Internet Telephony PBX System

IPX-1900

User's manual

Version 1.0.1

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CE mark Warning

The 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.

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Revision

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Chapter 1 Introduction

Overview

PLANET IPX-1900 IP PBX telephony systems are designed and optimized for the small business in daily communications. The IPX-1900 are able to accept 300 user registrations, and easy to install and manage a fully working system with the convenience and cost advantages. The PLANET IPX-1900 is also designed to operate on a variety of VoIP applications; it provides centralized call control, auto-attendant, voice conferencing, and PSTN access, digital and IP-based communications.
Based on state-of-the-art embedded technology, the IPX-1900 provides a solid, uniform platform for voice communications as well as data network communications. The IPX-1900 offers a seamlessly integrated solution for the up-to-date telecommunication needs. The future IP PBX telephony system offers all of the essential features of telephony which is required by small business/enterprise users for their telecommunication/data needs.

Being more flexible, the IPX-1900 integrates up to 4 calls via the IPX-19FO (2*FXO) / IPX-19FS (2*FXS) / IPX-19SL (1FXO+1FXS) module to become a feature-rich PBX system that supports seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints.

The IP PBX is the feature-rich SIP based IP PBX telephony system that integrates NAT functions to make it perfect for small business usage. The IP PBX integrates traditional PBX system functions and provides many advanced functions including voice mail to email, web management etc.
Designed to run on a variety of VoIP applications, the IP PBX provide IP-based communications, voice conferencing, and call detailed record (CDR), centralized Auto-Attendant (AA), and Interactive Voice Responses (IVR). The IP PBX utilizes standard PSTN/GSM lines via the interfaces of FXO/GSM gateway to become a feature-rich IP PBX telephony system that supports seamless communications among existing local calls, SIP-based endpoints including low cost of long distance service, telephone number portability and one network for both voice and data.

With the IP PBX, standard SIP phones can be easily integrated in your office. Users may integrate PLANET IP Phone VIP-254T series, VIP-255PT/ 350PT/ 351PT, the ATA-150 / VIP-156/ 157/ 158/ 161W of ATA (analog telephone adapter) series, the VIP-191/ 192 of Wi-Fi Phone, and Gateway series VIP-281/ 281GS/ 480 to build up the VoIP network deployment in minutes. Allowing distributed IP technology to meet traditional voice services with proactive managed interface, the IP PBX for enterprises in the daily business processes can make people more productive, more intelligent tasks and more customer satisfaction.

IP PBX Features

- **PBX Features**
  Automated Attendant (AA)
  Interactive Voice Responses (IVR)
  Voicemail support (VM)
  Call Detailed Record (CDR)
  User Management via Web Browsers
  Display 300 Registered User's Status: Unregistered / Registered / On-Call
  Multiple Service Providers Lines / SIP Accounts (30)
  Simultaneous Trunk Links: 30 concurrent trunk calls
  SIP Trunk / Gateway Trunk / FXO Trunk Management
  Two-stage / One-stage call to Trunk by Trunk Group Configuration
  Build in 2 / 4 FXO PSTN trunk (Modular)
  By adding external FXO analog gateway to use Terminal trunk Line
  By adding external GSM VoIP gateway to use GSM trunk line
  Built-in SIP Proxy Server Following RFC-3261
  Support password authentication using MD5 digest and RFC2833 for DTMF Relay

- **Call Features**
  Call Forward Immediate
  Call Forward on Busy
  Call Forward on No Answer
  Call Pickup / Call Park
  Call / Pickup Group
  Caller ID / T.38 FoIP
  Music on Hold / Music on Transfer
  Call Transfer / Call Hold / Call Waiting
  Three-way conference with feature phones

- **Router Features**
  DHCP Server for LAN Users
  Packet / URL Filter Virtual Server / DMZ/ Port Trigger
  Static Route
  NAT/Bridge mode
  UPnP

Package Content

The contents of your product should contain the following items:

Internet Telephony PBX system unit

Power Adapter

Quick Installation Guide

User's Manual CD

RJ-45 Cable

RS-232 Cable

Rack mount brackets

Physical Details

The following figure illustrates the front/rear panel of IP PBX.

Front Panel Indicators

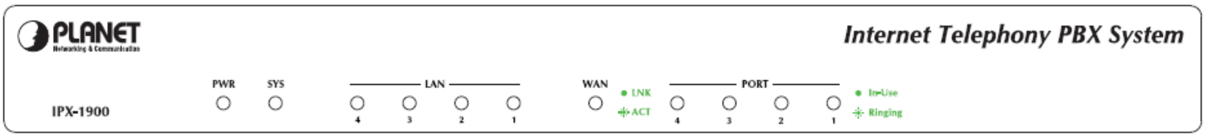


Figure 1-1. Front Panel of IPX-1900

Front Panel LED	State	Descriptions
PWR	On Off	PBX Power ON PBX Power OFF
SYS	On Flashing	System is booting System is ready
LAN	On Flashing Off	LAN is connected successfully Data is transmitting Ethernet not connected to PC
WAN	On Flashing Off	PBX network connection established Data traffic on cable network Waiting for network connection
FXO/FXS Port	On Flashing Off	Port is busy Ring indication. (FXS only) Port is not enabled.

Table1-1. Front Panel description of IP PBX

Rear Panel Indicators

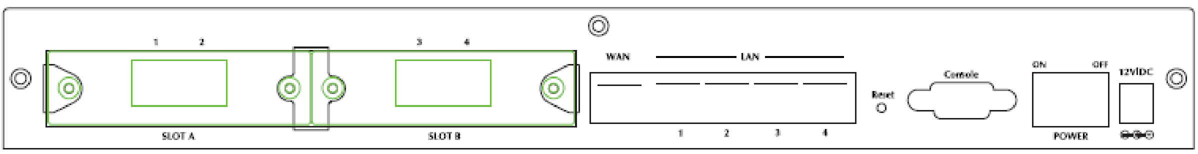


Figure 1-2. Rear Panel of IPX-1900

1	**12V DC**	12V DC Power input outlet
2	**Reset**	The reset button, when pressed, resets the IP PBX without the need to unplug the power cord.
3	**WAN**	The WAN port supports auto negotiating Fast Ethernet 10/100Base-TX networks. This port allows your IP PBX to be

		connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a CAT.5 twisted pair Ethernet cable.
4	**LAN**	The LAN port allows your PC or Switch/Hub to be connected to the IP PBX through a CAT.5 twisted pair Ethernet cable.
5	**Slost A/B**	2 external slosts with compliance FXO/FXS module. FXO module is connects to PBX or CO line with RJ-11(Write) analog line. FXO port was connected to the extension port of a PBX or directly connected to a PSTN line of carrier FXS module is connects to Phone with RJ-11 (Black) analog line. FXS port was connected to your telephone sets, FAX, or Trunk Line of PBX.
	FXS Port (Modular IPX-19FS)	Connect to Phone with RJ-11 (Black) analog line. FXS port was connected to your telephone sets, FAX, or Trunk Line of PBX.
	FXO Port (Modular IPX-19FO)	Connect to PBX or CO line with RJ-11(Write) analog line. FXO port was connected to the extension port of a PBX or directly connected to a PSTN line of carrier
	Note : IPX-19SL 2-Port PBX Life Line Module IPX-1900 (1FXO, 1FXS)	

Table 1-2. Rear Panel description of IP PBX

Chapter 2
Preparations & Installation

Physical Installation Requirement

This chapter illustrates basic installation of IP PBX

- Network cables. Use standard 10/100Base-TX network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem (for WAN port usage)

Administration Interface

PLANET IP PBX provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access:

To start IP PBX web configuration, you must have the web browsers installed on computer for management

- Microsoft Internet Explorer 6.0.0 or higher with Java support

Default LAN interface IP address of IP PBX is **192.168.0.1**. You may now open your web browser, and insert ***192.168.0.1*** in the address bar of your web browser to logon IP PBX web configuration page.

IP PBX will prompt for logon username/password, please enter: ***admin*** / ***123*** to continue machine administration.

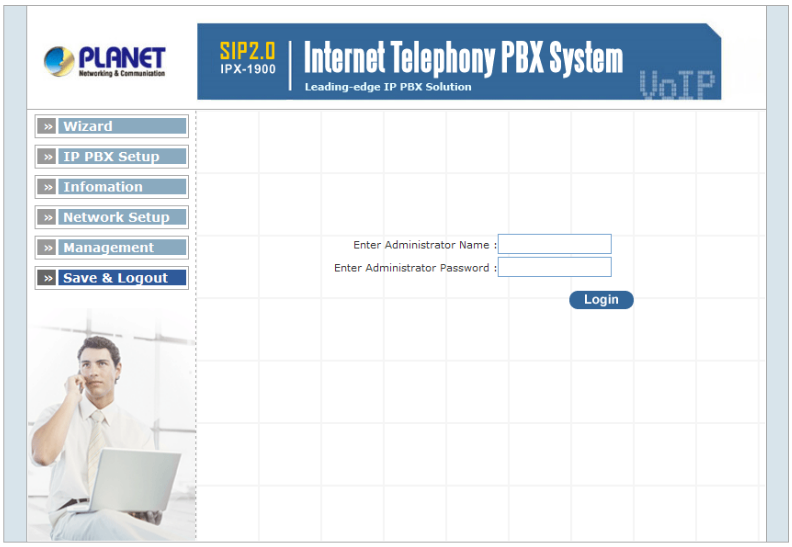


Figure 2-1. Input prompt

Note

In order to connect machine for administration, please locate your PC in the same network segment (192.168.0.x) of IP PBX. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

Network Interface quick configurations

Wizard is a tool to quickly setup IP PBX.

After pass the authentication, please click » Wizard for quick IPX PBX setup.

For most users, Internet access is the primary application. The IP PBX supports the WAN interface for Internet access and remote access. The following sections will explain more details of WAN Port Internet access and broadband access setup. When you click "Wizard Setup "the following setup page will be show.

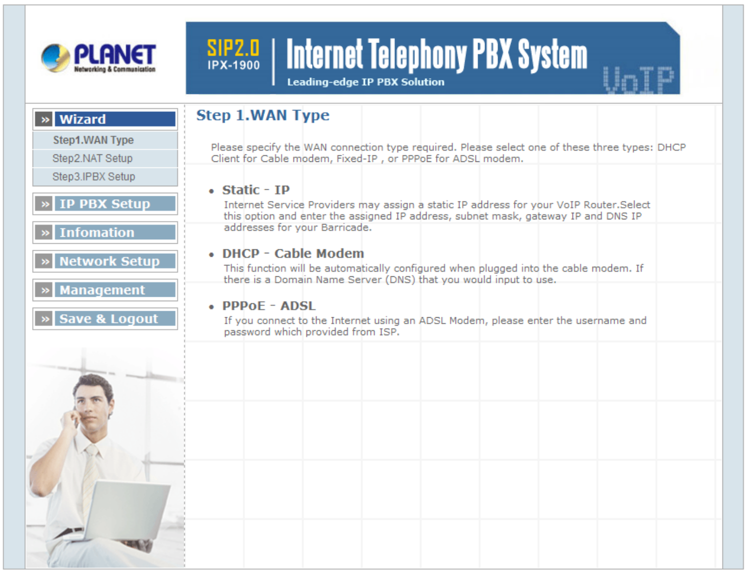


Figure 2-2. Wizard-Operating Mode settings

- **Step1. Wan Type**

WAN Setting

Static - IP	If you are a leased line user with a fixed IP address, fill out the following items with the information provided by your ISP.
DHCP – Cable Modem	This function will be automatically configured when plugged into the cable modem. If there is a Domain Name Server (DNS) that you would input to use.
PPPoE - ADSL	Some ISP's provide DSL-based service and use PPPoE to establish communication link with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to select this item.

Table 2-1. WAN description of IP PBX

➢ Step2. NAT Setting

LAN IP Setting

LAN IP Address	Private IP address for connecting to a local private network. (Default: 192.168.0.1)
Subnet Mask	Subnet mask for the local private network (Default: 255.255.255.0)
DHCP Server	Enable to open LAN port DHCP server
Assigned DHCP IP Address	DHCP server range from start IP to end IP
DHCP IP Lease Time	Client to ask DHCP server refresh time, range from 60 to 86400 seconds

Table 2-2. LAN IP description of IP PBX

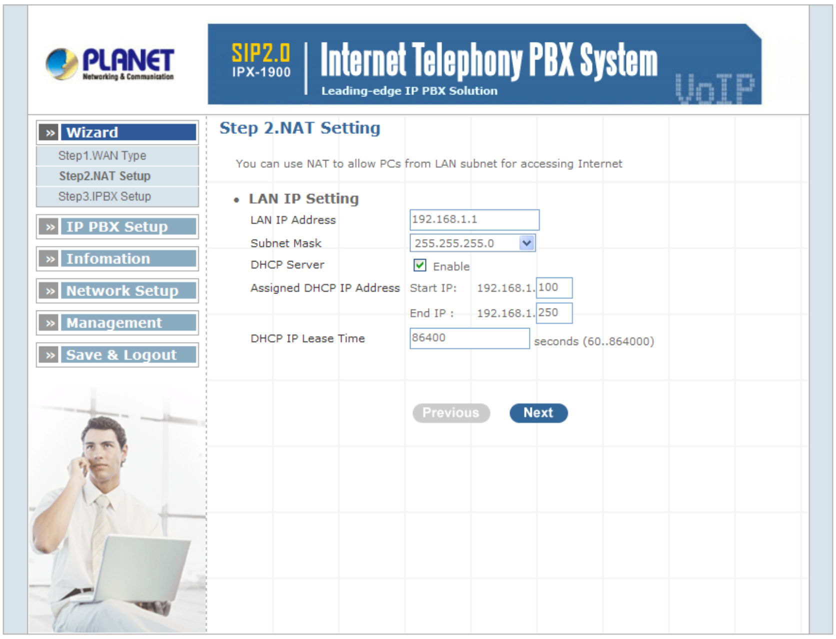


Figure 2-3. Wizard-NAT settings

➢ Step4. IPPBX Setup

The IP PBX allows multiple ITSP providers / User Extensions registration by simply fill-in the required information in the provided table.

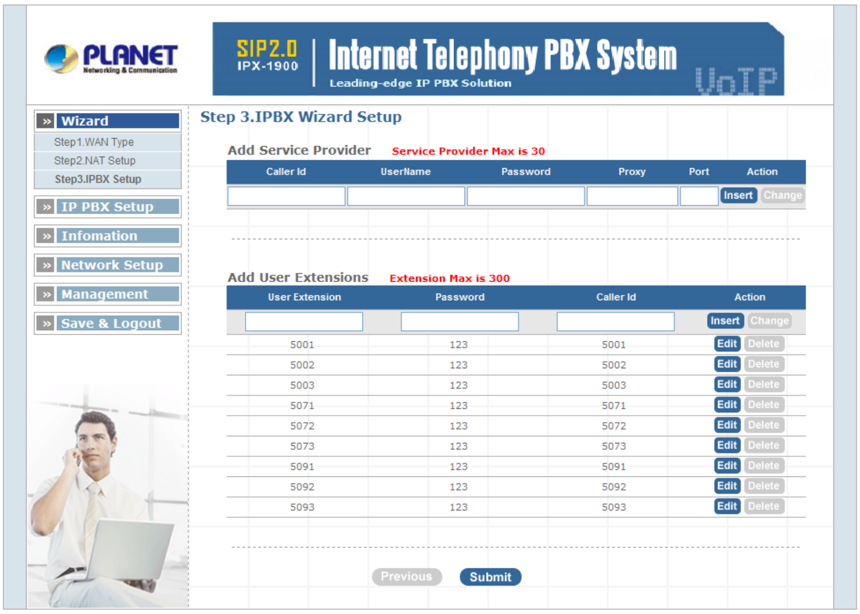


Figure 2-4. Wizard-IP PBX settings

Service Provider:

Caller ID	Service provider name
Username	Input Provider name
Password	Input Provider password
Host	Input Providers server address
Port	Providers server port

Table 2-3. Service provider description

User Extensions:

User Extension	Input Extension number
Password	Input Extension password
Caller Id	Input Extension caller id

Table 2-4. User extension description

After completing the wizard setup, click "**Submit**" button, The IP PBX will save configuration and reboot IP PBX automatically, after 50 seconds, you can re-load setting page again.

Figure 2-5. Wizard-Rebooting

Note

```
Please consult your ISP personnel to obtain proper PPPoE/IP
address related information, and input carefully.
If Internet connection cannot be established, please check
the physical connection or contact the ISP service staff
for support information.
```

RS-232 Console Port Configuration

RS-232 port (DB-9pin Female connector), Configure the COM Port Properties as following:

Bits per second: **57600**, Flow control: **None**

1. Connect Gateway RS-232 port to PC COM Port.
2. Power on gateway.
3. Open Terminal Program (ie. Windows XP Hyper Terminal)

   [Start] → [Program file] → [Accessories] → [communications] → [Hyper Terminal]

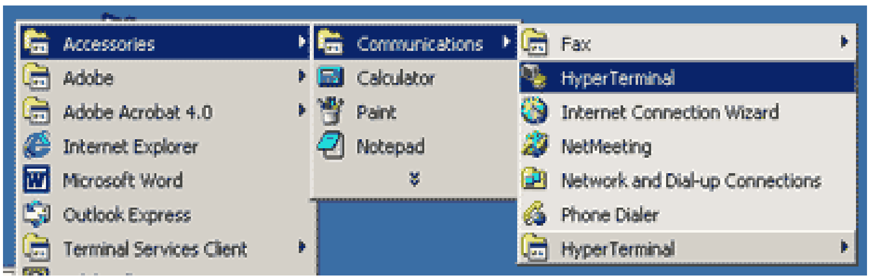


Figure 2-6. Windows Hyper Terminal Path

4. Create new connection. Select “COM” port that connect PC to gateway

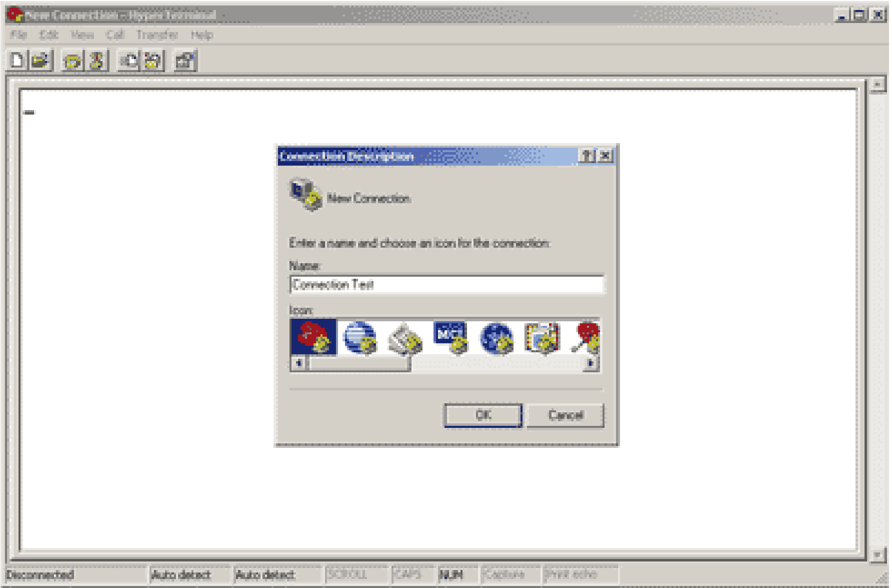


Figure 2-7. Hyper Terminal Screen

5. Make connection(Bits Pre second:**57600** Flow contact: **None**)
6. Input “Enter” and Show Welcome display.
7. Login, input the Username and Password to login.(Please contact with our VoIP Technical Support Team: support_voip@planet.com.tw for the login account and the further assistances.)

