: Protocol:	 ✓ ✓ Done 1194 443 TCP ▼ 	Create Delete
Device Node: Cipher: Compress Lzo: TLS-Server: Remote Network: Route: Client-to-Client:	TUN ▼ Default ▼ ∅ 172.16.0.0 172.16.0.0 ☑ Save Cancel	/ 255.255.255.0 / 255.255.255.0

Item	Explanation
Enable	Tick to enable OpenVPN server
Stealth	Some deep packet inspection firewalls might not allow OpenVPN
	traffic, stealth SSL tunneling can disguises your OpenVPN traffic
	under the HTTPS traffic which is often seen as HTTPS traffic by the
	DPI.
Certificate	Certificate is one of the client authentication methods of OpenVPN.
Port	OpenVPN service port, default is 1194.
Stealth Port	Stealth service port, default is 443.
Protocol	You can choose from UDP or TCP. As stealth requires TCP only so if
	with stealth enabled, this options is not configurable and will use TCP
	by default.
Device Node	TUN or TAP; A TAP device is a virtual Ethernet adapter, while a TUN
	device is a virtual point-to-point IP link.
Cipher	Cipher (or cypher) is an algorithm for performing encryption or
	decryption.
Compress LZO	LZO is an efficient data compression library which is suitable for data
	de-/compression in real time.
TLS-Server	TLS is an excellent choice for the authentication and key exchange
	mechanism of OpenVPN.
Remote Network	OpenVPN remote network.
Route	The route entries adjust the local routing table, telling it which network
	to route over the VPN.
Client-to-Client	Client-to-Client can enable the intercommunication between clients.



4.3.4 IPSec VPN

Internet Protocol Security (IPsec) is a protocol suite for secure Internet Protocol (IP) communications by authenticating and encrypting each IP packet of a communication session. IPSec can be configured to operate in two different modes, Tunnel and Transport mode. Use of each mode depends on the requirements and implementation of IPSec.

4.3.4.1 IPSec VPN Server (Tunnel mode)

Tunnel mode is used to encrypt all traffic between secure IPSec Gateways, for example two IP PBX's, each acts as an IPSec Gateway for the hosts/IP phones behind it. The WAN ports will be used to connect to each other to establish IPSec VPN connection; the PCs or IP phones on the LAN ports can communicate with each other on both sides via secured IPSec tunnel. Check the IPSec radio button to configure IPSec VPN server.

VPN Server

VPN Server	VPN Users Management
PN Server	
C L2TP) PPTP 🔘 OpenVPN 🖲 IPSec
Enable:	
Type:	Tunnel 👻
IPSec Local IP:	192.168.1.197 -
IPSec Password:	12345678
IPSec Remote IP 1:	192.168.10.1
IPSec Remote Network 1:	192.168.20.0 / 255.255.255.0
IPSec Remote IP 2:	
IPSec Remote Network 2:	/
IPSec Remote IP 3:	
IPSec Remote Network 3:	/
	Save Cancel

Item	Explanation
Enable	Tick the checkbox to enable IPSec VPN server.
Туре	Default Tunnel mode.
IPSec Local IP	IP PBX WAN IP, which can be used to connect to the client network.
IPSec Password	Define a password for authentication of the IPSec client.
IPSec Remote IP	IPSec VPN client IP. The client uses this IP to connect to IPSec
	server.



Item	Explanation
IPSec Remote	Specify the IPSec VPN client LAN network address.
Network	

1. If the IPPBX is behind NAT, ports 500 and 4500 need to be opened on the router/firewall.

2. If the IPPBX connects to Internet via PPPoE, then IPSec Local IP needs to be the IP address assigned by PPPoE.

3. IPSec VPN server can connect 3 IPSec clients.

4.3.4.2 IPSec VPN server (Transport mode)

IPSec Transport mode is used for end-to-end communications, NAT traversal is not supported with the transport mode. So if two IP PBX's are connected via IPSec transport mode, IPSec only encrypts the communication service ports, not like Tunnel mode which encrypts the whole LAN subnet.

Check the IPSec radio button.

VPN Server

	VPN Server	VPN Users Management
VPN Server		
	© l2tp © f	PPTP 🔘 OpenVPN 🖲 IPSec
Enable:		
Type:		Transport -
IPSec Loc	al IP:	192.168.1.197 -
IPSec Pas	sword:	12345678
	[Save Cancel

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



4.4 VPN Client

Planet IP PBX supports four kinds of VPN Clients: L2TP, PPTP, OpenVPN and N2N. Click [Network Settings] -> [VPN Client] :

4.4.1 L2TP VPN Client

VPN Client

VPN Client	
) PPTP 🔘 OpenVPN 🔘 N2N 🔘 IPSec
Enable:	
Server Address:	192.168.1.21
Username:	test1
Password:	••••
IPSec:	
IPSec Local IP:	192.168.1.197 👻
IPSec Password:	12345678
Default Gateway:	
	Save Cancel

Reference:

Item	Explanation
Enable	Tick to enable L2TP VPN client
Server Address	L2TP server public IP.
Username	L2TP VPN user name given by the VPN server.
Password	L2TP VPN user password given by the VPN server.
IPSec	Enable IPSec support.
IPSec Local IP	IPPBX WAN IP which can access Internet.
IPSec Password	Accordingly as the password specified on the server.
Default Gateway	All traffic goes through the L2TP VPN connection.



4.4.2 PPTP VPN Client

On the branch office site, check the radio button of PPTP to configure PPTP VPN client.

VPN Client

VPN Client	
© L2TP 🖲	PPTP OpenVPN N2N IPSec
Enable: Enable 40/128-bit encry	ption for MPPE:
Server Address: Username:	192.168.1.21 test1
Password: Default Gateway:	•••••
	Save Cancel

Item	Explanation
Enable	Tick to enable PPTP VPN client.
Enable 40/148-bit	Tick to enable 40-bit key (standard) or 128-bit key (strong) MPPE
encryption for	encryption schemes.
MPPE	
Server Address	PPTP VPN server public IP.
Username	PPTP VPN user name given by the VPN server.
Password	PPTP VPN user password given by the VPN server.
Default Gateway	All traffic goes through the L2TP VPN connection.

4.4.3 N2N VPN Client

N2N is an open source Layer 2 over Layer 3 VPN application which utilizes a peer-to-peer architecture for network membership and routing.



On IP PBX system we support N2N VPN client. Check the radio button of N2N VPN and configure the client info.

PN Client		
L2TP	○ PPTP ○ OpenVPN ● N2N ○ IPSec	
Enable:		
Server Address:	88.86.108.50	
Port:	82	
Local IP:	192.168.20.101	
Subnet Mask:	255.255.255.0	
Local Port:	30256	
Username:	user1	
Password:	•••••	

Item	Explanation
Enable	Tick this checkbox to enable N2N VPN client
Server Address	N2N server(supernode) IP address.
Port	N2N service port number. 82 by default.
Local IP	VPN local IP.
Subnet Mask	Netmask of the VPN network.
Local Port	N2N local service port.
Username/Password	Used for the N2N server to authorize the connection.

4.4.4 IPSec VPN Client (Tunnel mode)

On the remote site, open the web GUI of another Planet IPPBX system and navigate to web menu *Network Settings->VPN Client*.

On the VPN Client page, choose IPSec and tick "Enable" option to enable IPSec client.

VPN Client	
VPN Client	
○ L2TP ○ PP	TP 🔿 OpenVPN 🔿 N2N 🖲 IPSec
Enable:	v
Type:	Tunnel 💙
IPSec Local IP:	192.168.1.252 🗸
Server Address:	117.176.159.163
IPSec Password:	hPC2he@Q
IPSec Remote Network:	192.168.10.0 / 255.255.255.0
	Save Cancel

Item	Explanation	
Enable	Tick the checkbox to enable IPSec client.	
Туре	Accordingly as the IPSec server.	



Item	Explanation
IPSec Local IP	WAN port IP which can connect to the IPSec server.
Server Address	Specify the IPSec server IP.
IPSec Password	Specify the IPSec VPN password defined previously on the server.
IPSec Remote	The IPSec VPN server LAN network address.
Network	

4.5 DHCP server

DHCP (Dynamic Host Configuration Protocol) is a standardized network protocol used on Internet Protocol (IP) networks for dynamically distributing network configuration parameters, such as IP addresses for interfaces and services.

With DHCP, computers/IP phones request IP addresses and networking parameters automatically from IP PBX WAN/LAN port; it saves a lot of time for administrator to configure these settings manually.

Click [Network Settings] -> [DHCP Server] :

4.5.1 DHCP Service

DHCP Server

DF	ICP Server	DHCP Client List	Static MAC
DHCP Server Set	ttings		
	Enable: Interface Start IP: End IP: Subnet M Gateway Primary Lease Ti TFTP Ser	192.168.1. 192.168.1. 192.168.1. Mask: 255.255.25 /: 192.168.1. DNS: 192.168.1. me(min): 1440	199 55.0 1

Item	Explanation
Enable	Enable DHCP service.



Item	Explanation	
Interface	Choose the network port to implement DHCP service.	
Start IP, End IP	Specify the DHCP IP address pool.	
Subnet Mask	Netmask to be assigned to the client devices.	
Gateway	Gateway address to be assigned to the client devices	
Primary DNS	DNS to be assigned to the client devices.	
Lease Time(min)	DHCP server leases an address to a new device for a period of time.	
	When the lease expires, the DHCP server might assign the IP	
	address to a different device. Default value is 1440 minutes.	
TFTP Server	Point out the TFTP server address which may be used to auto	
	provision the IP phones.	

4.5.2 DHCP Client List

You'll have all the devices that are getting IP address from the IP PBX system.

Click [Network Settings] -> [DHCP Server] -> [DHCP Client List] :

DHCP Client List:				
Mac Address	IP Addres	s	Host Name	Expires i
6c:3e:6d:e0:f2:00 1	192.168.1	.101	iPhone	expired
00:03:58:45:87:9a 1	192.168.1	.102		expired
0c:74:c2:47:71:6d 1	192.168.1	.103	hnteki-iPhone	expired
20:c9:d0:85:3b:fb 1	192.168.1	.104		expired
08:ed:b9:e7:c5:7f 1	192.168.1	.105	DPVYE1J0WCAAC7	I expired
78:e4:00:8e:c3:99 1	192.168.1	.106	LBSZLACHCIC	22:10:25
68:a3:c4:ef:5d:8b 1	192.168.1	.107	HBWang	1 days 00
0c:72:2c:5a:39:41 1	192.168.1	.108	MW150R	00:00:57

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

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

4.5.3 Static MAC

Static MAC is a useful feature which makes the DHCP service on IP PBX always assigns the same IP address to a specific computer or IP phone on your LAN. To be more specifically, the DHCP service assigns this static IP to a unique MAC address assigned to each NIC on your LAN.



Click "New Static MAC" to add a record to the IP PBX system.

	New Static MAC
MAC Address: IP Address:	00304fdee5f6 192.168.1.123
	Save Cancel

4.6 DDNS Settings

Unlike DNS that only works with static IP addresses, DDNS (Dynamic Domain Name Server) is designed to also support dynamic IP addresses, such as those assigned by a DHCP server. Built-in DDNS feature on IP PBX system only needs a simply signs up with a Dynamic DNS provider, with the domain name they gave which maps your IP address on the Internet, you can access IP PBX and also other services within your LAN via the domain name without getting to know Dynamic public IP.

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

DDNS Settings	
Enable: Enable EasyDDNS: Easy Domain: DDNS Server: Username: Password:	✓ pl72c426.planetddns.com PlanetDDNS.com ▼
Domain:	Save Cancel

Click [Network Settings] -> [DDNS Settings] :

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



Item	Explanation
Domain	Your domain name.

DDNS Settings	
Enable: Enable Easy DDNS Easy Domain: DDNS Server: Username: Password: Domain:	S: V pl11223f.planetddns.com PlanetDDNS.com V
	6 change ip , do DDNS update ! 6 DDNS successfully updated om : IP=210.61.134.91

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

DDNS Settings		
	Enable: Enable Easy DDNS: Easy Domain: DDNS Server: Username: Password: Domain:	✓ PlanetDDNS.com ▼ PlanetDDNS.com PlanetDDNS.com Dyndns.org No-ip.com Zoneedit.com



4.7 SNMPv2 Settings

SNMP (Simple Network Management Protocol) is used for remote management. Click [Network Settings] -> [SNMPv2 Settings] :

SNMPv2	Settings
--------	----------

Read Only		
	Enable: RO Community: RO Network:	Dublic / 24
Read and Write		
	Enable: RW Community: RW Network:	private 192.168.10.0 / 24
		Save Cancel

Reference

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

4.8 TR069

TR069 (Technical Report 069) is a Broadband Forum (formerly known as DSL Forum) technical specification entitled CPE WAN Management Protocol (CWMP). It defines an application layer protocol for remote management of end-user devices.

TR069 Settings		
Enable:		
CPE to ACS URL:	http://192.168.1.69/acs	
ACS Authentication Mode:	BASIC 🔻	
ACS Username:	user	
ACS Password:	123456	
CPE Inform Interval(sec):	42200	
ACS to CPE URL:	http://192.168.1.78:7547	
Save Cancel		



Item	Explanation
Enable	Enable TR069 service.
CPE to ACS URL	Input URL to visit ACS, which is used by PBX to connect ACS via
	CPE WAN management protocol (CWMP).
ACS	Select ACS Authentication Mode: NONE/BASIC/DIGEST.
Authentication	
Mode	
ACS Username	When PBX sends request to ACS, ACS will provide username to the
	authorized PBX.
ACS Password	When PBX sends request to ACS, ACS will provide password to the
	authorized PBX.
CPE Inform	Interval for CPE to connect ACS.
Interval (sec)	
ACS to CPE URL	Input URL to visit CPE. Format: http://IP:port(7547).



Chapter 5. Security

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

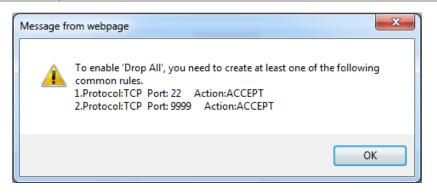
5.1 Firewall

The IP PBX system has been preconfigured with a built-in firewall which prevents your IP phone system from unauthorized accessing, phone calls and some other attacks.