Chapter 3
IP PBX Setup

SIP Basic Setting

SIP (Session Initiation Protocol) is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol. SIP determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

➢ SIP Configuration

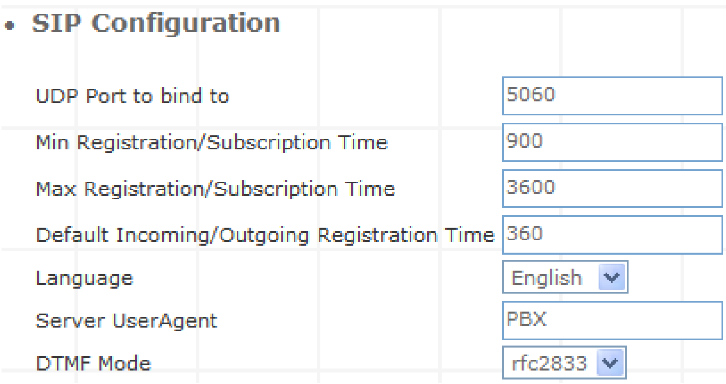


Figure 3-1. SIP configuration settings

UDP Port to bind to	This is SIP Local Port 5060, if you have any specific reason for change this port.
Domain	IP PBX Server's IP address.
Max Registration Time	Maximum duration of incoming registration/subscriptions we allow. Default *3600 seconds.*
Min Registration Time	Minimum duration of registrations/subscriptions. Default *60 seconds*
Default Incoming/Outgoing Registration Time	Default duration (in seconds) of incoming / outgoing registration.
Language	Set default language for all users.
Server UserAgent	Enable you to change the trunk User agent string, Default is *PBX.*
DTMF Mode	Set default DTMF mode for sending DTMF. Default: *rfc2833.*

Table 3-1. SIP configuration description

➢ SIP Codecs

The Codec is used to compress the voice signal into data packets. Each Codec has different bandwidth requirement. There are 7 kinds of codec. To determine the priority, selects one codec algorithm from the pull-down menus individually.

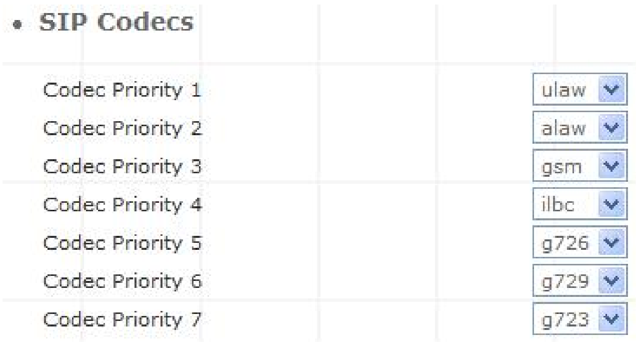


Figure 3-2. SIP codecs settings

➢ Outbound SIP Registrations

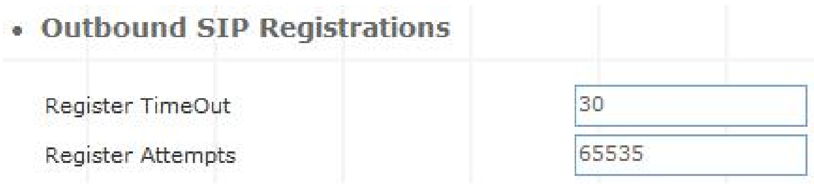


Figure 3-3. Outbound SIP Registrations settings

Register TimeOut	Retry registration calls at every 'x' seconds (default 20).
Register Attempts	Number of registration attempts before we give up; 0 = continue forever.

Table 3-2. Outbound DIP registration description

➢ NAT Support

The *externip*, *externhost* and *localnet* settings are used if you use IP PBX behind a NAT device to communicate with services on the outside.

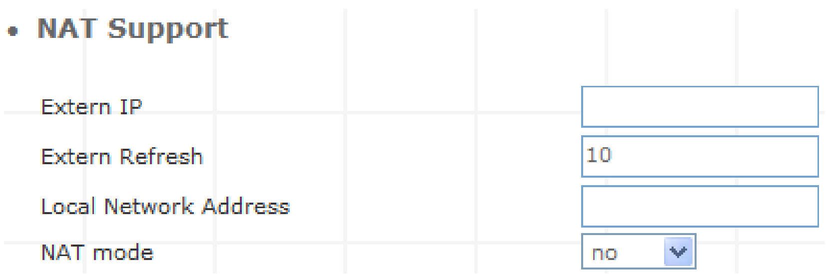


Figure 3-4. NAT support settings

Extern IP	Address that we're going to put in outbound SIP messages if we're behind a NAT.
Extern Refresh	How often to refresh externhost if used. You may specify a local network in the field below.
Local Network Address	localnet=192.168.0.0/255.255.0.0; All RFC 1918 addresses are local networks localnet=11.0.0.0/255.0.0.0 ; Also RFC1918 localnet=171.16.0.0/12 ; Another RFC1918 with CIDR notation localnet=168.254.0.0/255.255.0.0; Zero conf local network

Table 3-3. NAT support description

SIP Extension

This page allows the users to add /edit /delete extensions in the VoIP telephony network.

- **Extension List**

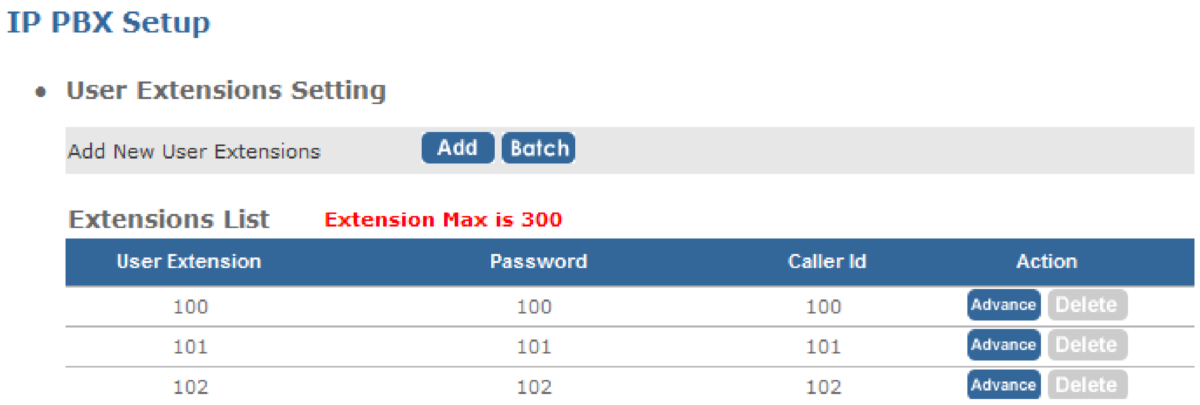


Figure 3-5. User extension settings

Advance	Click Advance to edit an extension other setting.
Delete	Click Delete to delete an extension.

Table 3-4. User extension description

➢ Advance Setup

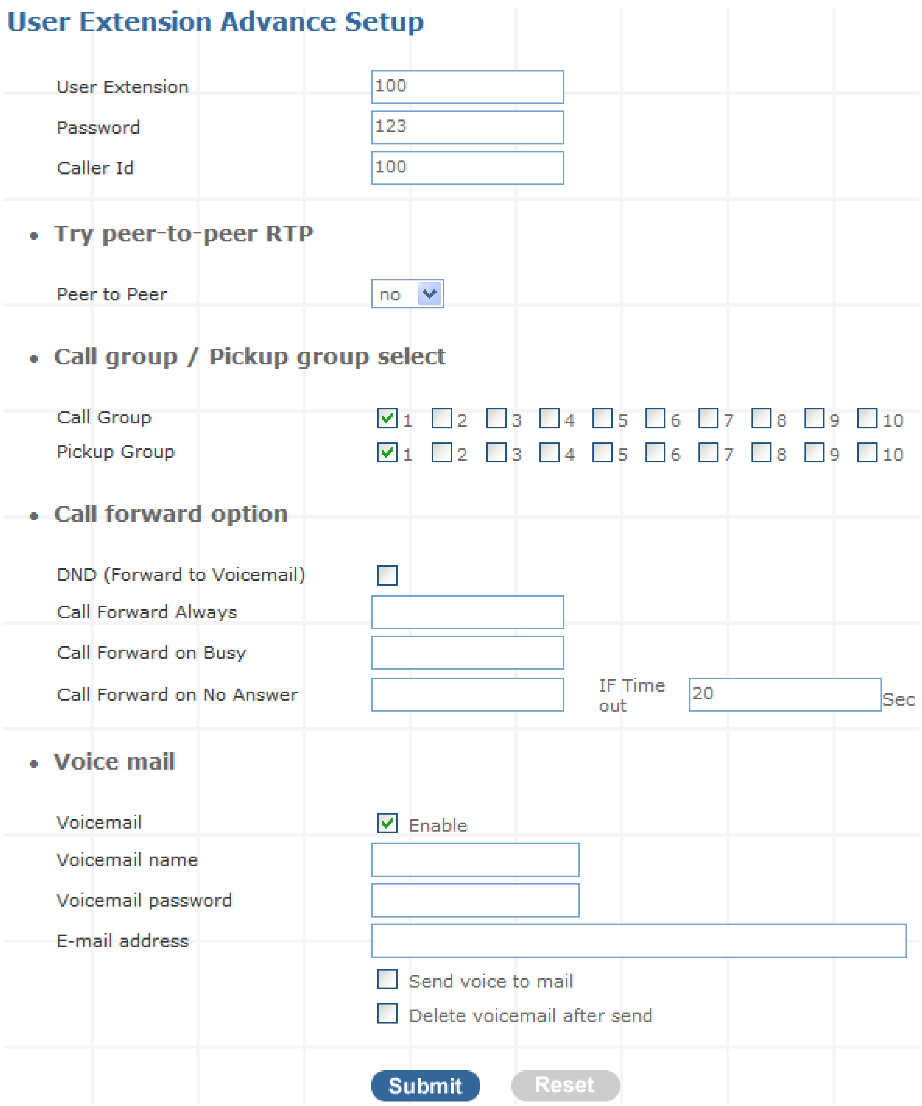


Figure 3-6. Extension advance settings

User Extension	Input Extension number
Password	Input Extension password
Caller Id	Input Extension caller id

Table 3-5. Extension advance description

- **Try peer-to-peer RTP :**

If select yes, allow RTP transmission to try peer-to-peer for sip extension device between.

- **Call group / Pickup group select :**

Call Group	An Extension can set single/multiple call group(s) 1-10 id
Pickup Group	An Extension can set single/multiple Pickup group(s) 1-10 id

Table 3-6. Call / Pickup group description

- **Call forward option :**

DND(Forward to Voice mail)	Enable / Disable forward to voice mail.
Call forward always	Input forward always number
Call forward on busy	Input forward on busy number
Call forward no answer	Input forward no answer number
If time out "XXX" sec	This is the maximum number allowed no answer time out used

Table 3-7. Call forward description

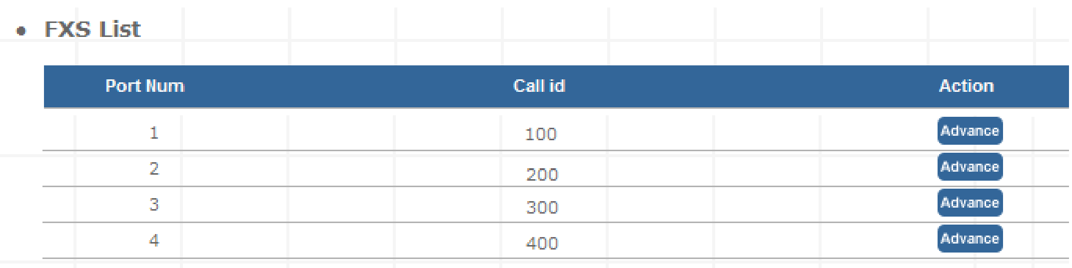
- **Voice mail :**

Voice mail select	Enable / Disable voice mail function
Voice mail name	Input voice mail name
E-Mail address	Input E-mail address
Send voice to mail	Enable / Disable send voice to mail
Delete voice mail after send	Save / Delete voice mail after send

Table 3-8. Voice mail description

FXS Extension

FXS (Foreign Exchange Station) port can be connected to analog telephone sets or Trunk Line of PBX.



• FXS List

Port Num	Call id	Action
1	100	Advance
2	200	Advance
3	300	Advance
4	400	Advance

Figure 3-7. FXS Extension settings

Click Advance to edit an Analog extension setting, fill in the required information in "User Extension Advance Setup "and click **Submit** to activate changes.

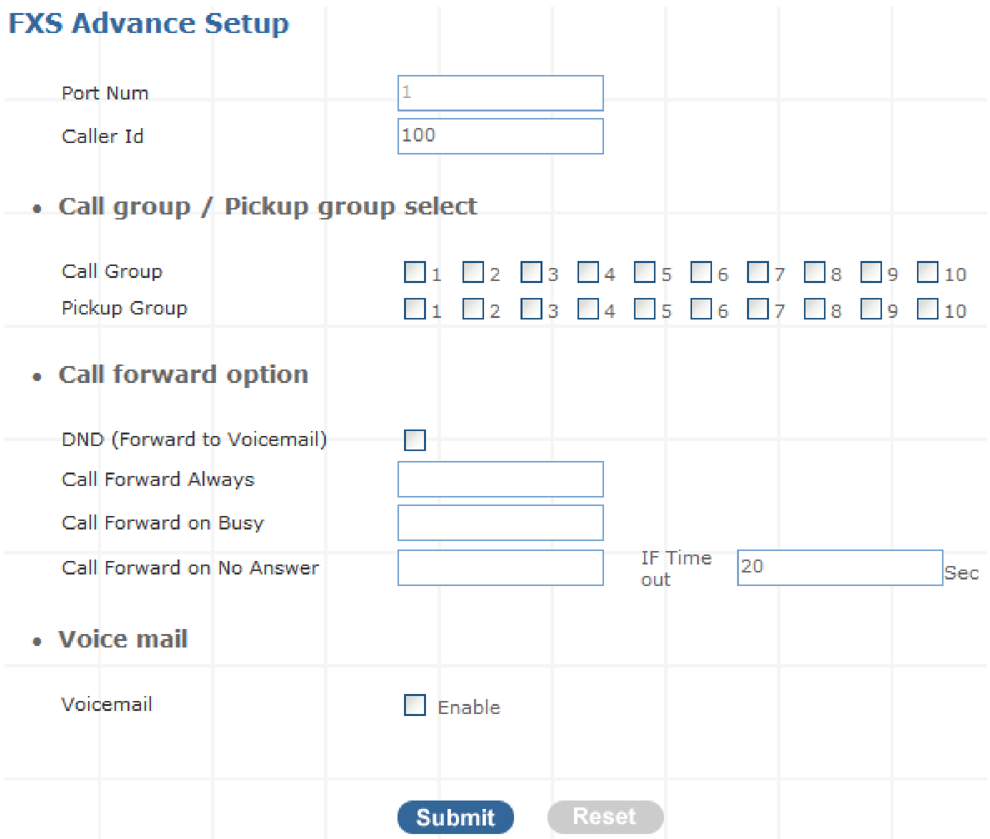


Figure 3-8. FXS Extension advance settings

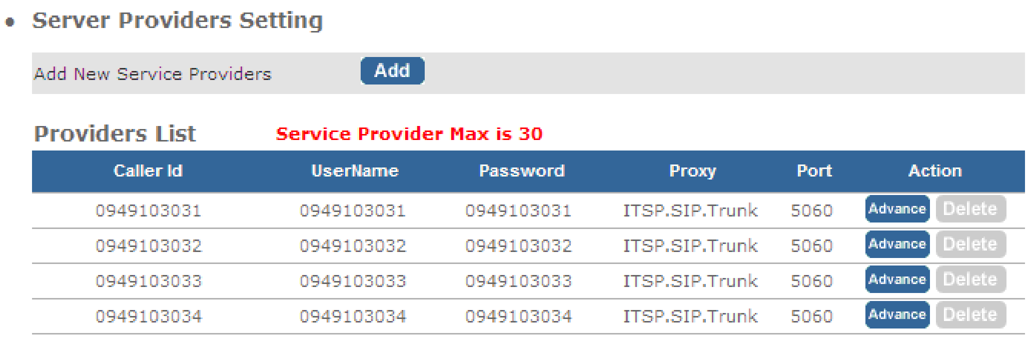
Port Num	Analog Port Number (System Define).
Caller id	Input Extension caller id

Table 3-9. FXS Extension advance description

SIP Trunk

Services Providers Setting allows IP PBX register to different SIP systems and ITSP Services (SIP Trunk).

On the “**Providers List**”, you can press “**Add**” to add a new service provider or press “**Advance**” to edit the information of specific Service Provider or press “**Delete**” to delete the specified service provider information. **Maximum 10 registrations on Server Provider list**



Caller Id	UserName	Password	Proxy	Port	Action
0949103031	0949103031	0949103031	ITSP.SIP.Trunk	5060	Advance Delete
0949103032	0949103032	0949103032	ITSP.SIP.Trunk	5060	Advance Delete
0949103033	0949103033	0949103033	ITSP.SIP.Trunk	5060	Advance Delete
0949103034	0949103034	0949103034	ITSP.SIP.Trunk	5060	Advance Delete

Figure 3-9. Server Providers Setting

➢ Add New Service Providers

Step 1. Press “**Add**” button to add an new service provider information.

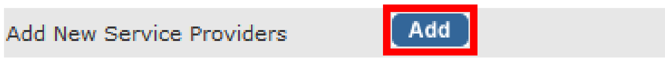


Figure 3-10. Add new service providers

Step 2. Fill in the required information in Service Provider Advance Setup page.