General

General			
	Enable Firewall: 🗹	Disable Ping: Save Cancel	Drop All:

Item	Explanation
Enable Firewall	By default, firewall is enabled. You may disable the built-in firewall by
	unchecking "Enable Firewall" checkbox, only if your IP PBX is behind
	a router/firewall without port forwarding to the Internet.
Disable Ping	Ignore ping request. If enabled, you cannot ping the IPPBX system.
Drop All	Drop all packets sent to the IP PBX system, this will cause IP PBX
	system blocking all communication with the outside world. So the
	system will prompt to add at least one grant rule on port 22 (SSH) or
	9999 (Web) to make sure the IPPBX system is totally unreachable.



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

Common Rules

In Common Rules section, you can configure the firewall to grant or deny an IP address or a network from communicating with the IPPBX system. Even the service port number can be specified so it can grant or deny a specific IP or network to access a specific service.



By clicking the "Add Rule" button you can add a custom rule for rejecting or accepting an IP address or network address.

Add Rule	X
Name: block-sip	
Description: suspected attack	
Protocol: TCP/UDP	//
Port: 5060 -	
IP: 5.189.154.148 /255.255.255.255	20
Note: Set a network segment(10.10.10.0/ or a network address(10.10.10.124/255.2	
MAC:	
Action: DROP V	
Save Cancel	

Item	Explanation
Name	A name for this rule.
Description	Optional, you may describe why this rule is created.
Protocol	Transmission protocol, UDP, TCP or UDP with TCP.
Port	Service port number.
IP	Can be an IP address or a network address.
MAC	Action to be taken according to the Mac address of a device instead
	of IP. Only works with the devices within the same local network
	because Mac address cannot transport on Internet.
Action	Select "Drop" to block and "Accept" to grant.

Auto Defense

The IPPBX system uses Fail2Ban to do intrusion detection, iptables is used for blocking the attack attempts. Fail2Ban is an intrusion prevention framework written in the Python programming language. It works by reading Asterisk logs and some other logs in the IP PBX system, and uses iptables profiles to block brute-force attempts.



In Auto Defense section you can define some custom rules to help the IP PBX system determine brute-force attempts.

Auto Defense Add Rule			
Port	Protocol	Rate	Options
5060	UDP	120/30s	Edit Delete
5060	UDP	40/2s	Edit Delete
5061	TCP	80/2s	Edit Delete
22	TCP	10/60s	Edit Delete

Click the "Add Rule" button to add a new custom rule.

Add Rule	х
Port: 9999	
Protocol: TCP 🔻	
Packets: <u>10 (</u> 1-200)	
Time Interval: 30 seconds	
Save Cancel	

In this case, it will block the IP which will send more than 10 packets to the port 9999 within 30 seconds. This rule will prevent brute-force attempts of the web login.

Rejected IP

Any IP address that is banned will be shown in the table of "Rejected IP". The table will show the IP address of the banned host, as well as what kind of service was detected as to the intrusion.

Rejected IP		
Туре	IP	Options
VOIP	212.83.154.178	Delete
VOIP	173.249.158.227	Delete
VOIP	5.189.154.148	Delete

If a host appears incorrectly in the list of rejected IP, you can click on the "Delete" button to remove it from the list.

5.2 Service

As we can see here on this page, you are able to configure the SSH and HTTPS services.



Click [Security] -> [Service] :

Service Settings

Service Settings	
	Enable SSH: Port:22 Remote SSH Administration:
	HTTPS Port: 443 Remote HTTPS Administration:
	Save Cancel

Item	Explanation
Enable SSH	With this option you can enable or disable SSH access to the IPPBX
	system. It's not enabled (unchecked) by default.
Port	By default SSH service port number is 22. You can change it to any
	other available port number.
Remote SSH	If this option is checked, remote SSH access will be enabled.
Administration	
HTTPS Port	Web GUI service port number, by default, is 9999. You can change to
	any other port number if needed.
Remote HTTPS	If this option is checked, remote web access will be enabled.
Administration	

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

5.3 Fail2Ban

Planet IPPBX system uses Fail2Ban to perform intrusion detection; iptables is used for blocking any attack attempts. Fail2Ban is an intrusion prevention framework written in the Python programming language. It works by reading Asterisk logs and some other logs in the IPPBX system, and uses iptables profiles to block brute-force attempts. In the Auto Defense section you can define some custom rules to help the IPPBX system determine brute-force attempts.

Allowed address allows you to add IP addresses and network addresses to the IPPBX system



as a whitelist. The IPs in the whitelist will be always treated as trusted IP and will not be filtered by the firewall rules.

Click the "Add New IP" button to add a trusted IP or network to the system IP whitelist.

	Add Allowed IP	х
Description:	all	
Protocol:	✓SIP ✓IAX2 ✓HTTPS✓SSH	
Allowed IP:	117.176.159.157	
Subnet Mask:	255.255.255.255	
Availability:	Yes 🔻	
	Save Cancel	

Item	Explanation
Description	A name for this entry.
Protocol	Select protocols this IP/network can access.
Allowed IP	IP address or network to be trusted.
Subnet Mask	Netmask for this IP or network.
Availability	Choose "Yes" to activate this entry; choose "No" to deactivate.



Fail2Ban

	Fail2Ban	Settings	
SIP			
	Max Retry: <u>10</u> Find Time: <u>600</u> Ban Time: <u>3600</u>		
IAX2			
	Max Retry: <u>10</u> Find Time: <u>600</u> Ban Time: <u>3600</u>	_	
HTTPS			
	Max Retry: <u>5</u> Find Time: <u>600</u> Ban Time: <u>600</u>	_seconds _seconds	
SSH			
	Max Retry: <u>5</u> Find Time: <u>600</u> Ban Time: <u>600</u>	_seconds _seconds	
	Save	Cancel	

These options are actually for Fail2Ban, the "Max Retry" limits the authentication attempts. "Find Time" defines the time duration from the first attempt to the last attempt which reaches the "Max Retry" limitation. "Ban Time" is the time in seconds the IPPBX system will block the IP which exceeded max retry. These settings also don't effect on the allowed addresses.



Chapter 6. Report

6.1 Record Status

On register status page you are able to check the extension and SIP/IAX2 trunk status intuitively. You can see from which IP the extension is registered and also you can see the connection state, for example how much delay is there between the IPPBX system and the end point.

6.1.1 SIP User Status

Register Status 🌣

SIP Users Status		IAX2 Users Status	SIP Trunks Status			IAX2 Trunks Status	
P Users Statu	5						
Name	Extension	IP	NAT	ACL	Port	Status	
800	800	N/A	No	No	N/A	Unregistered	
801	801	N/A	No	No	N/A	Unregistered	
802	802	N/A	No	No	N/A	Unregistered	
803	803	N/A	No	No	N/A	Unregistered	
804	804	N/A	No	No	N/A	Unregistered	
805	805	N/A	No	No	N/A	Unregistered	
806	806	N/A	No	No	N/A	Unregistered	
807	807	N/A	No	No	N/A	Unregistered	
808	808	N/A	No	No	N/A	Unregistered	
809	809	N/A	No	No	N/A	Unregistered	

Here on this page you can see the SIP/IAX2 extensions, web extensions and also the register status of the trunk users. Only the trunk is configured as peer mode which will be listed here.