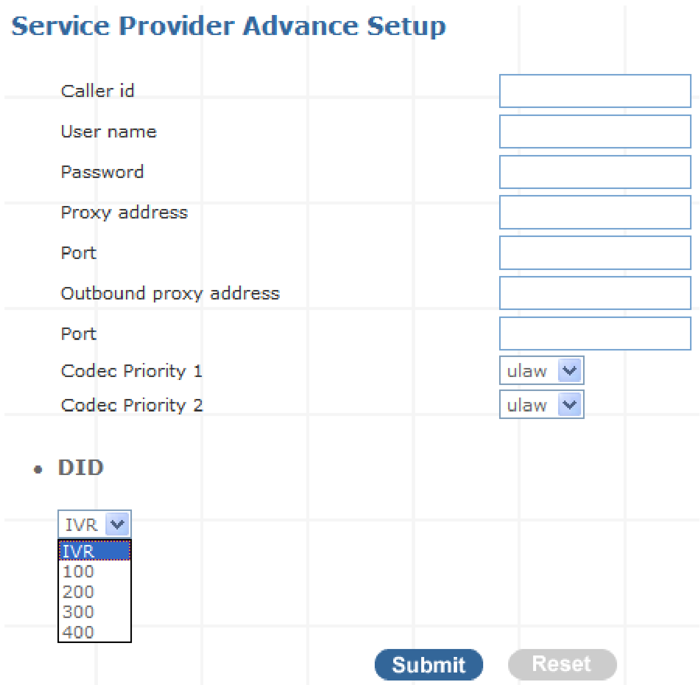


Figure 3-11. Service provider advance setup

Caller id	The caller ID will be sent between the callee and caller and will be displayed on SIP device LCD panel for identification.
User name	User name for authentication
Password	User password for authentication
Proxy Server address	Assigns the SIP Proxy Server’s IP address / Domain name
Proxy Server Port	Port number of SIP Proxy Server. Assigns a value from 1024 to 65535, the common default SIP port is 5060.
Outbound Proxy Address	Outbound Proxy server’s IP address / Domain name. Assign a server’s IP / Domain name which is in charge of call-out service.
Outbound Proxy Port	Port number of Outbound Proxy Server. Assign a number from 1024 to 65535, the common default SIP port setting is 5060.
Codec Priority 1	Set allow codec priority 1

Codec Priority 2	Set allow codec priority 2
DID	Choose a direct ring extension or a hunt group or hear the IVR voice prompt, default is “IVR”. (For how to make hunt group please refer “Hunt Group Setting”)

Table 3-10. Service provider advance setup description

FXO Trunk

FXO (Foreign Exchange Office) **Trunk Setting**, can be Connected to PBX or CO line with RJ-11 analog line. FXO port can be connected to the extension port of a PBX or directly connected to a PSTN line of carrier

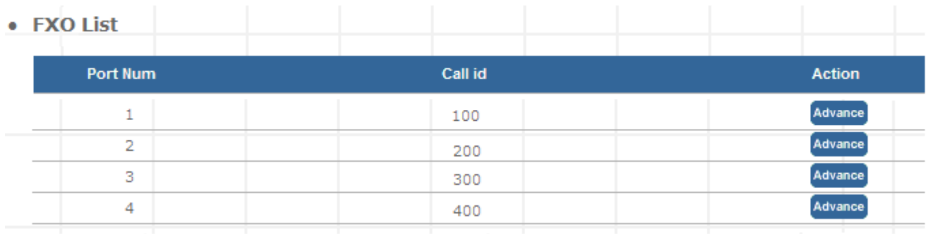


Figure 3-12. FXO Trunk setting

Press “Advance” to Edit an FXO Prot as below

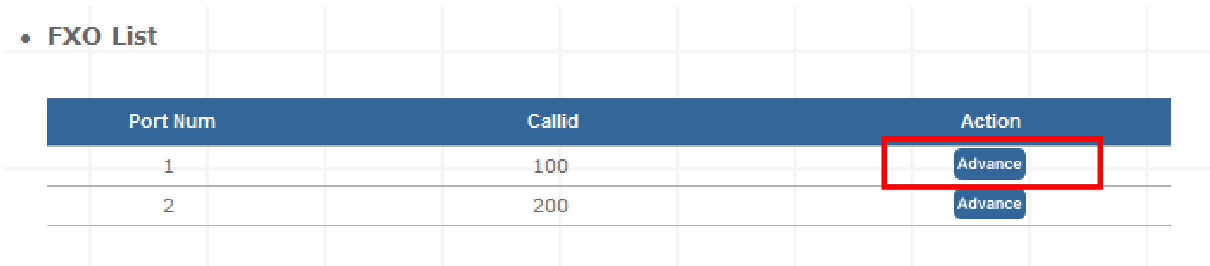


Figure 3-13. FXO Trunk list

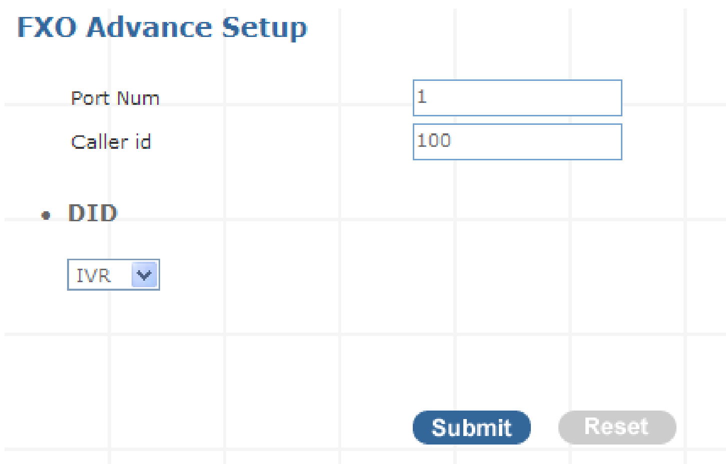


Figure 3-14. FXO Advance setting

Port Num	Analog Port Number (System Define)
Caller id	The caller ID will be sent between the callee and caller and will be displayed on SIP device LCD panel for identification.
DID	Any calls originating from the registered ITSP to IP PBX will go into the auto-attendant or direct to the selected user or hunting group.

Table 3-11. Trunk Management - FXO Trunk setup description

Gateway Trunk

Gateway Trunk Setting allows IP PBX makes VoIP calls to external Gateway by peer-to-peer mode. If the FXO ports of external Gateway have connected with PSTN lines, the user can make outgoing PSTN calls via external Gateway by this function.

Figure 3-15. Gateway Trunk setting

IP	Destination IP Address is the IP address of the destination Gateway that owns this phone number.
Port	Port is port of the destination Gateway use. (Default is 5060)

Table 3-12. Gateway Trunk setting description

Trunk Group

Trunk Group is defines the leading digit of the call out dialing number through SIP / FXO / Gateway Trunks of the same type between two given points. The IP PBX will in according to the leading digit to determine to use which SIP or Gateway Trunks for outgoing route.

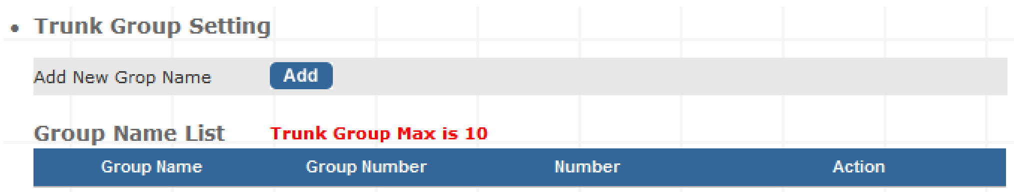


Figure 3-16. Trunk Group setting - 1

Press “more” to show the Service Provider Number under the group.

Figure 3-17. Trunk Group setting - 2

Figure 3-18. Trunk Group more information

➢ Add New Trunk Group

Step 1. Press “**Add**” button to add an new Group Name information.

Figure 3-19. Add an new Group Name

Step 2. Fill in the required information in Trunk Group Setup page.

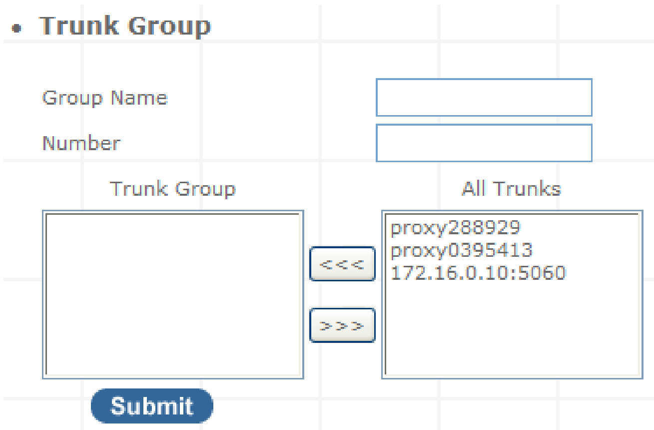


Figure 3-20. Trunk Group Setup

Group Name	The Trunk Group name
Number	If the leading digits are match with this number, IP PBX will delete this number and send out the following digits.
All Trunk	It will show all the available SIP Trunks and Gateway Trunks for selection.
Trunk Group	Choose the trunk at All Trunk box and press the <<< button to move the activated trunk to Trunk Group box.

Table 3-13. Trunk Group setting description

➢ **Scenario Sample**

IP PBX has created two different SIP trunks and one Gateway trunk for outgoing trunks.

Group Name List Trunk Group Max is 10

Group Name	Group Number	Number	Action
SIP_Trunk_1	81	proxy288929	Edit Delete
SIP_Trunk_2	82	proxy0395413	Edit Delete
FXO_Gateway	0	172.16.0.10:5060	Edit Delete

Figure 3-21. Trunk Group sample setting

One-Stage Call:

1. If user dials **81**123456, this call will hunt **SIP_Trunk_1** and send 123456 to call out.
2. If user dials **82**234567, this call will hunt **SIP_Trunk_2** and send 234567 to call out.
3. If user dials **0**345678, this call will hunt **FXO_Gateway** and send 345678 to call out.

Two-Stage Call:

1. If user dials **81** and hear the dial tone, then dial 123456. This call will hunt **SIP_Trunk_1** and send 123456 to call out.
2. If user dials **82** and hear the dial tone, then dial 234567. This call will hunt **SIP_Trunk_2** and send 234567 to call out.
3. If user dials **0** and hear the dial tone, then dial 345678. This call will hunt **FXO_Gateway** and send 345678 to call out.

Dialing Rules

When want to make VoIP calls through the above FXO, SIP or Gateway Trunk, the user can use the "**Dialing Rules**" function to simplify the dialing number.

In the "Dialing Rules" settings: Maximum Entries: **100 records**

• **Dialing Rules**

Max Rule is 100

Phone NO.	Delete Length	Prefix NO.	Trunk/Trunk Group	Action
				Insert Change

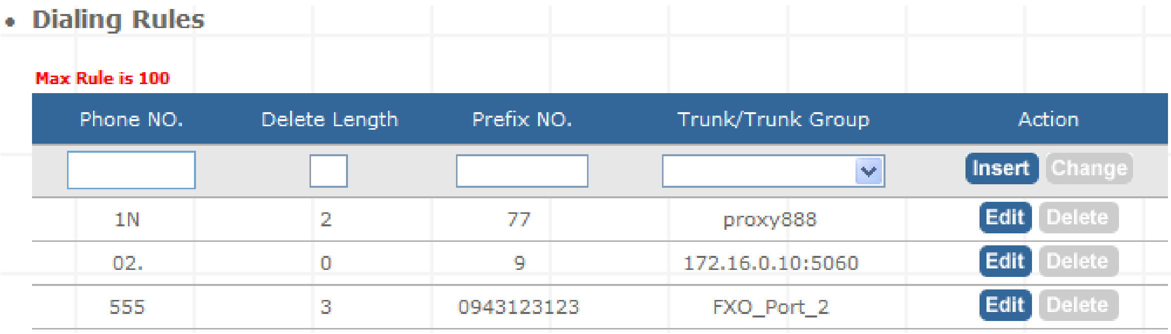
Figure 3-22. Dialing Rules settings

Phone NO	Phone Number. is the leading digit of the call out dialing number. Phone NO Pattern: "**N**" single digit from 2 to 9. "**z**" single digit from 1 to 9. "**X**" single digit from 0 to 9. "**.**" unlimited length of digit.

Delete Length	Delete Length is the number of digits that will be stripped from beginning of the dialed number.
Prefix NO	Prefix NO is the digits that will be added to the beginning of the dialed number.
Trunk/Trunk Group	To choose the FXO, SIP or Gateway Trunk.

Table 3-14. Dialing Rules description

➢ Scenario Sample



Phone NO.	Delete Length	Prefix NO.	Trunk/Trunk Group	Action
				Insert Change
1N	2	77	proxy888	Edit Delete
02.	0	9	172.16.0.10:5060	Edit Delete
555	3	0943123123	FXO_Port_2	Edit Delete

Figure 3-25. Dialing Rules list – 1

1. If user dials **12**, this call will hunt **SIP_Trunk (proxy888)** and send 77 to call out.
2. If user dials **02**345, this call will hunt **Gateway Trunk (172.16.0.10)** and send 902345 to call out.
3. If user dials **555**, this call will hunt **FXO_Port_2** and send 0943123123 to call out.

Attendant Number

Attendant Number in IP PBX system helps you to configure internal dial plan for extension setup. It can allow more calls to be handled by IVR from Gateway's FXO, and FXS port. **Attendant Extension Provide 10 sets of IVR.**

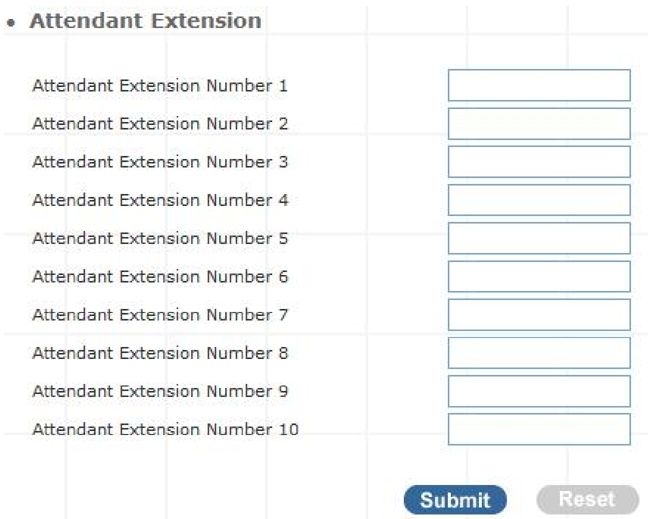


Figure 3-26. Attendant extension settings

The IP PBX will handle incoming *Caller ID* and show to remote / local registered IP-Phone.

Note

```
If your Gateway can bypass Mobile/Analog Phone number, The
IP PBX will handle incoming caller ID and show to remote
/ local registered IP-Phone.
```

- **Sample:**

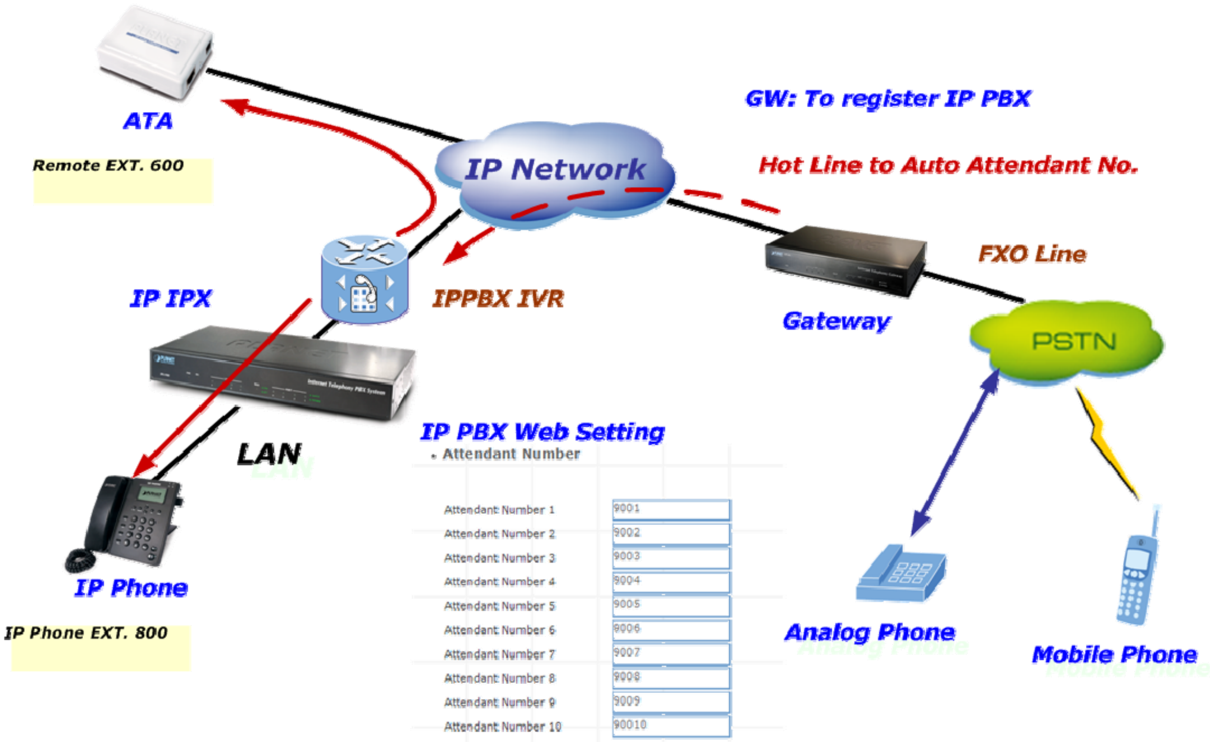


Figure 3-27. Auto-attendant sample

Attendant Message

The Attendant Message on the IP PBX systems, it is can auto-answer attendant message setting on the attendant time, IP PBX message can play voice to SIP Trunk and Gateway's FXO, and FXS port.

IP PBX Setup

- Attendant Message

Message	Service Digit	Action
onduty		Advance
offduty		Advance
custom1		Advance
custom2		Advance
custom3		Advance

Figure 3-28. Auto-attendant message

Attendant Message Advance Setting :

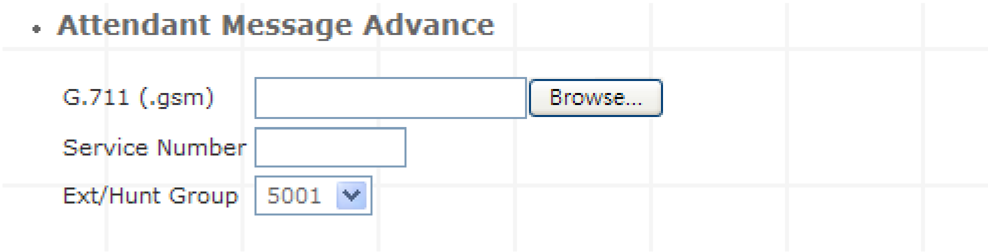


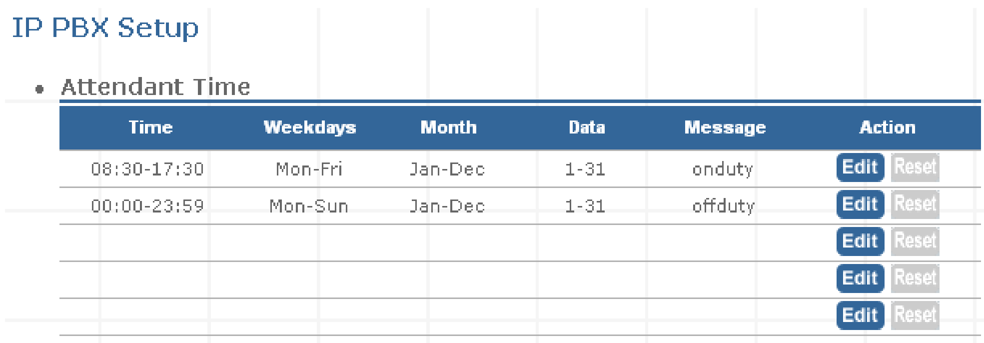
Figure 3-29. Auto-attendant message advance setting

G.11(.gsm)	You can upload gsm format voice file to IP PBX.
Service Number	Associate a dial number with a call group voice instruction to instruct incoming calls
Ext/Hunt Group	Specificity the call group hunting.

Table 3-15. Attendant Messages setup description

Attendant Time

Defined **Attendant Time o**n the IP PBX systems, it is can answer attendant message to match on the attendant time.



Time	Weekdays	Month	Data	Message	Action
08:30-17:30	Mon-Fri	Jan-Dec	1-31	onduty	Edit Reset
00:00-23:59	Mon-Sun	Jan-Dec	1-31	offduty	Edit Reset
					Edit Reset
					Edit Reset
					Edit Reset

Figure 3-30. Auto-attendant time list

Attendant Time Advance Setting :

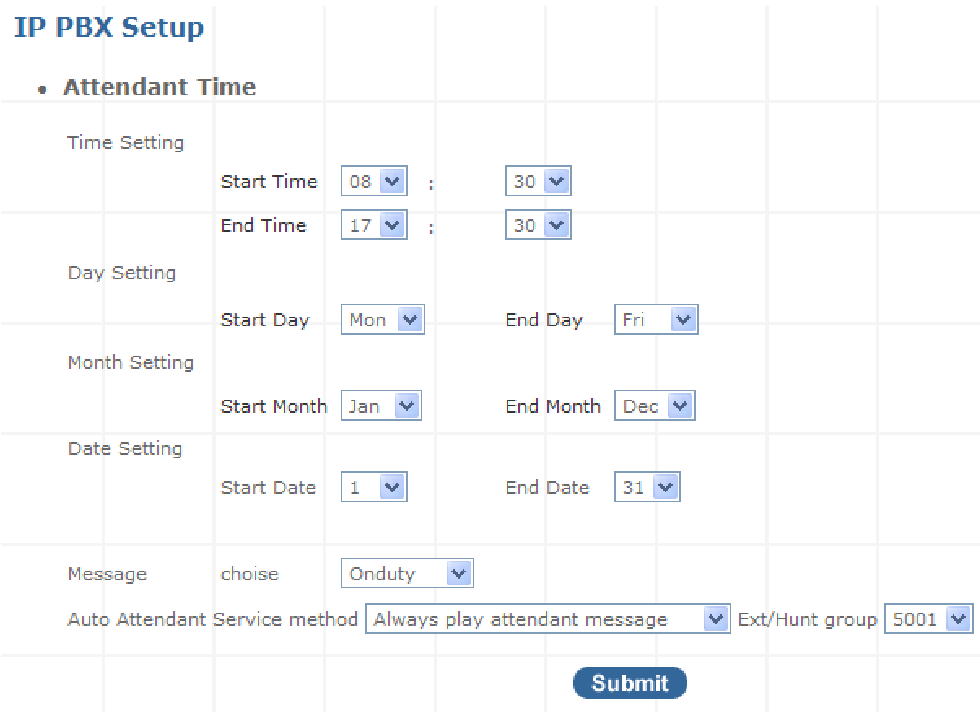


Figure 3-31. Auto-attendant time setting

Day Setting	Defined Start Day / End Day.
Time Setting	Defined Start Time / End Time.
Month Setting:	Defined Start Month / End Month .
Date Setting	Defined Start Date / End Date.
Message	Select play voice message.
Auto Attendant Service Method	Defined the Auto Attendant Service Method. a). Always play attendant messages b). Always goto EXT/HuntGroup c). User try error goto EXT/HuntGroup

Table 3-16. Attendant Time setup description

Record Auto Attendant

Allow you to record On / Off duty voice menu over a register ip-phone.

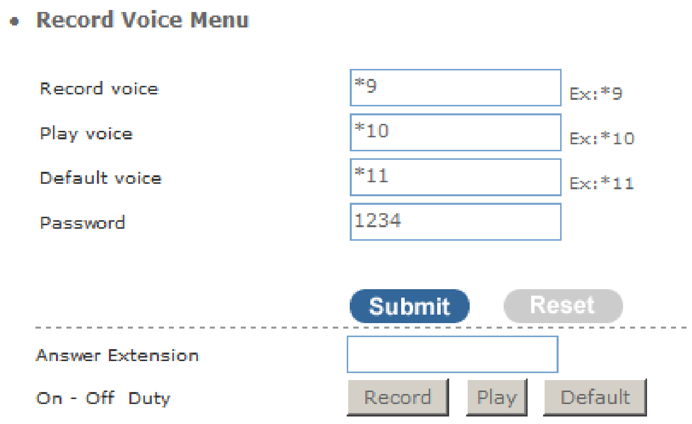


Figure 3-32. Record voice menu settings

Pick up your register IP-Phone handset and press “function key + password “ to enter into voice menu guide.

Record voice	Record your voice menu , Default is *9
Play voice	Play your record voice menu ,Default is *10
Default voice	To set default voice menu, Default is *11
Password	This is record / default voice password , Default is 1234

Table 3-17. Record voice menu description

Answer Extension enable you to record the customized voice menu remotely from a registered IP-Phone.

Answer extension	Call from registered IP-Phone to record the voice menu.

Table 3-18. Answer extension description

Upload Voice File

This page allows transfer music on hold file or PBX Voice Files from your PC to IP PBX. Please refer to the Appendix C for detail descriptions.

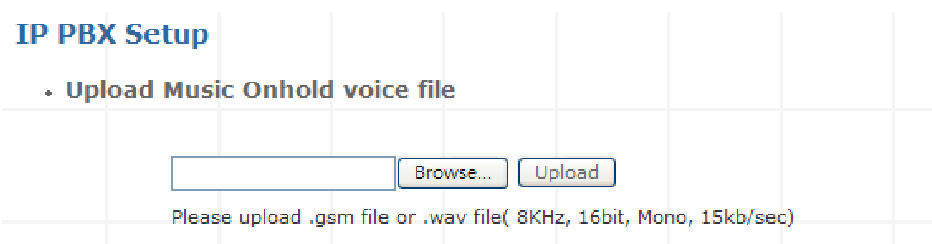


Figure 3-33. On-hold voice uploads

Click ***Browse*** and select your file, then click ***upload*** to finish.

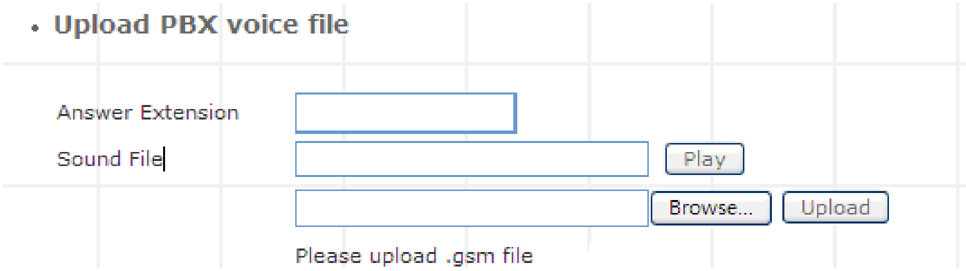


Figure 3-34. Answer extension voice upload

Answer extension	Call from registered IP-Phone to record the voice menu.
Sound File	Select G.711 Voice file, then click ***play*** to your registered device.

Table 3-19. Voice upload setup description

Call Parking

Build a calling rule for IP Phone to park the calls during the phone conversation.

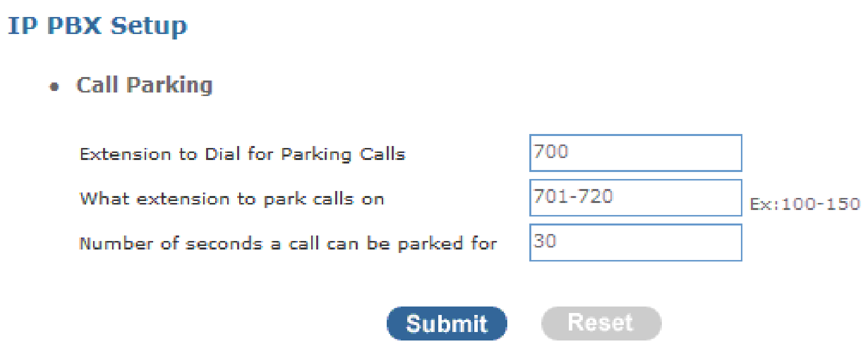


Figure 3-35. Call parking settings

Extension to Dial for Parking Calls	Set an extension number to dial when need to park the call. Default number is *700*.
What extension to park calls on	Set the Extension range for call parking retrieving. (*Example*: '701-720').
Number of seconds a call can be parked for	Set allowed parking time for the parking call. Default is 30/sec.
Pickup Extension	Set up a number for IP Phone to retrieve back the call. Default is *8.
Timeout for answer on attended transfer	*Set a timeout value for answer the transferred call. Default is* 30 Sec.

Table 3-20. Call parking description

Gereral Setting

IP Phone or sip device extension connected IP PBX, extension have call forward / transfer and pickup / voice key …

➢ Call Forward Key

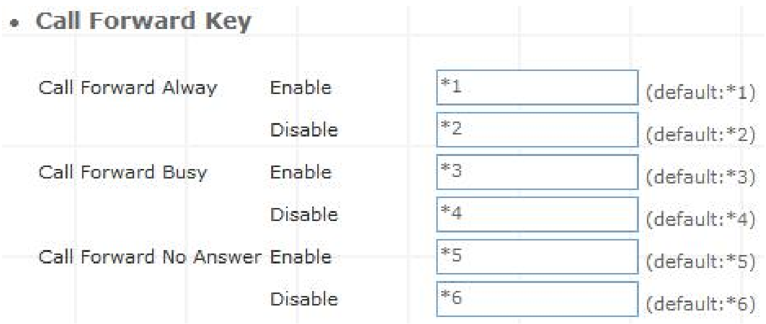


Figure 3-36. Call forward key settings

Call forward always	**Enable:** Dial the " *1 + number " enable call forward always function **Disable:** Dial the " * 2" disable call forward always function
Call forward Busy	**Enable:** Dial the " *3 + number " enable call forward busy function **Disable:** Dial the " * 4 " disable call forward busy function
Call forward no answer	**Enable:** Dial the " *5 + number " enable call forward no answer function **Disable:** Dial the " * 6 " disable call forward no answer function

Table 3-21. Call forward description

➢ Transfer Feature

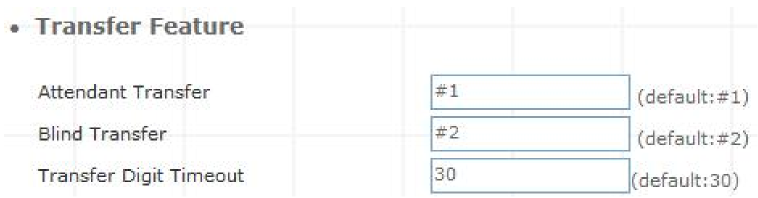


Figure 3-37. Transfer feature settings

Attendant Transfer	When you attendant transfer fail, you can definition other transfer number
Blind Transfer	Blind Transfer , When Ex: Ext 100 call Ext 200, Ext 200 blind transfer to Ext 300 , Ignore the Ext.300 status, the Ext.200 will immediately on-hook
Transfer Digit time out	Set (Attendant/blind) transfer digit time out sec

Table 3-22 Transfer feature description

➢ Pickup Key

Figure 3-38. Pickup key settings

Pickup Extension	Set call pickup (Default is *8)

Table 3-23. Pickip description

➢ Voice Mail

Figure 3-39. Voice mail settings

Max time of a voice mail	Set a voice mail max time
Max number of messages per folder	Max number of voice mail per folder
Dial voice mail number	Dial " *12 " into voice mail guide
Dial my voice mail number	Dial " *13 + Ext number " into voice mail guide

Table 3-24. Voice mail description

➢ SMTP Setting

SMTP is a relatively simple, text-based protocol, where one or more recipients of a message are specified. Input the valid account number, the extension setting voice mail will be been in used.

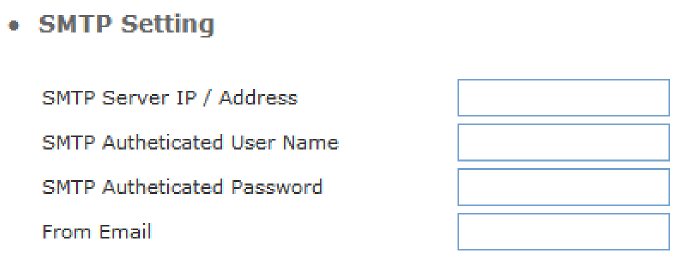


Figure 3-40. SMTP settings

SMTP server IP / Address	Input server IP / Address
SMTP Authentication user name	Input SMTP Authentication user name

SMTP Authentication password	Input SMTP Authentication password
From Email	Input your Email, if server to check your email address.

Table 3-25. SMTP description

Hunt Group Setting

This setting will allow the caller to choose the specific extension group to answer the phone (e.g. Press 9 for Operator). Every incoming call (from Service Provider or Attendant Extension) will first hear the pre-recorded On / Off Duty Voice for call group options for caller to select.

Users can also setup multiple groups to manage the incoming calls.

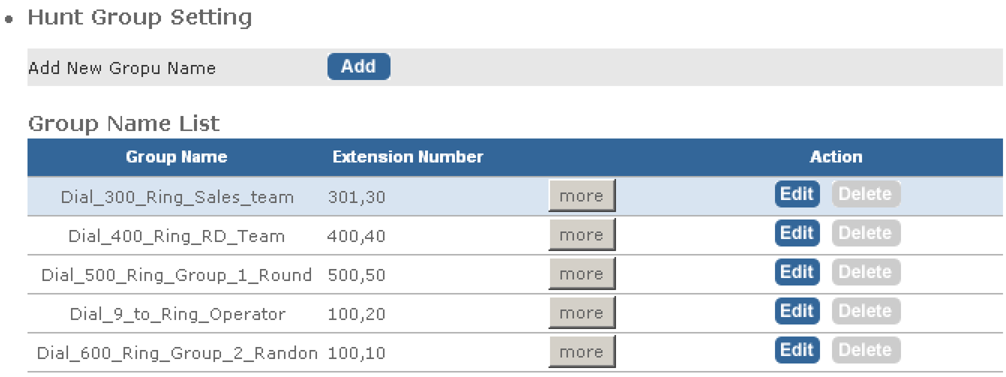


Figure 3-41. Hunt Group settings

Press "**Add**" to add a new Hunt Group;

Press "**Edit**" to the edit a specified hunt group;

Press "**Delete**" to delete a specified hunt group;

Press "**more**" to show the extension number under the group.

Figure 3-42. Hunt Group list

Figure 3-43. Hunt Group more information

Add New Hunt Group

Step 1. Press "**Add**" button to add a new Group Name information.

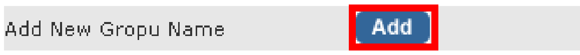


Figure 3-44. Add an new Group Name

Step 2. Fill in the required information in Hunt Group Setup page.

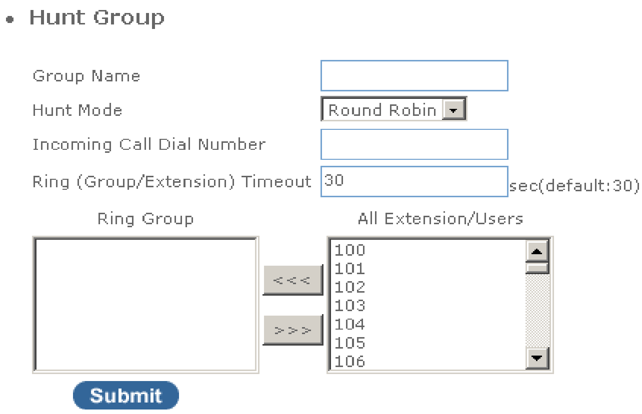


Figure 3-45. Hunt Group setup

Group Name	Input your group name
Hunt Mode	There are 3 modes available: **Round Robin** / **Ring All** / **Random** Mode. 1. Round Robin: Take turns ringing each available Extension / Users 2. Ring All: Ring all Extension/Users, until any one Extension / Users answer the call. 3. Random: Ring random group inside Extension / Users
Incoming Call Dial Number	Associate a dial number with a call group voice instruction to instruct incoming calls (e.g. If "20" is associated with Group A, when the caller dial "20", all extensions under Group A will ring). Default incoming call dial number is *empty*.
Ring (Group/Extension) Timeout	Setup a timeframe to control the call group hunting timeout. Default setting is *30 sec*.

Table 3-26. Hunt Group description

To add extension/users to Ring group

Step 1. Select your extension

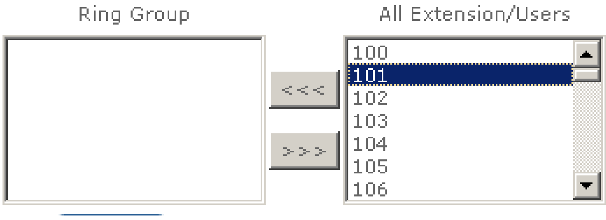


Figure 3-46. Add Extension/User

Step 2. Press  to add extension/users to ring group.

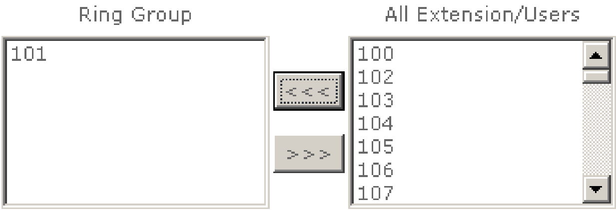


Figure 3-47. Add Extension/User

➢ To delete Ring Group inside extension/users

Step 1. Select the extensions

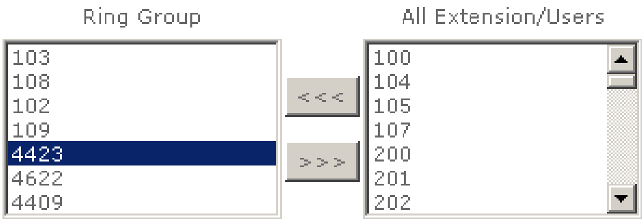
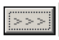


Figure 3-48. Delete Extension/User

Step 2. Press  to delete extension/users to ring group.

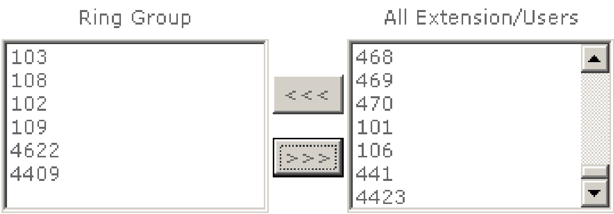


Figure 3-49. Delete Extension/User

Call Screen

Call Screen allows you to block outgoing (for SIP trunk / gateway trunk) calls from SIP extension user number.

Figure 3-50. Call Screen settings

➢ Add New call group

Step 1. In IP PBX Setup → Call Screen setting → Press "Add" button to add a new Call Screen Group information.

Figure 3-51. To add new group name

Step 2. Fill in the required information in call screen group Setup page

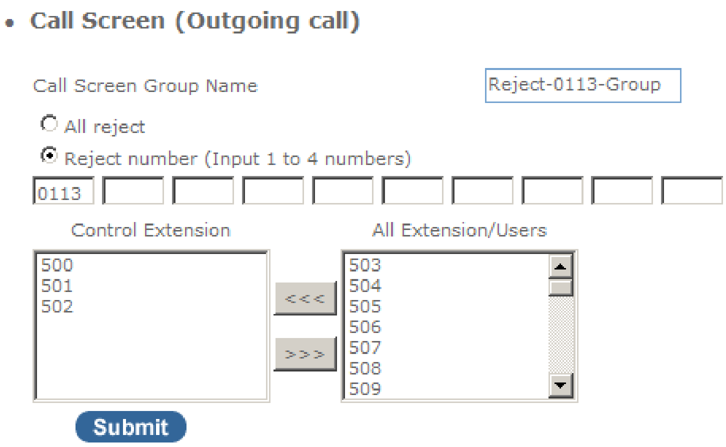


Figure 3-52. Call Screen settings

This sample reject prefix number is 0113 for sip extension 500,501,502 group.