Status and Description

- Registered: Registration success.
- Unregistered: Registration failure or unapplied.
- Unreachable: Network delay.
- Timeout: Network timeout.

6.1.2 IAX2 User Status

Users Status				
Name	Extension	IP	Port	Reachability
412	412	192.168.7.32	4569	Registered (2 ms)
413	413	N/A	N/A	Unregistered



Status and Description

- Registered: Registration success.
- Unregistered: Registration failure or unapplied.
- Unreachable: Network delay.
- Timeout: Network timeout.

6.1.3 SIP Trunk Status

Hostname/IP	Status	
gw1.sip.us:5060	Registered	
183.62.205.209:5060	Registered	
	gw1.sip.us:5060	gw1.sip.us:5060 Registered

Here you can see all your outbound SIP trunks' status.

Status and Description

- Registered: Successfully registered to the service provider and ready for phone calls.
- Request Sent: If this status, it's most probably the network is totally unreachable to the SIP server. Please make sure network setting on the IPPBX system is correct.
- Waiting for Authentication: If "Waiting for Authentication" then most probably the register request has already been received by the server side but cannot authenticate the register request due to credentials incorrect. Please double check your inputted credentials.
- Failed: After trying to register within certain time period without success, you get "Failed" on the trunk status.

6.1.4 IAX2 Trunk Status

AX2 Trunks Status		
Username	Hostname/IP	Status
asterisk	192.168.7.146:4569	Registered

Here you can see all your outbound IAX2 trunks' status.

Status and Description

- Registered: Successfully registered to the service provider and ready for phone calls.
- Request Sent: If this status, it's most probably the network is totally unreachable to the service provider. Please make sure network setting on the IPPBX system is correct.
- Waiting for Authentication: If "Waiting for Authentication" then most probably the register request has already been received by the server side but cannot authenticate the register request due to credentials incorrect. Please double check your inputted credentials.
- Failed: After trying to register within certain time period without success, you get "Failed" on the trunk status.



6.2 Fax List

You can search any fax received by the IPPBX system.

Fax List

Start Date: End Date:		15 ▼ Fie 15 ▼	eld: Caller ID 🔹	Filter
Caller ID	Destination	Date	File Name	Status
02037085791	800	12/04/15 13:1	5 fax000000007.tif 💟	Done
01085790903	800	11/24/15 20:3	7 fax00000006.tif 💟	Done
01085790903	800	11/20/15 16:2	6 fax000000005.tif 💟	Done
02082303466	800	11/18/15 16:0	6 fax000000004.tif 💟	Done
051786244043	800	11/12/15 09:5	2 fax000000002.tif 💟	Done

In the "Start Date" and "End Date" fields specify a time duration, and click "Filter" to get all faxes received during this period. If you specify a "Caller ID" or "Destination ID" in the field, you can get the fax sent/received by a specific number in this period.

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

6.3 Record List

6.3.1 Call Recording

You are able to search all recorded call conversations if you have configured the extension always to be recorded.

Call Recording

		Cal	l Recording	Confere	ences	On	e Touch Recor	ding			
Exte	ensio	on: 303 - Delete		Field	d: Caller ID	Ŧ					
Star	t Da	ite: Jan 👻 1 👻 201	6 🗸	End	Date: Jan	10 -	2016 - Filt	er			
List	of R	ecording Files					Delete Selec	ted			
		Caller ID	Destination II	D Date	2	[Duration(sec)		C	ptions	
	1	301	303	2016	/01/04 18:2	25:43	12	P	lay	Delete	8
	2	301	303	2016	/01/04 18:2	20:36	9	P	lay	Delete	M

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





Item	Explanation
Field	Filter the recordings by specifying caller ID or destination ID. For
	example, if you select "Caller ID" and specify number 301, you get the
	recordings of the calls made by extension 301; if you select
	"Destination ID" and specify number 301, you get the recordings of
	the calls which called extension 301.
Start Date/End	Search the recordings during this time period.
Date	
Delete Selected	Delete the select recording items.
Caller ID	Caller ID of this recorded call.
Destination ID	The number the caller called.
Date	Exact time when this call recording began.
Duration (sec)	Duration of the recording.
Options	Playback, delete and download options of the recording files.
Play	You can play back the recordings directly on the web page or play
	back on a specific phone.

6.3.2 Conferences

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

Conferences

	Call Recording	Conferences	One Touch Recording	
Start Date: Jul 🔻 15	▼ 2016 ▼	End Date:	Jul • 15 • 2016 • Filter	
List of Conf	ference Record Files		Delete Selected	Delete All
Conference F	Room	Date		Options

Item	Explanation
Start Date/End	Specify a time duration to search the recorded conferences.
Date	
Delete Selected	Delete the selected searched results.
Delete All	Delete all searched results.
Conference Room	The number of the recorded conference.
Date	Exact time when the conference began.
Options	Playback, delete or download the recording file.
Play	Playback the recordings directly on the web page or playback on a
	specific phone.
Delete	Delete the recorded audio file.



6.3.3 One Touch Recording

Call recordings recorded by one touch recording feature code *1 can be found on the *Report-> Record List->One Touch Recording* page.

One Touch Recording

			Call Recording	Confe	erences	One	e Touch Recordin	g	
Exte	ensio	n: 301 🔻 🕻	elete						
Star	t Da	te: Jan 👻 1	▼ 2016 ▼	Er	nd Date: Jan 🗸	10 -	2016 - Filter		
List	of R	ecording Fi	es				Delete Selected		
		Caller ID	De	estination ID	Dat	e		Options	
	1	301	303	3	2016	5/01/04	18:25:43	Play Delete	6

Item	Explanation
Extension	Extensions that use one touch recording to record calls would be
	listed here.
Delete	Delete all recordings of the selected extension number.
Start Date/End	Search the recordings during this time period.
Date	
Delete Selected	Delete the select recording items.
Caller ID	Caller ID of this recorded call.
Destination ID	The number the caller called.
Date	The time when exactly this call began.
Play	Playback, delete and download options of the recording files.
Delete	Delete the recorded audio file.

6.3.4 Call Recording Playback

On IP PBX system, you have two ways to play back the recording files.

- Play back on the web interface
- Play back on a specific phone



By clicking the "Play" button on a call recording file, you'll see a dialog like below:

Play	Х
Туре 1:	
Type 2: Extension used for playing: 401	•
Extension used for playing. 401	
Play Cancel	

With "Type 1", you can click the lotton to play back the recording directly on the web interface.

With "Type 2", you can specify an extension number and click on "Play" to enable the extension to ring and the extension is answered that will play on the phone.

6.4 Call Logs

Call log is also known as CDR (Call Detailed Records). On the call logs page you can check any call records that go through the IPPBX system. Navigate to web menu *Report->Call Logs*. By specifying the time duration and/or Caller ID/Destination ID/Account, you can find out the logs you want.