Call screen group name	Input your call screen group name.
All Reject	This option is reject all outgoing call.
Reject Number (Input 1 to 4 number)	Input 1 to 4 reject prefix number.

Table 3-27. Call Screen description

Application

A. Group 2 must be open only to dial local calls and this group wants to use some passwords or keys to dial Long distance.

May I say it also means group only allow to dial local calls, so we reject all long distance call.

For example,

For dialing long distance calls, the number start with 01, 02, 00.

Now extension 200, not able to dial those numbers.

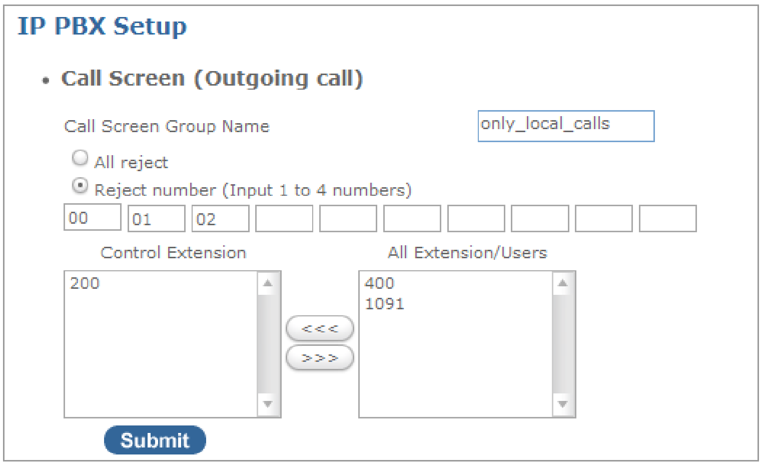


Figure 3-53. Call Screen settings-Application 1

B. This group wants to use some passwords or keys to dial Long distance.

I think this can be easily be done by using long group number.

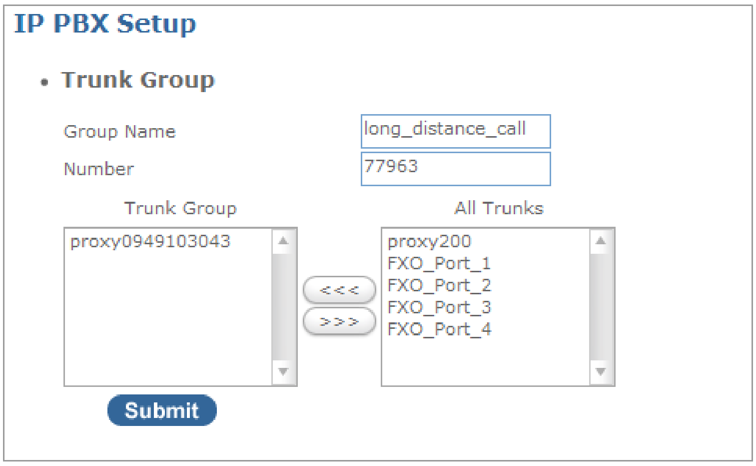


Figure 3-54. Call Screen settings-Application 2

The user now needs to dial 77963(as password) to make call by this trunk

C. Group 3 must be close to all traffic, open only to dial extension numbers.

Select All Reject

Choose the extensions that you don't want it to make calls.

For instance 200

Extension 200 is not able to call out, ONLY able to dial extension numbers.

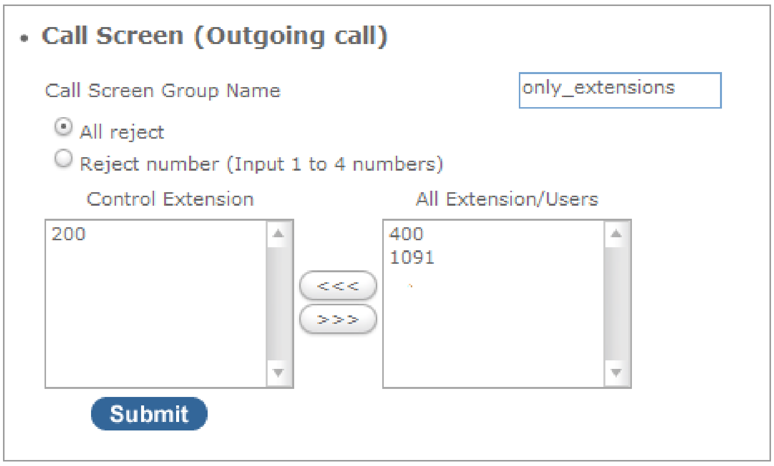


Figure 3-55. Call Screen settings-Application 3

Meet Room Setting

IP PBX provides Meet me conference rooms, support to 15 conference rooms for admin or users PIN access.

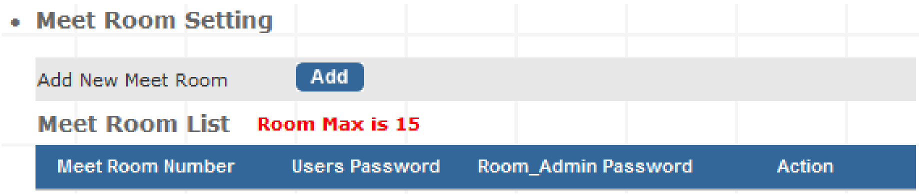


Figure 3-56. Meet Room settings

Press “Add” button to add a new Meet Room.

.

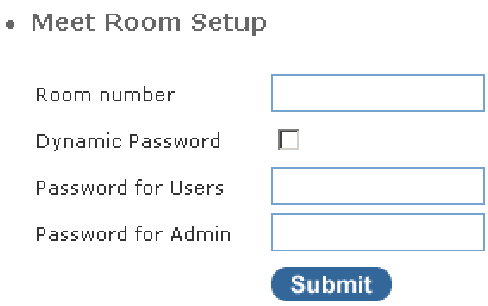


Figure 3-57. Meet Room settings

Room Number	Input admin / user join Room Number.
Dynamic password	If dynamic password is select, prompting for a PIN.
Password for Users	Input password for users.
Password for Admin	Input password for admin.

Table 3-28. Meet Room description

Chapter 4
Network Setup

WAN & LAN Setup

WAN (Wide Area Network) is a network connection connecting one or more LANs together over some distance. For example, the means of connecting two office buildings separated by several kilometers would be referred to as a WAN connection. The size of a WAN and the number of distinct LANs connected to a WAN is not limited by any definition. Therefore, the Internet may be called a WAN.

WAN Settings are settings that are used to connect to your ISP (Internet Service Provider). The WAN settings are provided to you by your ISP and often times referred to as "public settings". Please select the appropriate option for your specific ISP.

For most users, Internet access is the primary application. IP PBX supports the WAN interface for internet access and remote access. The following sections will explain more details of WAN Port Internet access and broadband access setup. When you click “**WAN & LAN Setup**”, the following setup page will be shown. Three methods are available for Internet Access.

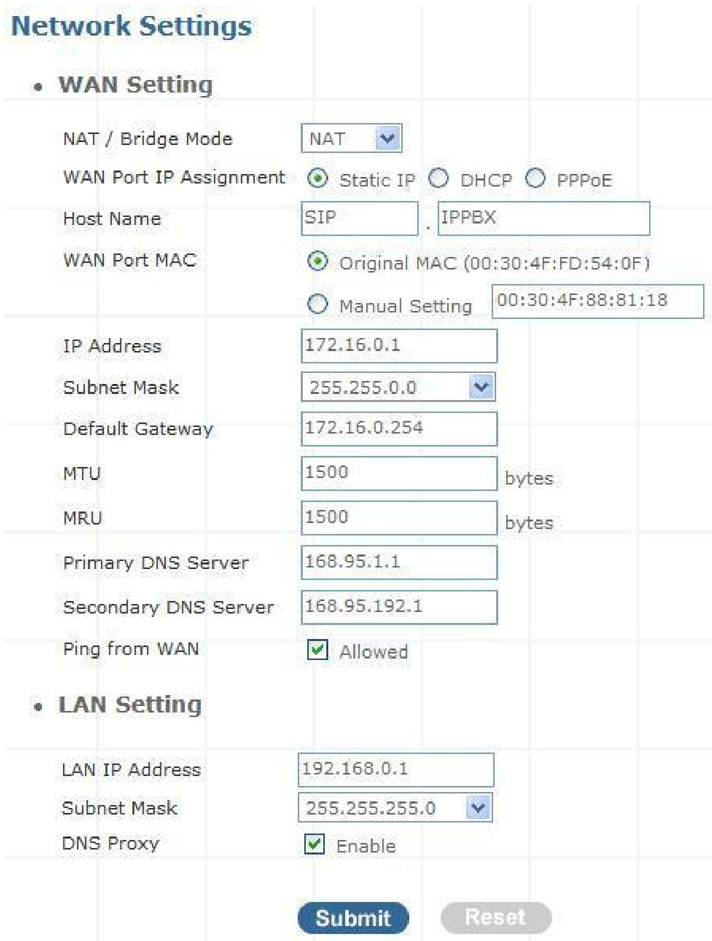


Figure 4-1. Network settings

➢ Static IP

If you are a leased line user with a fixed IP address, enter in the IP address, subnet mask, gateway address, and DNS (domain name server) address(es) provided to you by your ISP. Each IP address entered in the fields must be in the appropriate IP form, which are four IP octets separated by a dot (x.x.x.x). The Router will not accept the IP address if it is not in this format. *Example: 168.95.1.2*

Figure 4-2. WAN-Static IP settings

IP Address	Check with your ISP provider.
Subnet Mask	Check with your ISP provider.
Default Gateway	Check with your ISP provider.

Table 4-1. WAN-Static IP description

➢ DHCP

Dynamic Host Configuration Protocol (DHCP), Dynamic IP (Get WAN IP Address automatically). If you are connected to the Internet through a Cable modem line, then a dynamic IP will be assigned.

Note

WAN port gets the IP Address, Subnet Mask and default gateway IP address automatically, if DHCP client is successful.

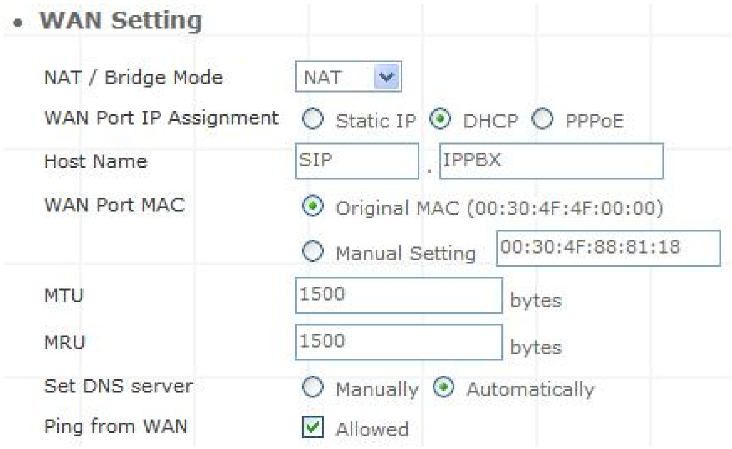


Figure 4-3. WAN-DHCP settings

➢ PPPoE

Point-to-Point Protocol over Ethernet (PPPoE). Some ISPs provide DSL-based services and use PPPoE to establish communication link with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to make sure the following items, PPPoE User name: Enter username provided by your ISP. PPPoE Password: Enter password provided by your ISP.

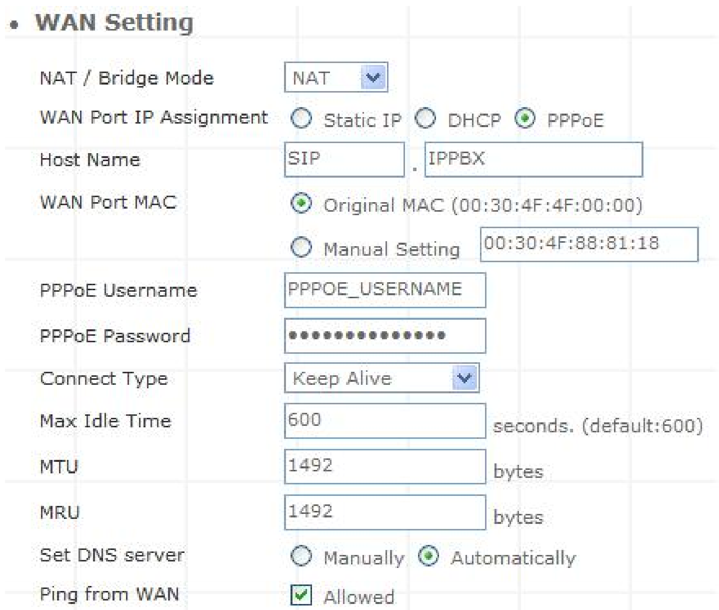


Figure 4-4. WAN-PPPoE settings

➢ Host Name

The Host Name field is optional but may be required by some Internet Service Providers. The default host name is the model number of the device. It is a computer that is connected to a TCP/IP network, including the Internet. Each host has a unique IP address. Assign the domain name or IP address of your host computer. When the host operating system is set up it is given a name. This name may reflect the prime use of the computer. For example, a host computer that converts host names to IP addresses using DNS may be called cvs.IP-PBX.com and a host computer that is a web server may be called www.IP-PBX.com. When we need to find the host name from an IP address we send a request to the host using its IP address. The host will respond with its host name.

➢ WAN Port MAC

The MAC (Media Access Control) Address field is required by some Internet Service Providers (ISP). The default MAC address is set to the MAC address of the WAN interface in the device. It is only necessary to fill the field if required by your ISP.

The WAN port allows your voice gateway to be connected to an Internet Access Device, e.g. router, cable modem, ADSL modem, through a CAT.5 twisted pair Ethernet Cable. MAC addresses are uniquely set by the network adapter manufacturer and are sometimes called "physical addresses" for this reason. MAC assigns a unique number to each IP network adapter called the MAC address. The MAC address is commonly written as a sequence of 12 hexadecimal digits as follows: **00:3f:4f:88:81:18**. The first six hexadecimal digits of the address correspond to a manufacturer's unique identifier, while the last six digits correspond to the device's serial number.

Some Internet service providers track the MAC address of a home router for security purposes. Many routers support a process called cloning that allows the MAC address to be simulated so that it matches one the service provider is expecting. This allows end-user to change their router (and their real MAC address) without having to notify the provider. For example, you could allow packets which have your name server's IP on them, but come from another MAC address (one way of spoofing packets).

Figure 4-5. WAN port MAC settings

➢ MTU and MRU

MTU stands for Maximum Transmission Unit, the largest physical packet size, measured in bytes that a network can transmit. Any messages larger than the MTU are divided into smaller packets before being sent.

MRU stands for Maximum Receiving Unit. The largest physical packet size, measured in bytes that a network can receive. Any messages larger than the MRU are divided into smaller packets before being received.

The key is to be deciding how big your bandwidth pipe is and select the best MTU for your configuration. For example, you have a 33.6 modem, you use a MTU and MRU of 576, and if you have a larger pipe you may want to try 1500.

Figure 4-6. MTU and MRU settings

Note

For Static IP, both MTU and MRU are set to 1500 bytes as default value.
For DHCP, both MTU and MRU are set to 1500 bytes as default value.
For PPPoE, both MTU and MRU are set to 1492 bytes as default value.

➢ DNS Server

DNS stands for Domain Name System. Every Internet host must have a unique IP address; also they may have a user-friendly, easy to remember name such as www.ippbx.com. The DNS server converts the user-friendly name into its equivalent IP address. The original DNS specifications require that each domain name is served by at least 2 DNS servers for redundancy. When you run your DNS, web, and mail servers all on the same MAChine - if this MAChine goes down, it doesn't really matter that the backup DNS server still works.

The recommended practice is to configure the primary and secondary DNS servers on separate MAChines, on separate Internet connections, and in separate geographic locations.

Figure 4-7. DNS server settings

Primary DNS Server	Sets the IP address of the primary DNS server.
Secondary DNS Server	Sets the IP address of the secondary DNS server.

Table 4-2. DNS server description

➢ Ping From WAN

Ping is a basic Internet program that lets you verify that a particular IP address exists and can accept requests. Ping is used diagnostically to ensure that a host computer you are trying to reach is actually operating.

The default setting is allowed user can ping the host computer from remote site. If you disallow, the host computer doesn't response any user who issues Ping IP address command from any remote sites.

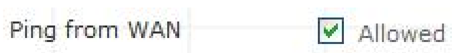


Figure 4-8. Ping from wan settings

➢ LAN Setting

These are the IP settings of the LAN (Local Area Network) interface for the device. These settings may be referred to as "private settings". You may change the LAN IP address if needed. The LAN IP address is private to your internal network and cannot be seen on the Internet. The default IP address is 192.168.0.1 with a subnet mask of 255.255.255.0.

LAN is a network of computers or other devices that are in relatively close range of each other. For example, devices in a home or office building would be considered part of a local area network.

Figure 4-9. LAN settings

LAN IP Address	Assign the IP address of LAN server, default is 192.168.0.1
Subnet Mask	Select a subnet mask from the pull-down menu, default is 255.255.255.0

Table 4-3. LAN description

➢ DNS Proxy

A proxy server is a computer network service that allows clients to make indirect network connections to other network services. The default setting is Enable the DNS proxy server.

Figure 4-10. DNS proxy settings

DHCP

DHCP stands for Dynamic Host Control Protocol. The DHCP server gives out IP addresses when a device is starting up and request an IP address to be logged on to the network. The device must be set as a DHCP client to "Obtain the IP address automatically". By default, the DHCP Server is enabled in the unit. The DHCP address pool contains the range of the IP address that will automatically be assigned to the clients on the network.

DHCP client computers connected to the unit will have their information displayed in the DHCP Client List table. The table will show the Type, Host Name, IP Address, MAC Address, Description, and

Expired Time of the DHCP lease for each client computer. DHCP Server is a useful tool that automates the assignment of IP addresses to numbers of computers in your network. The server maintains a pool of IP addresses that you use to create scopes. (A DHCP scope is a collection of IP addresses and TCP/IP configuration parameters that are available for DHCP clients to lease.) Then, the server automatically allocates these IP addresses and related TCP/IP configuration settings to DHCP-enabled clients in the network. The DHCP Server leases the IP addresses to clients for a period that you specify when you create a scope. A lease becomes inactive when it expires. Through the DHCP Server, you can reserve specific IP addresses permanently for hardware devices that must have a static IP address (e.g., a DNS Server).

An advantage of using DHCP is that the service assigns addresses dynamically. The DHCP Server returns addresses that are no longer in use to the IP addresses pool so that the server can reallocate them to other machines in the network. If you disable this DHCP, you would have to manually configure IP for new computers, keep track of IP addresses so that you could reassign addresses that clients aren't using, and reconfigure computers that you move from one subnet to another. The DHCP Static MAP table lists all MAC and IP address which are active now.

Figure 4-11. DHCP server settings

When you enable the DHCP server, you are able to enter:

Assigned DHCP IP Address	Enter the starting IP address for the DHCP server's IP assignment and the ending IP address for the DHCP server's IP assignment.
DHCP IP Lease Time	Assign the length of time for the IP lease, default setting is 86400 seconds.

Table 4-4. DHCP server description

Static Route

Static routes are special routes that the network administrator manually enters into the router configuration for local network management. You could build an entire network based on static routes. The problem with doing this is that when a network failure occurs, the static route will not change without you performing the change. This could be IP-PBX if the failure occurs when the administrator is not available.

The route table allows the user to configure and define all the static routes supported by the router.

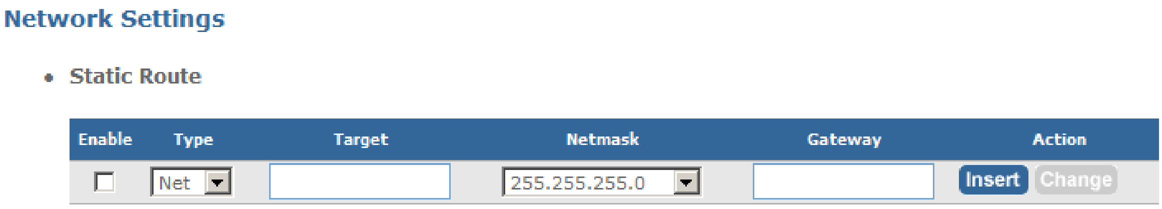


Figure 4-12. Static route settings

Enable	Enable/Disable the static route.
Type	Indicates the type of route as follows, Host for local connection and Net for network connection.
Target	Defines the base IP address (Network Number) that will be compared with the destination IP address (after an AND with NetMask) to see if this is the target route.
NetMask	The subnet mask that will be AND'd with the destination IP address and then compared with the Target to see if this is the target route.
Gateway	The IP address of the next hop router that will be used to route traffic for this route. If this route is local (defines the locally connected hosts and Type = Host) then this IP address MUST be the IP address of the router.
Action	Insert a new Static Router entry or update a specified entry.

Table 4-5. Static route description

NAT

NAT (Network Address Translation) serves three purposes:

1. Provides security by hiding internal IP addresses. Acts like firewall.
2. Enables a company to access internal IP addresses. Internal IP addresses that are only available within the company will not conflict with public IP.
3. Allows a company to combine multiple ISDN connections into a single internet connection.

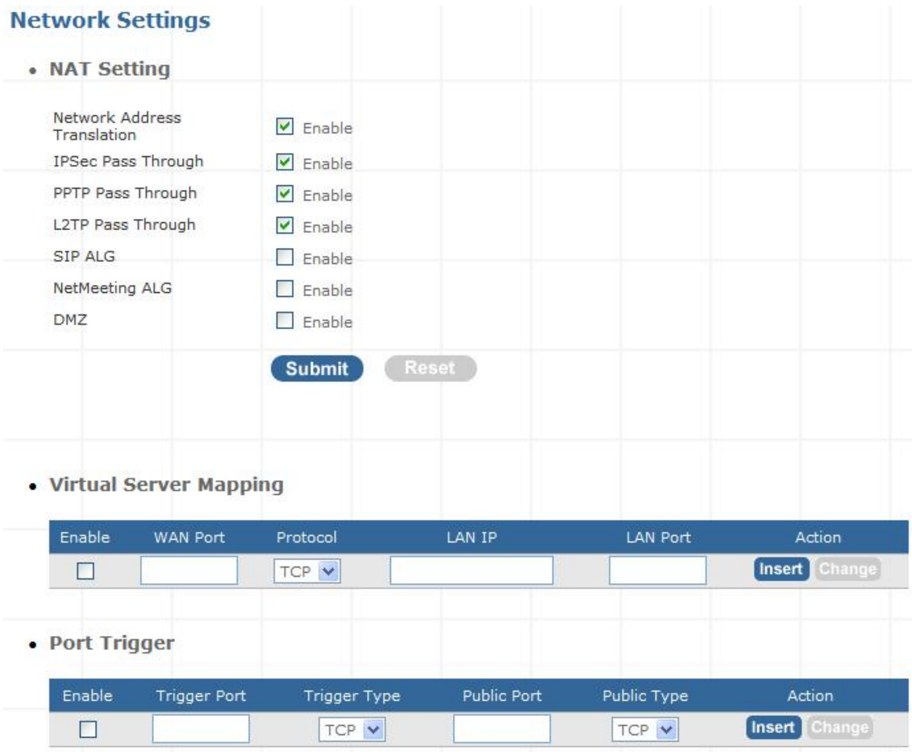


Figure 4-13. NAT settings

➢ NAT Setting

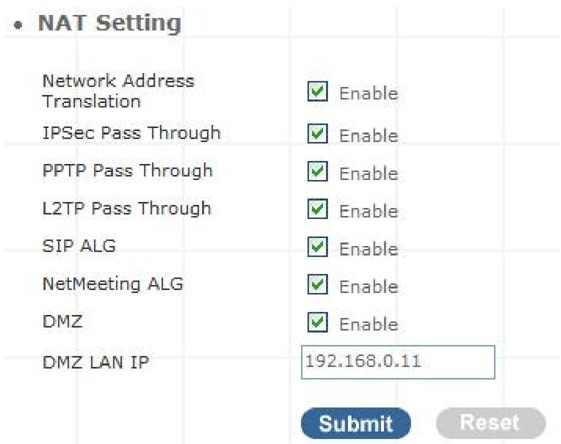


Figure 4-14. NAT settings

Network Address Translation	Enable/Disable NAT.
IPSec Pass Through	IPsec (Internet Protocol Security) is a framework for a set of protocols for security at the network or packet processing layer of network communication. Enable/Disable this framework verification.
PPTP Pass Through	PPTP (Point-to-Point Tunneling Protocol) is a protocol that allows corporations to extend their own corporate network through private "tunnels" over the public Internet. Enable/Disable this protocol verification.

L2TP Pass Through	L2TP (The Layer 2 Tunnel Protocol) is an emerging Internet Engineering Task Force (IETF) standard that combines the best features of two existing tunneling protocols: Cisco's Layer 2 Forwarding (L2F) and Microsoft's Point-to-Point Tunneling Protocol (PPTP). L2TP is an extension to the Point-to-Point Protocol (PPP), which is an important component for VPNs. VPNs allow users and telecommuters to connect to their corporate intranets or extranets. Enable/Disable this function.
SIP ALG	SIP, the Session Initiation Protocol, is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. Enable/Disable this protocol verification.
DMZ	In computer networks, a DMZ (Demilitarized Zone) is a computer host or small network inserted as a "neutral zone" between a company's private network and the outside public network. It prevents outside users from getting direct access to a server that has company dIP-PBX. Think of DMZ as the front yard of your house. It belongs to you and you may put some things there, but you would put anything valuable inside the house where it can be properly secured. Setting up a DMZ is very easy. If you have multiple computer s, you can choose to simply place one of the computers between the Internet connection and the firewall.
DMZ IP LAN	If you have a computer that cannot run Internet applications properly from behind the device, then you can allow the computer to have unrestricted Internet access. Enter the IP address of that computer as a DMZ host with unrestricted Internet access. Adding a client to the DMZ may expose that computer to a variety of security risks; so only use this option as a last resort.

Table 4-6. NAT description

- **Virtual Server Mapping**

The device can be configured as a virtual server so that remote users accessing services such as Web or FTP services via the public (WAN) IP address can be automatically redirected to local servers in the LAN network. Depending on the requested service (TCP/UDP port number), the device redirects the external service request to the appropriate server within the LAN network. You will only need to input the LAN IP address of the computer running the service and enable it.

A Virtual Server is defined as a service port, and all requests to this port will be redirected to the computer specified by the server IP.

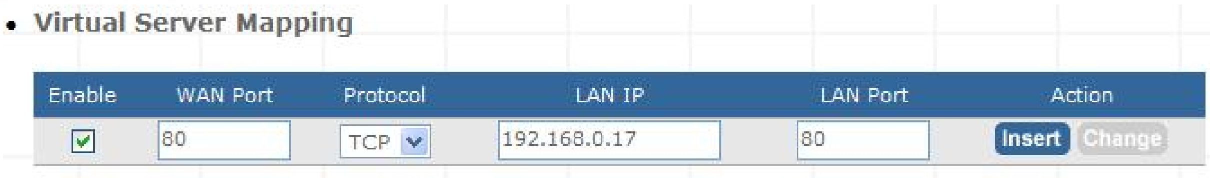


Figure 4-15. Virtual server mapping settings

Enable	Enable/Disable the virtual server mapping, default setting is Disable.
WAN Port	The port number on the WAN side that will be used to access the virtual service. Enter the WAN Port number, e.g. enter 80 to represent the Web (http server), or enter 25 to represent SMTP (email server). Note: You can *specify maximum 32 WAN Ports.*
Protocol	The protocol used for the virtual service. Select a protocol type is TCP or UDP.
LAN IP	The server computer in the LAN network that will be providing the virtual services. Enter the IP address of LAN.
LAN Port	The port number of the service used by the Private IP computer. Enter the LAN port number.
Action	Insert a new WAN port or update a specified WAN port.

Table 4-7. Virtual server mapping description

- **Port Trigger**

Some applications require multiple connections, such as Internet gaming, video conferencing, Internet telephony and others. These applications have difficulties working through NAT (Network Address Translation). If you need to run applications that require multiple connections, specify the port normally associated with an application in the "Trigger Port" field, select the protocol type as TCP (Transmission Control Protocol) or UDP (User DIP-PBXgram Protocol), then enter the public ports associated with the trigger port to open them for inbound traffic.

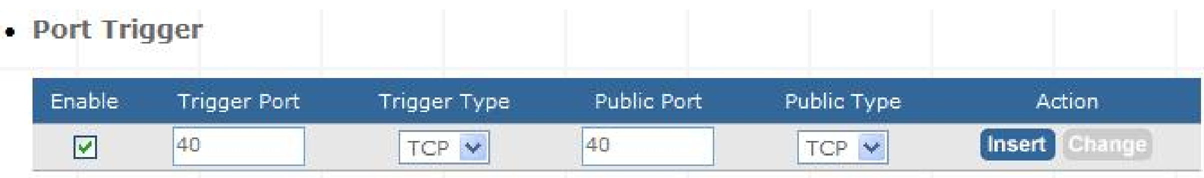


Figure 4-16. Port trigger settings

Enable	Enable/Disable the port trigger, default setting is Disable.
Trigger Port	This is the port used to trigger the application. It can be either a single port or a range of ports.

Trigger Type	This is the protocol used to trigger the special application.
Public Port	This is the port number on the WAN side that will be used to access the application. You may define a single port or a range of ports. You can use a comma to add multiple ports or port ranges.
Public Type	This is the protocol used for the special application.
Action	Insert a new Port Trigger or update a specified Port Trigger.

Table 4-8. Port trigger description

Packet Filter

Controlling access to a network by analyzing the incoming packets and letting they pass or halting them based on the IP addresses of the source. (This function can be useful for residential screening as well – for parental screening or other)

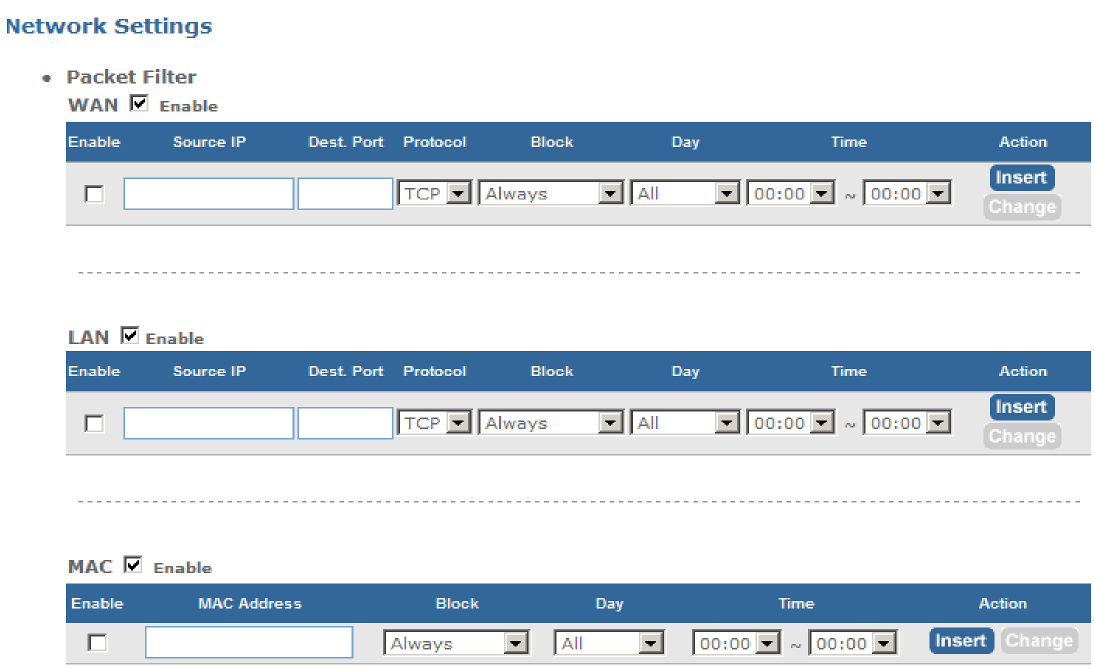


Figure 4-17. Packet filter settings

➢ **WAN**

WAN Enable/Disable	The WAN IP port packet filter function, control a network IP port, default setting is *Enable*.
Enable	Enable/Disable the Internet to WAN IP source port rules, default setting is *Disable*.
Source IP	This is the filter WAN IP address. *Example: 209.131.36.158*
Dest. Port	This is the port used for source IP service.
Protocol	This Protocol Used for the source IP service. Select either TCP or

	UDP.
Block	Wan IP Port Block time setting. Select *Always* or *By Schedule.*
Day	Block Day setting, select a All / Mon-Sat./ Mon-Fri./Mon./ Tues./ Wed./Thu./Fri./Sat./Sun.
Time	Block Time setting, select time range is 00:00 to 23:59.

Table 4-9. Packet filter-WAN description

- **LAN**

LAN Enable/Disable	Internet to LAN filter function, default setting is *Enable*. A prohibitive rule set should only allow the necessary Internet/DMZ services to LAN (Local Area Network) clients.
Enable	Enable/Disable the WAN IP source port rules, default setting is *Disable*.
Source IP	This is the filter source IP address to LAN.
Dest. Port	This is the port used for source IP.
Protocol	This Protocol Used for the WAN Filter service. Select either TCP or UDP.
Day	Block Day setting, select All / Mon-Sat./ Mon-Fri./Mon./ Tues./ Wed./Thu./Fri./Sat./Sun.
Time	Block Time setting, select time range is 00:00 to 23:59

Table 4-10. Packet filter-LAN description

- **MAC**

MAC Enable/Disable	Form internet MAC filter function, default setting is *Enable.*
Block	Wan IP Port Block time Setting. Select Always or *By Schedule*.
Day	Block Day setting, select a All / Mon-Sat./ Mon-Fri./Mon./ Tues./ Wed./Thu./Fri./Sat./Sun.
Time	Block Time setting, select time range is 00:00 to 23:59

Table 4-11. Packet filter-MAC description

URL Filter

URL filter allows you to block sites based on a black list and white list. Sites matching the black list but not matching the white list will be automatically blocked and closed.

Figure 4-18. URL filter settings

Enable	Enable/Disable the URL filter function, default setting is Disable.
Enable	Enable/Disable Block URL to the Clinet IP, default setting is Disable
Client IP	This is the Clinet IP is LAN address. *Example:* 192.168.0.100
URL Filter String	This is the filter URL. *Example*: “http://www.yahoo.com/”

Table 4-12. URL filter description

Security

Intrusion Detection has powerful management and analysis tools that let your IT administrator see what's going on in your network. Such as whose surfing the Web, and gives you the tools to block access to inappropriate Web sites.

Malicious code (also called vandals) is a new breed of Internet threat that cannot be efficiently controlled by conventional antivirus software alone. In contrast to viruses that require a user to execute a program in order to cause damage, vandals are auto-executable applications

Figure 4-19. Security settings

Intrusion Detection	Enable / Disable , network / internet security protection.
Drop Malicious Packet	Enable / Disable , Detect and drop malicious application layer traffic.

Table 4-13. Security description

UPnP

UPnP provides support for communication between control points and devices. The network media, the TCP/IP protocol suite and HTTP provide basic network connectivity and addressing needed. On top of these open, standard, Internet based protocols, UPnP defines a set of HTTP servers to handle discovery, description, control, events, and presentation.

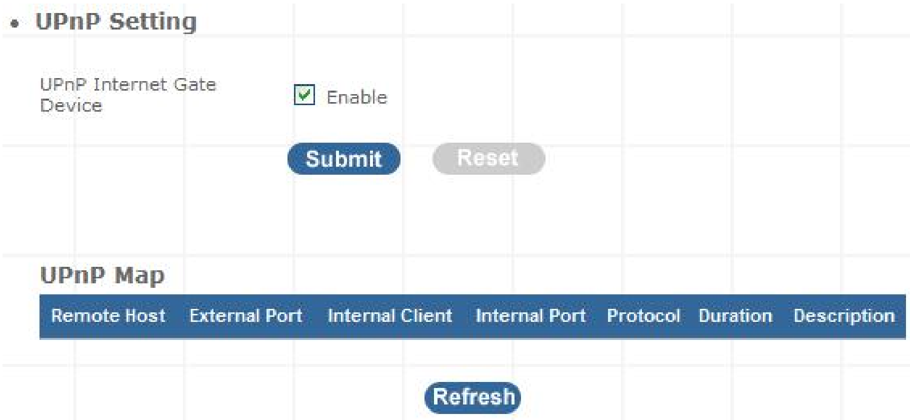


Figure 4-20. UPnP settings

UPNP Internet Gate Device	Enable/Disable UPNP Service to working, default setting is *Disable.*

Table 4-18. UPnP description

DDNS

The DDNS (Dynamic DNS) service allows you to alias a dynamic IP address to a static hostname, allowing your computer to be more easily accessed from various locations on the Internet. Without DDNS, the users should use the WAN IP to reach internal server. It is inconvenient for the users if this IP is dynamic. With DDNS supported, you apply a DNS name (e.g., www.IPPBX.com) for your server (e.g., Web server) from a DDNS server. The outside users can always access the web server using the www.IP-PBX.com regardless of the WAN IP.

When you want your internal server to be accessed by using DNS name rather than using the dynamic IP address, you can use the DDNS service. The DDNS server allows to alias a dynamic IP address to a static hostname.

Unlike DNS that only works with static IP addresses, DDNS works with dynamic IP addresses, such as those assigned by an ISP or other DHCP server. DDNS is popular with home networkers, who typically receive dynamic, frequently-changing IP addresses from their service provider.

DDNS is a method of keeping a domain name linked to a changing (dynamic) IP address. With most Cable and DSL connections, you are assigned a dynamic IP address and that address is used only for the duration of that specific connection. With the IP-PBX, you can setup your DDNS service and the

IP-PBX will automatically update your DDNS server every time it receives a different IP address.