Call Logs

Start Date: End Date:	Jan → 1 → 2016 → Jan → 10 → 2016 →	Field:	Caller ID 🔻	Download	Filter Delete
Call Start	Caller ID	Destination ID	Account Code	Duration(sec)	Disposition
2016-01-04 18:25:43	301 <301>	303		12	Answered
2016-01-04 18:25:38	301 <301>	301		0	Busy
2016-01-04 18:20:36	301 <301>	303		9	Answered
2016-01-04 18:19:57	301 <301>	303		0	No Answer
2016-01-04 16:46:56	303 <303>	305		0	No Answer
2016-01-04 16:03:20	802 <802>	801		9	Answered
2016-01-04 16:01:50	802 <802>	800		13	Answered
2016-01-04 16:00:42	802 <802>	800		0	No Answer
2016-01-04 16:00:51	802 <802>	801		0	No Answer
2016-01-04 15:37:46	802 <802>	800		0	No Answer
2016-01-04 15:36:59	802 <802>	801		0	No Answer
2016-01-04 14:52:06	801 <801>	800		0	No Answer

Item	Explanation
Start Date/End	Define the searching time period by "Start Date" and "End Date".
Date	
Field	Search criteria.



Item	Explanation	
Caller ID	Searching by the caller number.	
Destination ID	Searching by the called number.	
Account Code	Searching with the pin code had been used for outbound dialing.	
Download	Download the searching results.	
Delete	Delete the searching results.	
Call Start	The time when exactly this call began.	
Caller ID	The number of the caller.(By clicking on the number you can add this	
	number to the IPPBX system phone book.)	
Destination ID	The number which has been called. (By clicking on the number you	
	can add this number to the IPPBX system phone book.)	
Account Code	The pin code had been used for outbound dialing.	
Duration	The duration of this phone call.	
Disposition	How the calls been handled. Answered, No Answer and Failed.	

6.5 System logs

These logs are IPPBX journals which store all system activities. They can be used for debug purpose if the system is running into exception. Please do not enable these logs if the system is functioning properly, because there is a lot of data being generated and wrote into the logs files about every details of the system activities.

In the IP PBX system, there are 4 kinds of log files.

Item	Explanation
System Log	System Logs store all the system events.
PBX Log	PBX Logs store all the Asterisk events.
PBX Debug Log	Asterisk debug logs.
Access Log	Web and SSH access logs.

To enable these logs for the IPPBX system, please Navigate to web menu *Report->System Logs*. And enable the logs by ticking the corresponding checkboxes.

System Logs

System Logs			
	Enable System Log: Enable PBX Debug Log:	\$	Enable PBX Log: 🕜 Enable Access Log: 🖉
		Save Ca	ancel

After checking the checkboxes, please click "Save". The log files will be generated.



List Logs			Download Selected Del	ete Selected
		Name	Туре	Options
	1	debug20151221.log	Debug Log	Delete Download
	2	login201512.log	Login Log	Delete Download
	3	pbx20151221.log	PBX Log	Delete Download
	4	sys20151221.log	System Log	Delete Download

Each day there will be a new log file generated for each of the log types. Enable them only if you are familiar with these logs for troubleshooting.



Chapter 7. System

7.1 Time Settings

System time is very important for the IP PBX system. If the IP PBX system handles the inbound phone calls using time rule, then only the system time will correct the calls that can be handled properly. Beside call logs and debug logs, they record the system events using system time as well. The IPPBX system supports NTP (Network Time Protocol) and manual time set.

7.1.1 NTP

Navigate to web menu System->Time Settings.

By default, IP PBX system uses NTP to obtain time from Internet time servers. All you have to do is tell the IP PBX system where to find the server by specifying its domain or IP address. And also don't forget to select the correct time zone you are in.

Time Settings		
Time Settings		
	●NTP	©Manual Time Set
	time.stdtime Asia/Taipei	
	Sav	e Cancel

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



IPPBX can access Internet or please change an NTP server and try again.

7.1.2 Manual Time Set

If you want to manually set time for the IP PBX system or for some special reasons, the IP PBX cannot access Internet. You can choose to manually set the system time by checking the "Manual Time Set" radio button.

me Settings		
	ONTP	Manual Time Set
	Year:	(YYYY, eg: 2010)
	Month:	(MM, eg: 05)
	Day:	(DD, eg: 08)
	Hour:	(HH, eg: 09)
	Minute:	(MM, eg: 30)
	Synchron	ize with current PC time Sync

There are two ways to manually set a time to the system.

1. Manually write down the time and date info and click "Save".

2. Synchronize the IP PBX system time with your PC time by clicking the "Sync" button and then click on the "Save" button.

Once "Save" is clicked the time manually written or synchronized from the PC will be stored into the hardware clock chip on board the IP PBX motherboard.

7.2 Module Settings

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



Module Settings

SLOT 1		
	Module Type:	FXS/FXO/GSM/WCDMA 🔻
SLOT 2		
	Module Type:	FXS/FXO/GSM/WCDMA 🔻

7.3 Data Storage

Data storage allows you to upload the recorded files, log files and voicemail messages to an FTP server through the Ethernet.

7.3.1 Data Storage

With your existing FTP server you can configure the IP PBX to upload the call recordings, voicemails and call log files to your FTP server. If you don't have one you can even use your Windows PC to set up an FTP server for the IPPBX system to connect. Just make sure your PC is always turned on or at least by the time IPPBX is going to upload files you have to turn on your Windows PC.

Data Storage			
	Data Storage	Data Storage Log	
Data Storage			
		Enable: 🔽	
	Serve	r Address:192.168.1.48	
	1	Username: <mark>test</mark>	
		Password:	
		Directory:	
Aut	tomatically upload frequ	ency(day): 7 💌	
	Time of automatica	lly upload: 10 💙 : 00 💙	
Forcibly uploa	ad when the flash stora	ge is over: 70% 💙	
	Call Recording: 🗹 🛛 Vo	oicemail: 🗹 🛛 Call Logs: 🗹	
	Save	Cancel	
Status: Successfully	connect to FTP Server.		Upload Now

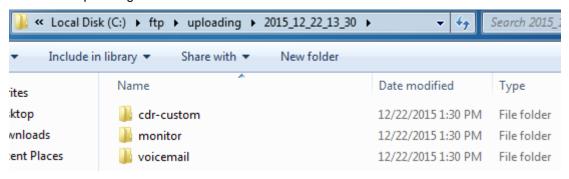
After these settings click "Save" to see the status "Successfully connect to FTP Server." You can click "Upload Now" to instantly upload a data.



Click on the "Data Storage Log" tab to enable the logs of each automatic data to upload as shown below.

Data Storage Log		
	Data Storage	Data Storage Log
Data Storage Log		Refresh Clear
Backup_Date_Time: FTP Backup Result Successfully uplo	:	

After each upload, you'll get a new folder on your FTP server directory named by the date and time of this uploading.





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



7.4 Management

7.4.1 Change Password

In the "Change Password" section, you are able to change admin password, also admin username can be changed by adding some extra letters following name string "admin".

Change Password
Username: admin
Password: •••••
New Username: admino01
New Password: •••••
Retype New Password: ••••••
Apply

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

7.4.2 Set System Voice Prompts

What's system voice prompts?