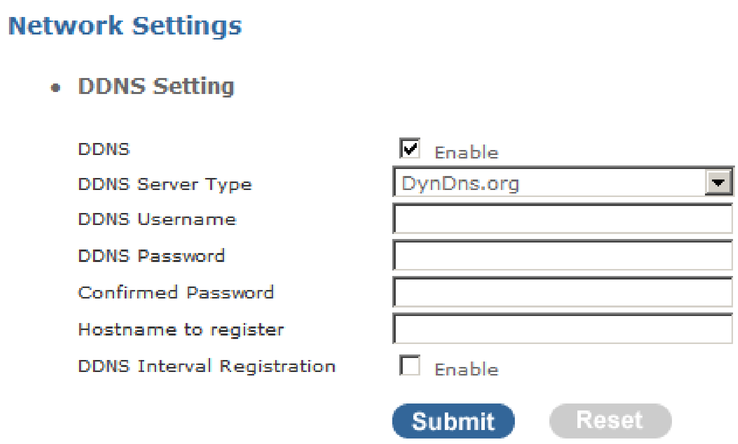


Figure 4-21. DDNS settings

Enable	Enable/Disable the DDNS service, default setting is Disable.
DDNS Server Type	The IP-PBX support two types of DDNS, DynDns.org or No-IP.com
DDNS Username	The username which you register in DynDns.org or No-IP.com website.
DDNS Password	The password which you register in DynDns.org or No-IP.com website.
Confirmed Password	Confirm the password which you typing.
Hostname to register	The hostname which you register in DynDns.org or No-IP.com website

Table 4-14. DDNS description

SNMP

The simple network management protocol (SNMP) forms part of the internet protocol suite as defined by the Internet Engineering Task Force (IETF). SNMP is used by network management systems to monitor network-attached devices for conditions that warrant administrative attention. It consists of a set of standards for network management, including an Application Layer protocol, a IP-PBXbase schema, and a set of IP-PBX objects.

Figure 4-22. SNMP settings

Enable	Enable/Disable the SNMP service, default setting is Disable. (Support SNMP version 1 or SNMP version 2c).
SNMP Read Community	SNMP Read Community string so that EPICenter can retrieve information.(default :public)
SNMP Write Community	Specifies the name of the SNMP write community to which the printer device that this actual destination represents belongs.(Default:private)
SNMP Trap Host	Defines an SNMP trap host to which AppCelera will send trap messages. (Default address is empty)
SNMP Trap Community	The SNMP trap community name. The community name functions as a password for sending trap notifications to the target SNMP manager. (Default: public).

Table 4-15. SNMP description

Chapter 5
Management

Admin Account

The administrator account can access the management interface through the web browser.

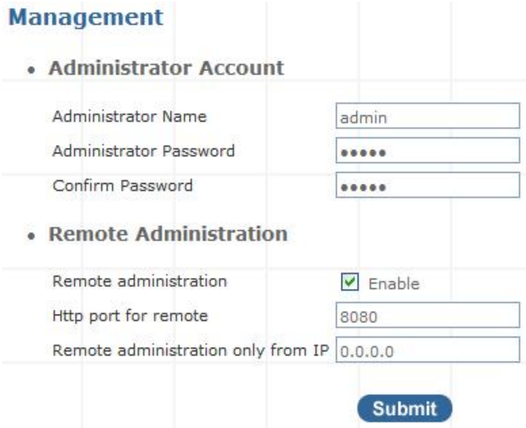


Figure 5-1. Management settings

Administrator Name	Assign a name to represent the administrator account. Maximum 16 characters. Legal characters can be the upper letter “A” to “Z”, lower letter “a” to “z”, digit number “0” to “9” and an underscore sign¡ ”_”.
Administrator Password	Assign an administrator password. Maximum 16 characters and minimum 6 characters with mix of digits and letters characters. Legal characters can be the upper letter “A” to “Z”, lower letter “a” to “z”, digit number “0” to “9” and an underscore sign”_”.
Confirm Password	Enter the administrator password again. Remote Administrator allows the device to be configured through the WAN port from the Internet using a web browser. A username and password is still required to access the browser-based management interface.
Remote Administration	Enable/Disable to access from remote site. Default setting is *“Disable”.*
Http port for remote	If you allowed the access from the remote site, assign the http port used to access the IP-PBX. Default port number is *“8080”.*
Remote administration only from IP	Internet IP address of the computer that has access to the IP-PBX. Assign the legal IP address. *Example:* http://x.x.x.x:8080 where as x.x.x.x is the WAN IP address and 8080 is the port used for the Web-Management interface.

Table 5-1. Management description

Note

- The administrator name and password are *case-sensitive* and the "blank" character is an *illegal character*
- Only the administrator account has the ability to change account password.

Date & Time

➢ Manual Time Setting

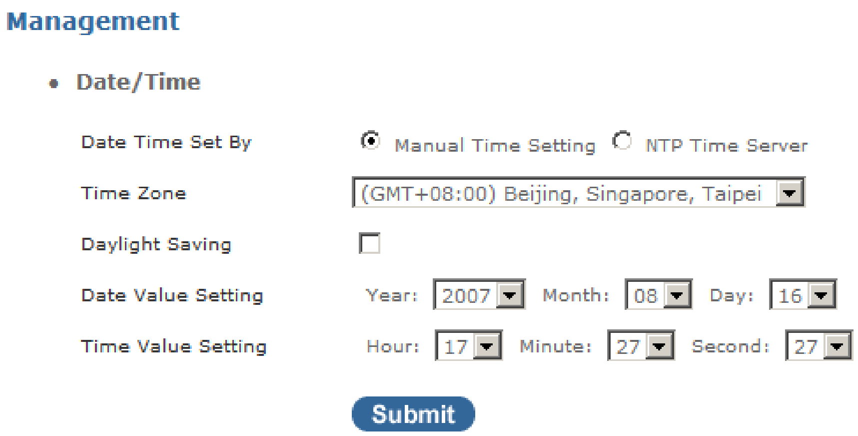


Figure 5-2. Date/Time-Manual time settings

Manual Time Setting	Set up the time manually.

Table 5-2. Date/Time-Manual time description

➢ NTP Time Server

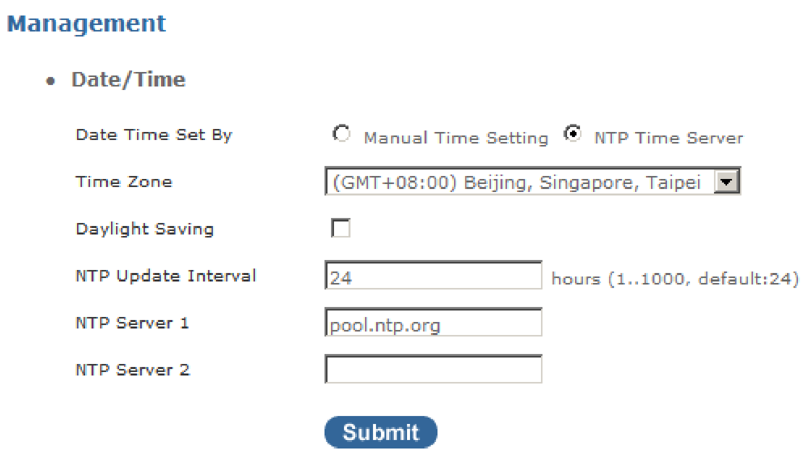


Figure 5-3. Date/Time-NTP time settings

NTP Time Server	Protocol used to help match your system clock with an accurate time source. For example atomic clock or a server.
Time Zone	Choose your time zone, Default is (GMT+8:00) Beijing, Singapore, Taipei.
Daylight Saving	Enable / Disable. Default is Disabling, time during which clocks are set one hour ahead of local standard time; widely adopted during summer to provide extra daylight in the evenings.
NTP Update Interval	Default is 24 hours; This is used to select the frequency of. NTP updates.
NTP Server 1	Default is "pool.ntp.org", NTP Server address.
NTP Server 2	Default is empty.

Table 5-3. Date/Time-NTP time description

Ping Test

This useful diagnostic utility can be used to check if a computer is on the Internet. It sends ping packets and listens for replies from the specific host. Enter in a host name or the IP address that you want to ping (Packet Internet Groper) and click Ping. *Example:* www.yahoo.com or 209.131.36.158

Figure 5-4. Ping test settings

Ping Destination	Assign a legal IP address.

Table 5-4. Ping test description

Save & Restore

All settings can be saving to a local file. Pervious device configuration can also be restored by upload a local file back to the device.

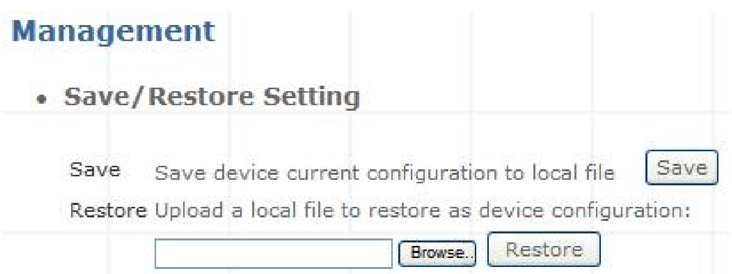


Figure 5-5. Save/Restore settings

Factory Default

This function is used to restore all the parameters back to factory default setting. You can use the Save/Restore Setting to check the factory default configuration, after you click on the Set button.

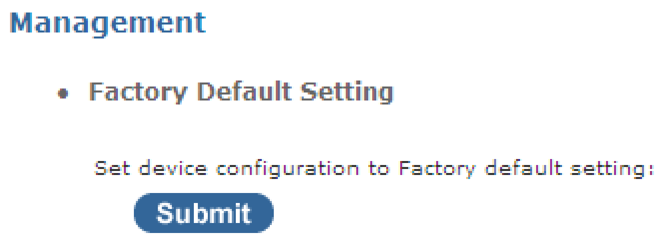


Figure 5-6. Factory default settings

Firmware Update

You can upgrade the firmware of the device using this tool. Make sure that the firmware you want to use is saved on the local hard drive of your computer. Click on Browse to search the local hard drive for the firmware to be used for the update. Upgrading the firmware will not change any of your system settings but it is recommended that you save your system settings before doing a firmware upgrade.

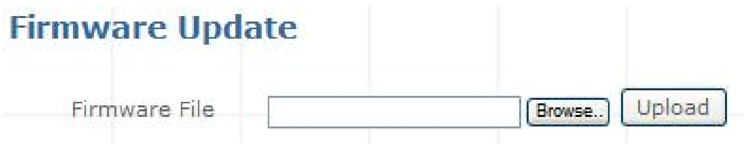


Figure 5-7. Firmware update settings

Firmware Name	Select that you want to upgrade Firmware version.

Table 5-5. Firmware update description

Chapter 6
Information

System Information

System Information page indicates the current setup-status of the device, it includes LAN, WAN, (Status and MAC Address), Host Name / System Date time / Machines Life time and system firmware information. The information and options on this page will vary according to your WAN setting (Static IP, DHCP, or PPPoE).

-If your WAN connection is set up for *Dynamic IP address*, the page will display “Release” and “Renew” buttons. Use “Release” to disconnect from your ISP and use “Renew” to connect to your ISP.

-If your WAN connection is set up for *PPPoE*, the page will display “Connect” and “Disconnect” buttons. Use "Disconnect" to drop the PPPoE connection and use "Connect" to establish the PPPoE connection

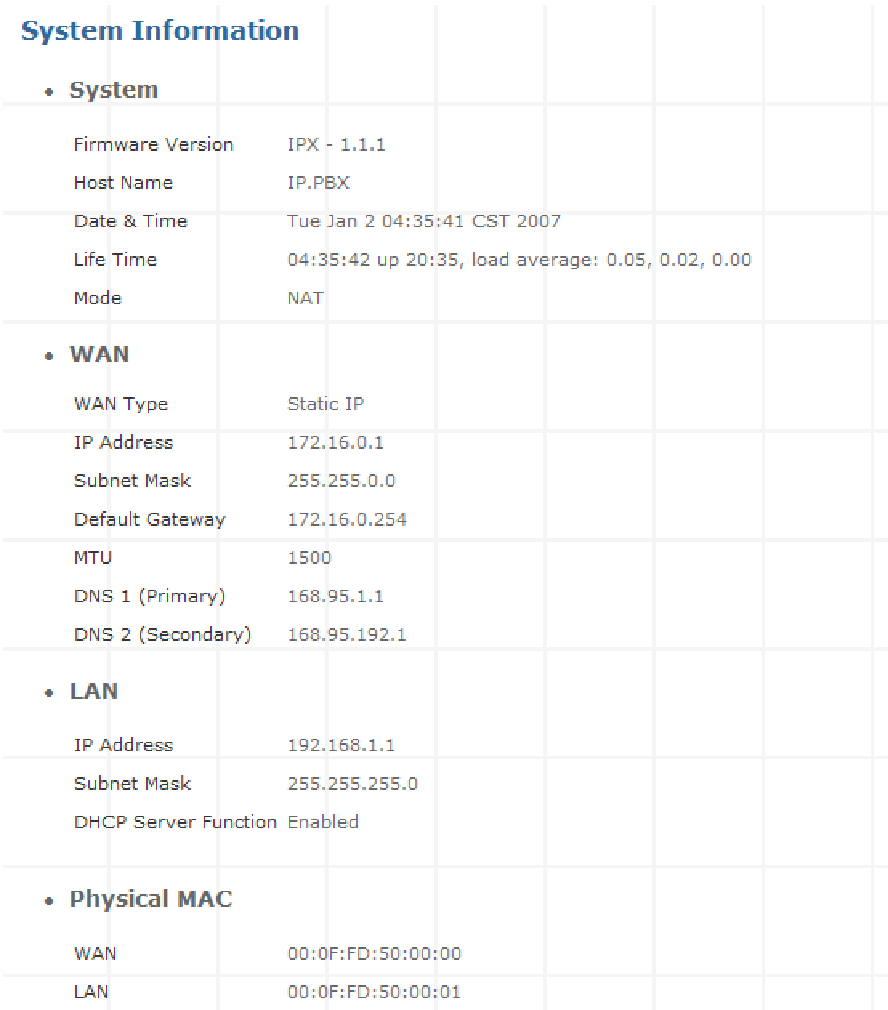
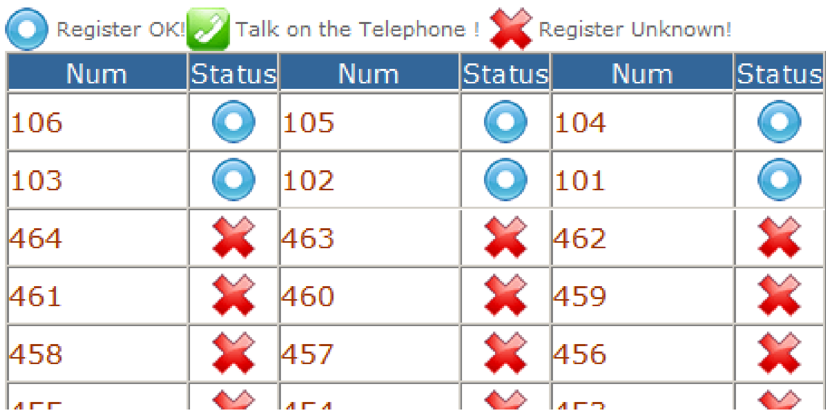


Figure 6-1. System Information

PBX Extension Status

This page displays the information of Extension/Users Registration status.



Num	Status	Num	Status	Num	Status
106		105		104	
103		102		101	
464		463		462	
461		460		459	
458		457		456	

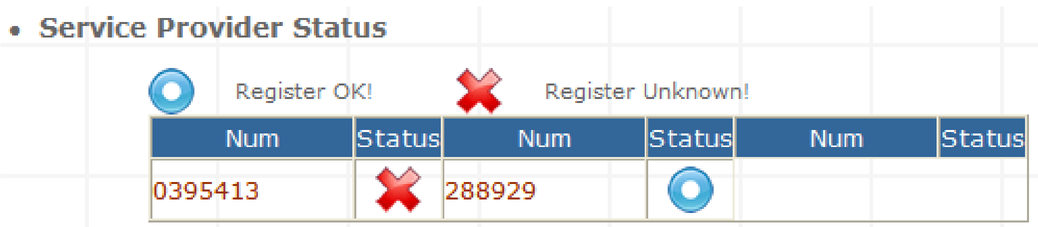
Figure 6-2. Extension Status

Register OK	SIP device is connected to IPPBX
Talk on the telephone	The connection from/to the other end of SIP device is established.
Register Unknown	Sip device is not connected to IPPBX

Table 6-1. Extension Status description

PBX Trunk Status

This page displays the information of Service Provider Registration status.



Num	Status	Num	Status	Num	Status
0395413		288929			

Figure 6-3. Service Provider Status

Register OK	SIP Trunk is registered
Register Unknown	SIP Trunk is not registered

Table 6-2. Service Provider Status description

Call Detail Record

Call Detail Record (CDR) contains the call history of the extensions when calls was made or received. Recorded information include: Source Number, Destination Number, Start Time, Answer Time, End Time, Duration Time and Status.

- Call Detail Record

<< [1] >>

Source No	Destination No	Start Time	Answer Time	End Time	Duraction Time	Status
200	100	2007-11-28 14:23:51	2007-11-28 14:23:51	2007-11-28 14:24:16	25	ANSWERED
100	out	2007-11-28 14:24:41	2007-11-28 14:24:42	2007-11-28 14:24:47	6	ANSWERED
2010	s	2007-11-28 14:24:42	2007-11-28 14:24:42	2007-11-28 14:24:47	5	ANSWERED
100	out	2007-11-28 14:24:52	2007-11-28 14:24:57	2007-11-28 14:24:58	6	ANSWERED
431	100	2007-11-28 14:29:06	2007-11-28 14:29:07	2007-11-28 14:29:11	5	ANSWERED
431	100	2007-11-28 14:30:12	2007-11-28 14:30:14	2007-11-28 14:30:26	14	ANSWERED

Figure 6-4. Call Detail Record

Press << to go to the Next page; Press >> to go to the Previous page

Source No	Caller's ID
Destination No	ID of destination extension / user
Start Time	The date/time when the call initiated
Answer Time	The date/time when the call answered
End Time	The date/time when the call terminated
Duration Time	Duration of the call, in seconds, from Start Time to End Time.
Status	4 status available (1) Answered; (2) No Answer; (3) Busy; (4) Failed.

Table 6-3. Call Detail Record description

Note

- IPPBX / WIPPBX have save Maximum 500 Records to the memory. If you press Reset bottom or reboot the system, the record will be erased.

Appendix A

How to use Call Parking function

The followings are the Call Park function settings, and all of VoIP devices (ATA, GW and IP Phone) were registered with Wi-Fi IP PBX.