System voice prompts guide the callers to how to place a call or how to use the IPPBX system functionalities. For example, while checking voicemail the system voice prompts indicate the user to enter voicemail password and if nobody is answering a call, system voice will indicate leaving a message.

In the "Set Language" section you can set the language you want.

Set Language	
Set Voice Language: English * Download Delete	
Save	

For now, IP PBX system supports 22 different languages as the system voice prompts. They are English, English (Australia), Chinese, French, French (Canada), Spanish, Spanish (Mexico), Portuguese, Portuguese (Brazil), Italian, Persian, Arabic, Turkish, Thai, Russian, Polish, Dutch, Korea, Hungary, Vietnamese, Hebrew, Greek and German.

The items with * mean these languages are already existing in the system; others can be



downloaded here by clicking the "Download" button.

7.5 Backup

7.5.1 Take a Backup

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

Normally the first backup should be taken when you finish configuring the IPPBX to work for the very first time. And maybe later you'll apply new changes to the configurations in which you can take new backups as well.

Navigate to web menu *System->Backup*. Click the "Take a Backup" button to create a backup file which will contain all current system configurations.

Backup

	Backup	Upload Backup File				
List of Backups Take a Backup						
Name		Date	Options			
1 backup_2015nov	30_175928	Nov 30, 2015	Restore Delete 🛛			

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

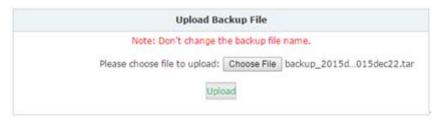


If you are downloading the backup to your computer hard drive, please keep this file confidential, because this file contains web admin password, user extension password and many other sensitive information which may compromise your IP PBX system.



7.5.2 Upload Backup File

Click on the "Upload Backup File" tab to enable to upload a backup file from your computer hard drive.





If you are uploading a backup from another IP PBX system, please make sure they have the exact same hardware configurations. It's not recommended to upload backup files to different IP PBX systems, unless you are pretty comprehensive with IP PBX systems

7.6 Troubleshooting

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

7.6.1 Ping

The ping command is a very common method for troubleshooting the accessibility of devices. It uses a series of Internet Control Message Protocol (ICMP) Echo messages to determine:

- Whether a remote host is active or inactive.
- The round-trip delay in communicating with the host.
- Packet loss.

Troubleshooting

Ping	Traceroute	Tcpdump	Channel Monitor
------	------------	---------	-----------------



Ping192.168.1.254Packets:ARunStopPING192.168.1.254(192.168.1.254):56data bytes64bytesfrom192.168.1.254:seq=0ttl=64time=5.77364bytesfrom192.168.1.254:seq=1ttl=64time=12.41164bytesfrom192.168.1.254:seq=2ttl=64time=3.63764bytesfrom192.168.1.254:seq=3ttl=64time=2.461

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

By specifying the domain or IP of the host and how many packets to be sent, click the "Run" button to enable the command to begin the process. You'll get results indicating the reachability of the destination.

7.6.2 Traceroute

The traceroute command is used to discover the routes that packets actually take when traveling to their destination. Click the "Traceroute" tab and specify the domain or IP address you want to look up and then click the "Run" button to start the process.

Troubleshooting

	Ping	Ping Traceroute		Channel Monitor	
Trace	eroute <u>8.8.8.8</u>	Rur	Stop		
trac	eroute to 8.	8.8.8 (8.8.8.8),	30 hops max, (60 byte packets	
1	192.168.1.25	4 (192.168.1.254) 0.523 ms 0.	.317 ms 0.810 ms	
2	210-61-134-2	54.HINET-IP.hine	t.net (210.61.1	134.254) 16.735 ms	16.685
3	tpe4-3302.hi	net.net (168.95.	229.86) 16.911	1 ms 16.864 ms 16.8	327 ms
4	211-22-226-1	.HINET-IP.hinet.	net (211.22.22)	6.1) 100.742 ms 100).679 I
5	209.85.243.3	0 (209.85.243.30) 33.506 ms 3	33.457 ms 72.14.233.2	20 (72.
6	209.85.242.1	63 (209.85.242.1	.63) 23.604 ms	209.85.252.161 (209.	85.252
7	209.85.243.2	3 (209.85.243.23) 24.911 ms 20	09.85.247.57 (209.85.	247.51
8	* * *				
9	google-publi	c-dns-a.google.c	om (8.8.8.8)	25.697 ms 25.660 ms	21.55

After the process, system will notice "Trace Complete" and you can see which routes the packets being taken before reaching the final destination.

7.6.3 Tcpdump

TCPDUMP is a common pocket analyzer that allows users to capture TCP/IP and other



packets being transmitted or received over a network to which the Planet IPPBX is attached. The captured packets can be downloaded from the IPPBX system and been analyzed on your Windows PC to display the SIP traffic details. It can be used to debug a VoIP call problem. On the 【System】->【Troubleshooting】->【TCPDUMP】 page, you can do a capture on one of Planet IPPBX Ethernet interfaces.

Tcpdump

	Ping	Traceroute	Tcpdump		Channel Monitor	
Tcpdump						
	Capture Trace on Adapter: WAN Duration(seconds): <u>20</u>(1-300) 					
Start						
List of Files 🍁 Delete Selected						
	Name			0	ptions	
	1 2016071	3113833.pcap		Delete	e Download	

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

7.6.4 Channel Monitor

Channel Monitor is technically a DAHDI Monitor that allows you to monitor signal level on analog channel and record the output to a file. Recorded audio files are by default raw signed linear PCM. You can play it to the speaker to listen to the phone call signaling on the analog channel. Or you can use a sound editor to visually display the audio level at both the Rx (audio Received by Asterisk) and Tx (audio Transmitted by Asterisk).

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



Channel Monitor

	Ping	Traceroute	Tepdump	Channel Monitor			
Chanı	nel Monitor						
		onitor on channel: uration(seconds):	FXS Port 3 20 (1-30 Start	•			
List of Files 🌵 Delete Selected							
	Name		Options No Files				

In the "Monitor on channel" field, you should select a channel to be monitored. And then you have to specify the duration to monitor. Then click on "Start" to enable the capture to begin. Now you should make a call in from this channel (port). After the capture is done, you'll get the file listed in the "List of Files" section.

7.7 Reset & Reboot

Navigate to web menu System->Reset & Reboot.

Reset & Reboot
Factory Defaults
Warning:All the configuration data will be lost when the system is reset to factory default. Please confirm that you have already backed up the configuration before reset. Keep the current network settings
Factory Defaults
Reboot
Warning: Rebooting the system will terminate all active calls!
Reboot

As you can see here on this page, you are able to reset and reboot the IPPBX system directly via web GUI.



7.7.1 Reset

By clicking the "Factory Defaults" button, you can reset all configurations of the IP PBX system. Except the configurations to be reset, the recording files, voicemail messages and call logs will also be erased. So please make sure you have backed up the files you need before resetting. The whole resetting process will be done in 2 minutes. If you choose to reset network settings also, then you need to login with the default URL https://172.16.0.1 on WAN. Username and password will all be reset to **admin**.

7.7.2 Reboot

By clicking "Reboot" you can restart the IPPBX system. The whole process will be done in 2 minutes.

7.8 Upgrade

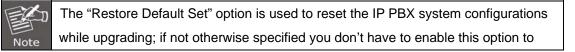
Planet will update the IPPBX firmware irregularly for new features and bug fixes. You can visit our office website <u>www.planet.com.tw</u> to check the updates for your IP PBX system. The downloaded firmware package should be in .zip format. Please extract the package first and upgrade with the ulmage-md5.xxx file to upgrade your IP PBX system. You can see there are two methods – Web upgrade and TFTP upgrade -- to upgrade the IPPBX firmware.

7.8.1 Web Upgrade

Upgrade

Upgrade System Package							
WEB Upgrade	C TFTP Upgrade						
Restore Default Set: Please choose file to upload:	Browse						
Uploa	d						

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





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

7.8.2 TFTP Upgrade

If you don't have a TFTP server, you can Google tftpd32 and download this application to set up a lightweight TFTP server on your Windows.

🏘 Tftpd.32 by	Ph. Jounin	
Current Directory Server interface Tftp Server Connection re Read request DACK: <tsize=< td=""><td></td><td><u>B</u>rowse Show <u>D</u>ir</td></tsize=<>		<u>B</u> rowse Show <u>D</u> ir
Clear	Current Action <a>(ulmage-md5.ipx330v2>: sent 6803 blk	.s, 3482655 bytes in 6 s. 0 blk
About	Settings	<u>H</u> elp

Please click "Browse" on the TFTP application window to locate the new firmware. And in the "Server Interface" dropdown list, it's a list of your PC network interfaces. Please select a correct interface (in the same network) which can access the IP PBX system. On the IPPBX web GUI please check the "TFTP Upgrade" radio button, and specify the exact firmware file name on the "Enter The Package Name" blank, and in the "TFTP Server IP address" blank please specify the IP address displayed on the TFTP application window.

Upgrade

Upgrade Sy	Upgrade System Package						
C WEB Upgrade	TFTP Upgrade						
Restore Default Set: Enter The Package Name:u TFTP Server IP address: A							

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



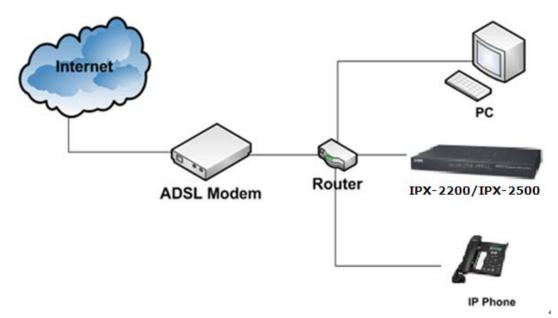


Chapter 8. Operating Instructions

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

8.1 How to connect the IP PBX to the Internet

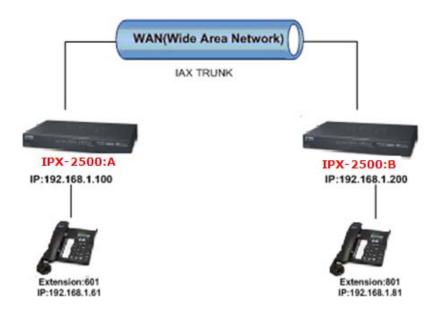
If your office accesses the public network through router, you can put Planet IP PBX behind the router. You should connect the WAN port of the IP PBX to the LAN port of the router.





8.2 How to combine two IP PBXs in a different network

Normally, two sets of the IPX-2500 are located in different places with different IP addresses for Internet access.



For external line configuration, you must use public IP address. Take the following instructions as an example:

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

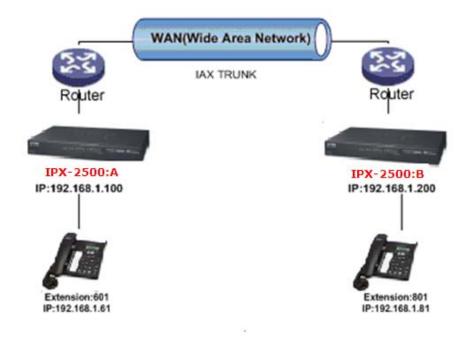
- 1. IP Phone registers on IPX-2500-A as extension 601.
 - 1. Another IP Phone registers on IPX-2500-B as extension 801.
 - 2. IPX-2500-A IP: 192.168.1.100.
 - 3. IPX-2500-B IP: 192.168.1.200.
 - 4. Extension format of IPX-2500-A: 6XX.
 - 5. Extension format of IPX-2500-B: 8XX
 - 6. Create an extension 888 with password 123456 on IPX-2500-B.
 - 7. All extensions on IPX-2500-A can call extensions on IPX-2500-B with format 8XX.
 - 8. All extensions on IPX-2500-B can call extensions on IPX-2500-A with format 6XX.

For detailed steps, please take Chapter 8.2 as reference.



Two sets of IPX-2500 behind router

Sometimes the IPX-2500 doesn't have a public IP address, and you have to configure port mapping for your router.



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

The IPX-2500-B is connected behind the router, and registers on IPX-2500-A through internet. You need to configure the port mapping of IAX2 port (4569) on the router. Then, all data received from RJ11 port of router (192.168.1.100:4569) will be sent to IPX-2500-A

Now, take the web management panel of ADN-4102 router as an example.



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

NAT	зегисе туре.									
> DMZ	• Usual Service Name:			AUTH 💌						
> Virtual Server	O User-defined Service Name:									
> ALG	Protocol:			TCP 💌						
> NAT Exclude IP	WAN Setting:			Interface V						
Port Trigger	WAN Interface			pppoe1	*					
FTP ALG Port										
Nat IP Mapping	WAN Port:		113 (ex. 5001:5010)							
	LAN Open Port:			113						
QoS	LAN Ip Addres	s:								
CWMP										
Port Mapping	Apply Changes									
☑ Others	S Current Virtual Server Forwarding Table:									
	ServerName	Protocol	Local IP A	ddress	Local Port	WAN IP Address	WAN Port	State	Action	
	IAX_TCP	tcp	192.168	.1.100	4569-4569	pppoe1	4569-4569	Enable	Delete Disable	
	IAX_UDP	udp	192.168	.1.100	4569-4569	pppoe1	4569-4569	Enable	Delete Disable	

Step 2: IPX-2500 Configuration

Configure the trunk and dial plan on IPX-2500-B, and register IPX-2500-B IP to IPX-2500-A. The configuration is the same as the above, but you have to replace the public IP address with the internal IP: 192.168.1.21.

Step 3: Configure port mapping rule of IPX-2500-B on the router Configure port mapping of IPX-2500-B on the router according to Step 1.

Step 4: Connect two sets of the IPX-2500 and make the call

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



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

8.3 How to resolve the problem about hearing one

side only

If the IPX-2500 is behind router, to resolve the problem, please set up IP address as shown below:

Click [Advanced] -> [Option] -> [Global SIP Settings] :



NAT Support

External IP:	
External Host:	
External Refresh(sec):	
Local Network Address:	

ItemExplanationExternal IPExternal IP or domain to replace the device IPExternal HostExternal domain to replace the device IP.External Refresh(sec)Refresh time, default is 10 secondsLocal Network AddressIP address and subnet mask needed to be converted.
e.g. 192.168.1.100/255.255.255.0

8.4 How to use soft phone in IPX-2200 or IPX-2500

8.4.1 Softphone on Windows PC

The softphones 3CX, Bria, Zoiper and many other softphone Apps all can work with IP PBX. Below is an example of registering Zoiper to IP PBX system as an extension from your Windows PC.

Step 1:

Dowload Zoiper from http://www.zoiper.com/.

Step 2:

Install and run Zoiper on your Windows.

Step 3:

Click menu "Settings" and select "Create a new account" and select "SIP" protocol and click Next.



Step 4:

Fill in the register credentials shown below.

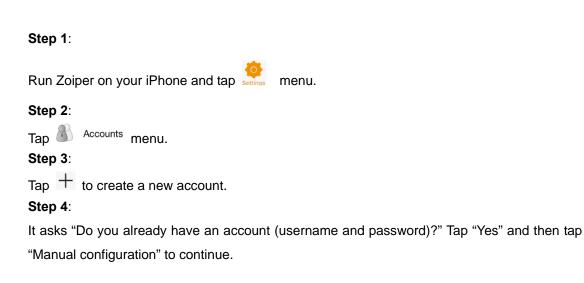
Account wizard	8
Credentials	
user / user@host 402	
Password ••••••	
Domain / Outbound proxy 192.168.1.254	
ACK NEXT -	,

Step 5:

Click Next to complete registering.

8.4.2 Softphone on Android Phone, iPhone or iPad

Most of the softphones mentioned previously have mobile editions for both Android and iOS platforms. You can download to install from your mobile phone App Store. Below is how you register Zoiper softphone to IP PBX as an extension from your iPhone:





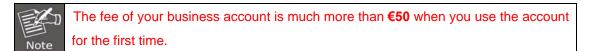
Step 5:

Tap ቆ	SIP account	to configure the acc	count she	own below:
		••••••	15:18	I00%
		Accounts SIP A	Account	
		Account name:	403	
		Domain:	192.168	.1.254
		User name:	403	
		Password:	•••••	,
		Caller ID:	403	

Step 6:

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

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





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	
Planet.com	
Forgotten your Skype Name?	Safe & Secure Quick & Easy
Password	Manage your account
	Change your settings
Forgotten your password?	

Sign me in

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

		\square	Payment s	etting	gs	Stored	l payment de	etails and Au	uto-re	char	je settii	ngs. Viev	w det
		¢	Skype Mar	ager	-	You ar	re the admin	istrator of P	lanet	. Sky	pe Mar	nager · M	emb
m		×××	Redeem v	ouche	er	Redee	em your voud	cher or prepa	aid ca	ard. R	edeem	ı	
		(î:	Skype WiF	i		Learn	about Skype	e WiFi					
secret. rd	David Yao			Se	ttings and ex	tras							
	David Yao Your Skype Planet.com				ttings and ex Payment settings	tras	Stored payment (details and Auto-re	echarge	e setting	ıs. View de	tails	
	 Your Skype				-	tras		details and Auto-re set to EUR (Euros)	-	-	s. View de	etails	
	 Your Skype Planet.com	ils			Payment settings	tras	Your currency is). Chang	ge			



3 Please click the Skype connect

Your features	Your members	•
Some features have been suspended	Your Skype Manager has 2 members	N
S Allocate Skype Credit to your members	Add members	th
Set up Subscriptions for your members	Since you last signed in No changes since you last logged in.	s C M
Set up Group video calling for your members	Still unresolved	e. S
Set up Online Numbers for your members	One unresolved invite	fc
Set up Call forwarding for your members		М
Set up Voicemail for your members		
7 profiles set up for Skype Conflect 9		



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.

Connect your existing SIP-enabled PBX to Skype with Skype Connect. Learn more

Your SIP Profiles

Set up a SIP Profile

档案2 View profile

Ch-----



4 Create a SIP profile

Create a SIP pro	ofile
1 Choose name	2 Set up subscription 3 Authentication
	s as easy as three steps. Simply choose a name for your profile, purchase a channel our authentication details.
Choose a profile name	
Choose a profile name	

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.

	Profile settings	
aaa	Profile name	888
Profile settings	Calling channels	Buy a channel subscription to activate this profile
Authentication details	Outgoing calls	Set up outgoing calls
Reports		To make outgoing calls from this SIP Profile you need to add Sk
« Back to SIP Profile list		You can also set up Auto-recharge so you never run out of credi call. Outbound calls to landlines and mobiles in the US* are cha cents/min. For all other destinations see Skype's standard per r rates.
		Add credit Auto-recharge settings
		S € 0.30 Add credit



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

aaa	Authentication details	S of authentication needed for your PBX.
Profile settings	Registration	or, IP Authentication 🔗
Authentication details	(Username/password)	
Reports	SIP User	Skype user name
	Password	Skype password Generate a new password
« Back to SIP Profile list	Skype Connect address	sip.skype.com
	UDP Port	5060
	🛕 SIP user is not yet registered	at sip.skype.com

5 Settings on IP PBX

5.1 Build one SIP trunk with Skype for SIP account

Provider Type: Custom Trunk

Host: sip.skybe.com

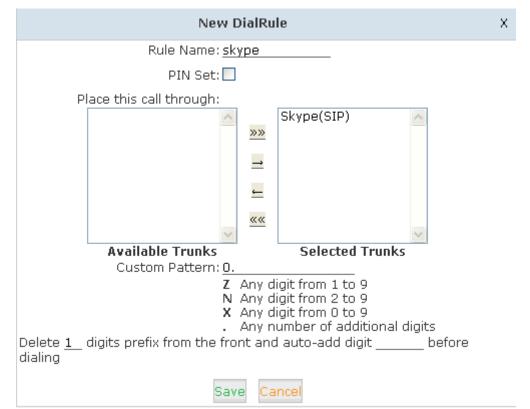
User name: the user name you defined in Authentication detail

Password: the password you defined in Authentication detail

	New VoIP Trunk		×
Description:	Skype		
Protocol:	SIP 💌	5000	
Host:	sip.skype.com	:5060	
Maximum Channels*:	<u>0 </u>		
Prefix:			
Caller ID:			
Without Authenticati	on		
Username: Skype user	name		
Authuser: Skype pass	sword		
Password: ••••••••	••••		
Advanced Options			
	Save Cancel		



5.2 Set one outbound rule



Edit		х
DialPlan Name: <u>DialPlan1</u> Include External Calling Rules Skype	Include Internal Calling Rules Extensions Spy Conference Ring Groups IVR Call Queues Paging and Intercom Directory DISA	

5.3 Make an outbound call

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

For example, dialing number 00 (outbound prefix number) + 001 (International Code) + 886



(Country code) + 2 (city Area code without 0) + 22199518 (local phone number) will enable you to contact Taiwan Planet Company

5.4 Set inbound rule

New Number DID		×	
DID Number: Destination:	Skype number Goto IVR Save	vorking time v	