- **Extension to Dial for Parking Calls: 700**
- **Extensions to park calls on :701-720**

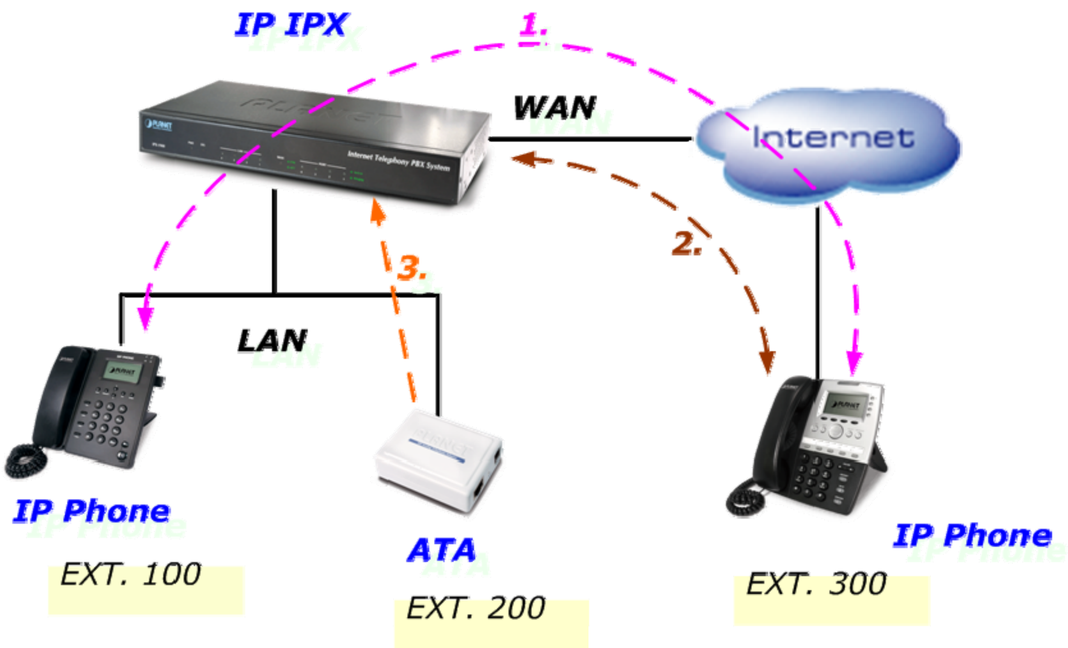


Figure A-1. Call Parking sample scenario

1. Ext.100 and Ext.300 are talking.
2. Ext.300 press Transfer button and dial “**700#**” to carry out the Call Parking function, and the voice guide will tell Ext.300 a retrieve number (ex:701) to set parking call (At this moment, the remote extension will hear the holding music.)
3. Ext.200 dial retrieve number (ex:701) to pick up call.
4. Ext.100 are talking with Ext.200

Appendix B

How to use Call Pick-up function

The followings are the Call Pickup function settings, and all of VoIP devices (ATA, GW and IP Phone) were registered with IP PBX.

- **Pickup Extension: *8**

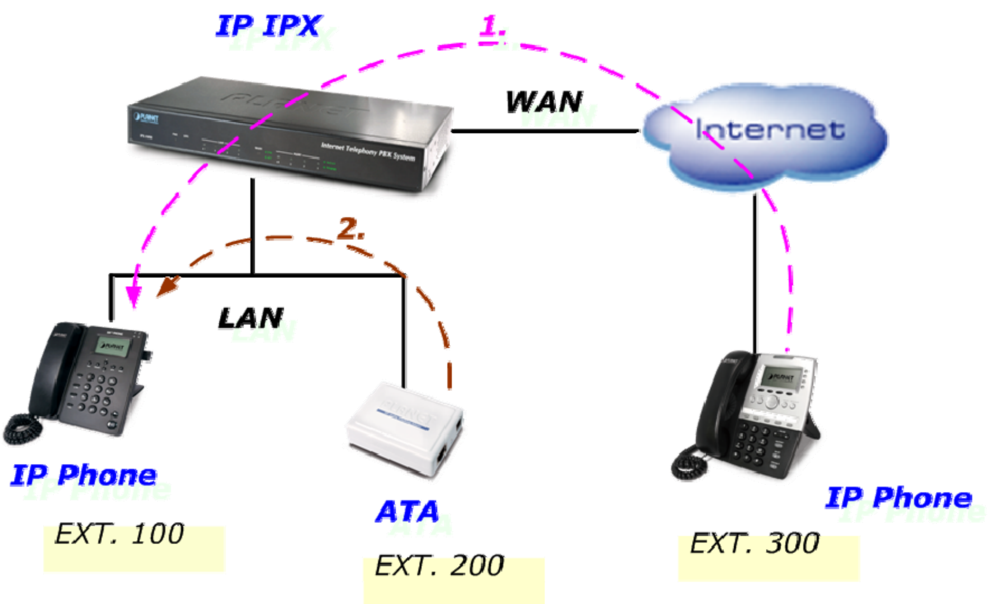


Figure B-1. Call Pickup sample scenario

1. Ext.300 call to Ext.100, and Ext.100 is ringing.
2. Ext.200 dial "***8#**" to pickup the call for Ext.100, and Ext.200 is talking with Ext.300.

Appendix C

How to record Sound and replacement Sounds package

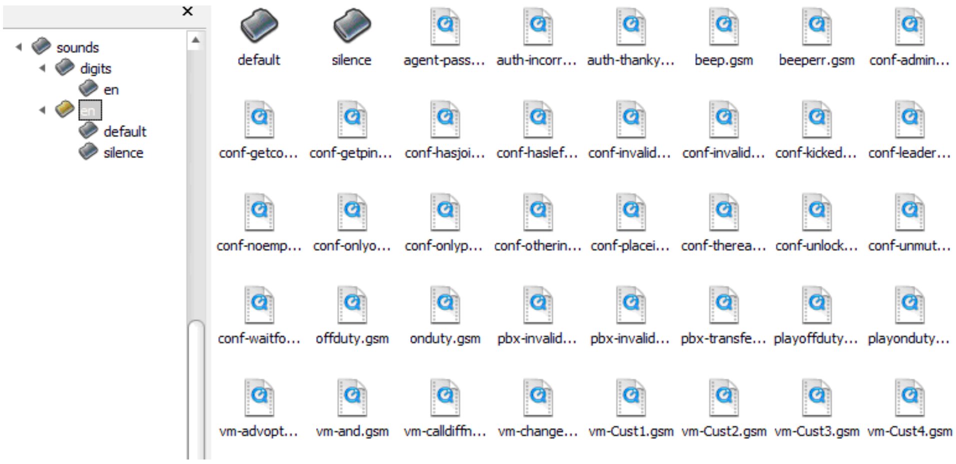
This sample for how to record sound as gsm format file for IP PBX use, and to replacement the Sounds package.

➢ **What do you need?**

1. Original Sound file in English version: **sounds.tar.gz** (*)
2. IVR menu Script: **IVR script.txt** (*)
3. A Software can record the sound file, such as Wavepad Editor (http://www.nch.com.au)
4. A Software can compress the folders into tar file, such as IZarc (http://Izarc.org)

(*): Please contact with our VoIP Technical Support Team (support_voip@planet.com.tw) for getting the related IVR files.

Step1: Uncompress the sound file **(sounds.tar.gz)**, and you will see the file architecture and what voice files included.



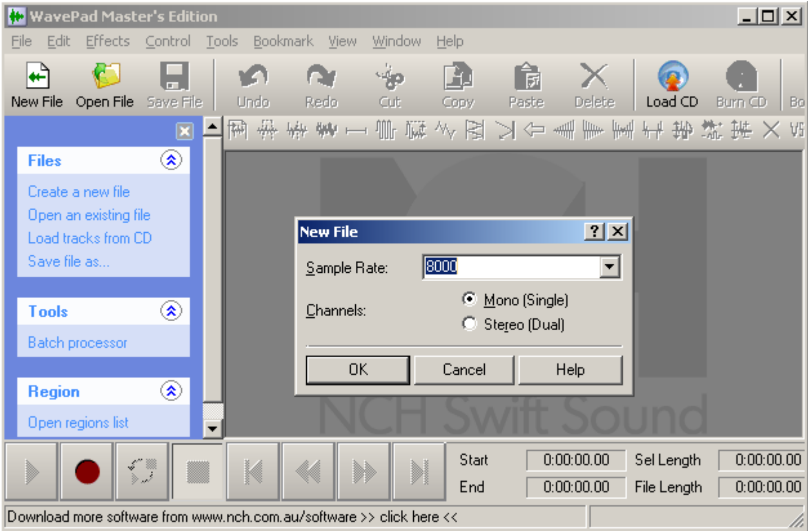
Step2: Find the script, and translate it to your language, first record the sound with Wavepad Editor Software.

Example: Thank you (English) ---→ Merci (French)

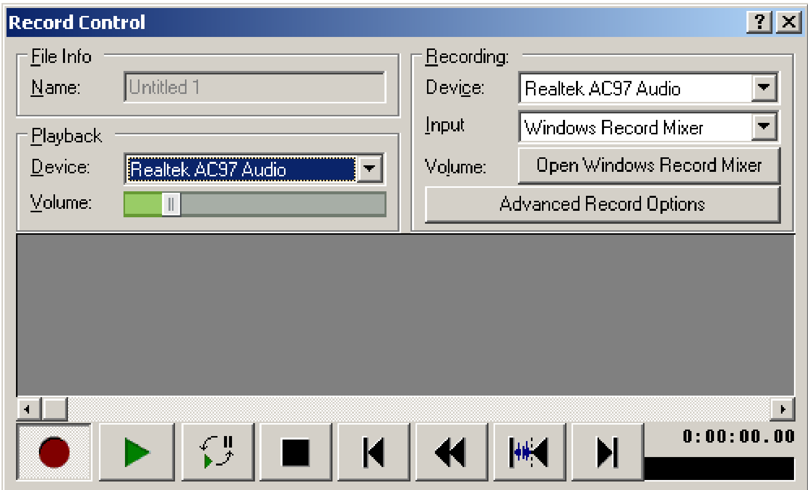
➢ **How to Record IVR?**

This sample is for how to record sound, as gsm format file for IP PBX series use.

1. Visit to www.nch.com.au sound home-page, to download Wavepad v3.05 (for windows) sound tools install your pc.
2. In this screen, create a new file, Input **8000**Hz and select **mono** (Single) channel then press ok to finish.

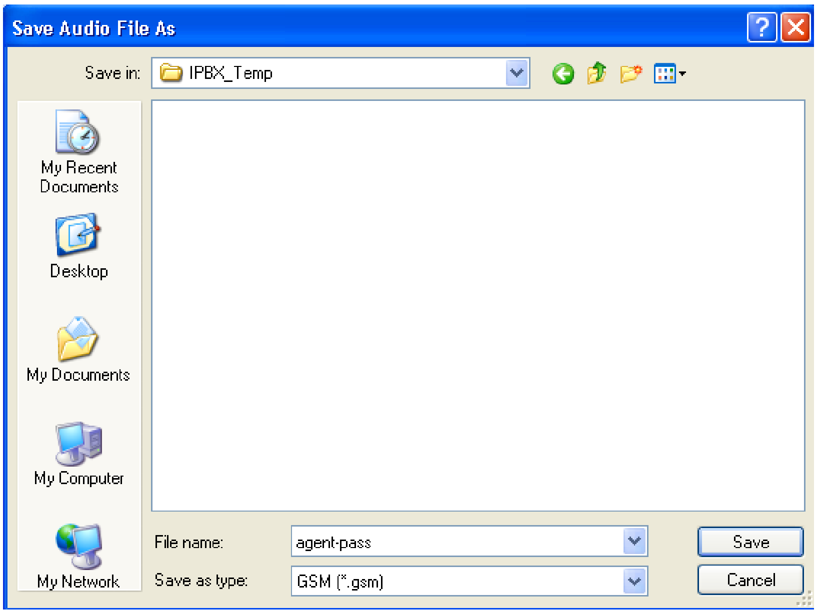


3. Press **F5** Select your recording sound device and record channel or recording volume level.



4. Press  Start recording

5. Save the current file as WAV or gsm format to finish.

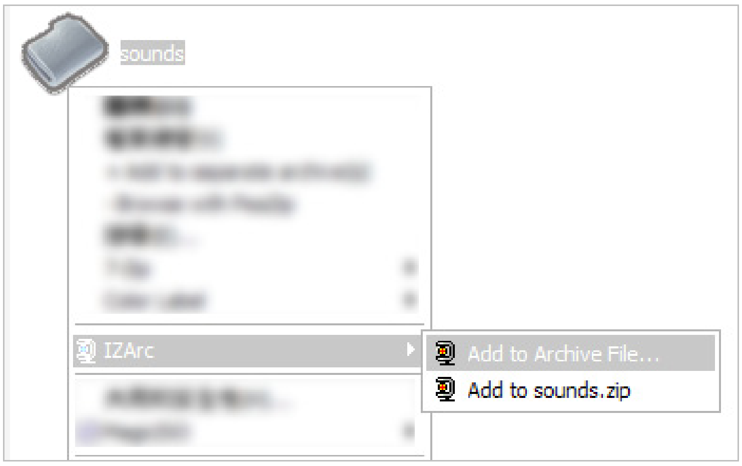


Step3: Replace the original file, and please don't change the file name.

Step4: Compress the sounds folder to a zipped file with using izarc, and please reverence the following for the steps.

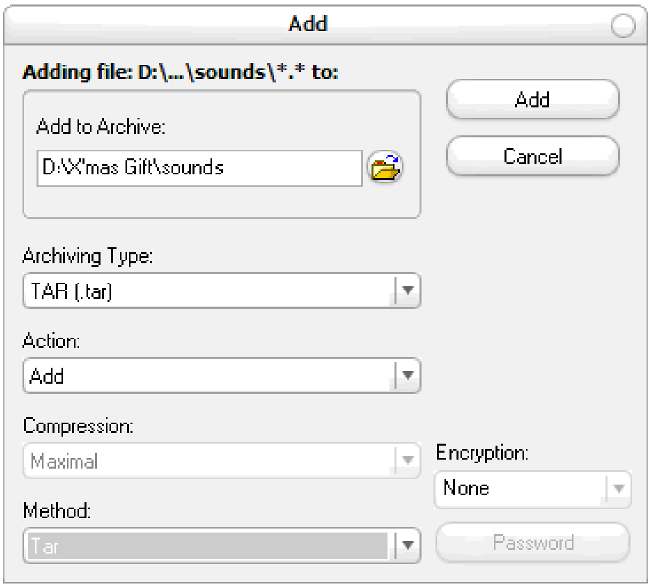
1. After installed Izarc.

2. Right Click and then select "**Add to Archive File...**".



3. Select "**TAR (.tar)**" for the Archiving Type, and "**Tar**" for the Method, and then click "**Add**" to create the compress file.

Note: Please don't change the file location of the sounds folder.



Step5: Please login the Web UI of IP PBX, and select **Voice Management** - > **Upload Voice File**, and then click the "**Browse**" button to allocate the "**sounds. tar**" on your PC. Once the file is selected, please click "**Upload** "to start the upgrade process. Once the upgrade is complete you can start using your devices.

Appendix D

Record Voice Guide Process

IPX-1900 provides **Record Voice Menu by Phone** function. Please register your VoIP devices to Wi-Fi IP PBX at first, and then check the Record voice code from "**IP PBX Setup -> record Voice Menu**" page.

Figure C-1. Record voice menu settings

VoIP devices dial ***9** to entry the Record Voice Menu, then refer to the following record processes to record the Voice Menu.

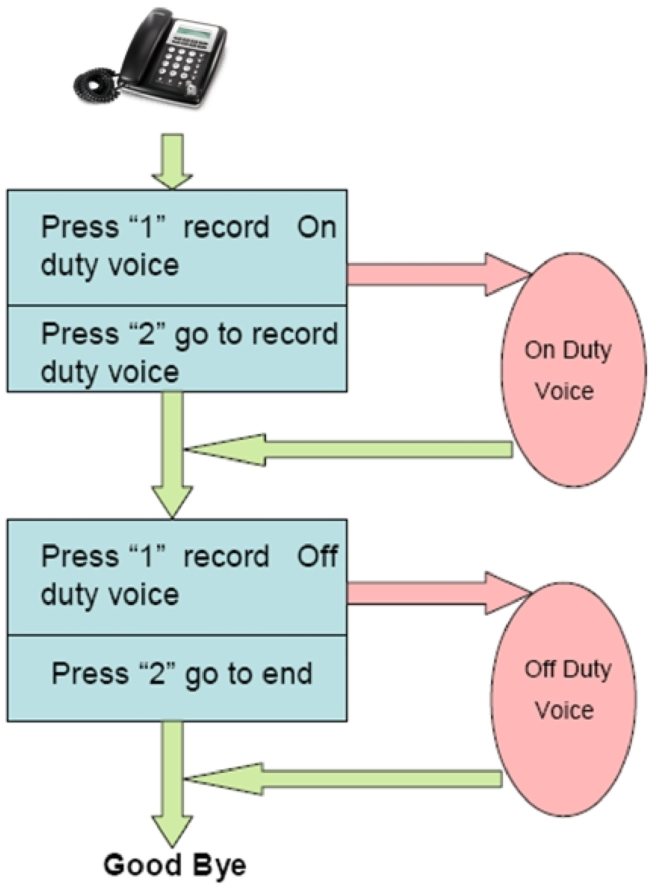


Figure C-2. Voice record processes

Appendix E

Voice Communication Samples

The chapter shows you the concept and command to help you configure your IP PBX System through sample configuration. And provide several ways to make calls to desired destination in IP PBX. In this section, we'll lead you step by step to establish your first voice communication via web browsers operations.

IP Phone register to IPX-1900

In the following samples, we'll introduce IP Phone register to IP PBX applications.

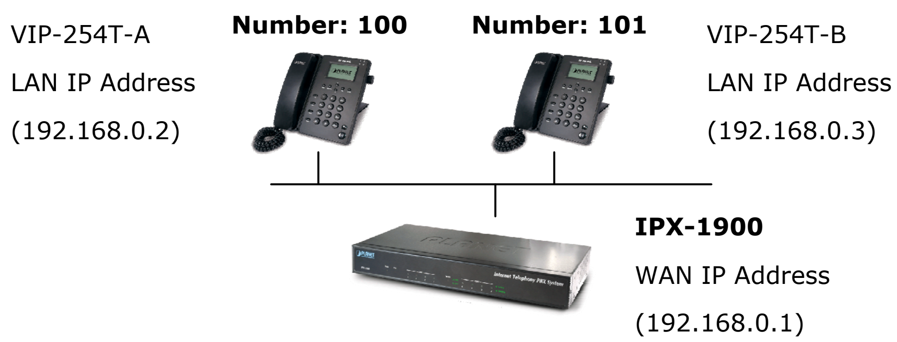


Figure D-1. Topology of instruction example

- **Machine Configuration:**

STEP 1:

Browse to "**IP PBX Setup → User Extensions Setup**" configuration menu.

Figure D-3. User extension setting of IP PBX

STEP 2:

Click the "**Add**" button to create extension account ext.100 and ext.101.

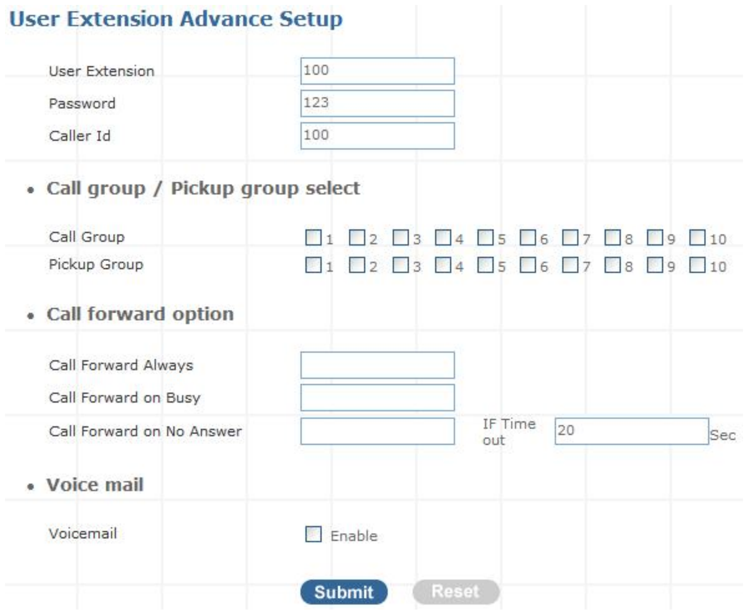


Figure D-4. Add extension setting of IP PBX

STEP 3:

Please log in VIP-254T_B and browser to "**SIP setting → Domain Service**" configuration menu. Insert the account/password information then save and reboot machine. The sample configuration screen is shown below:

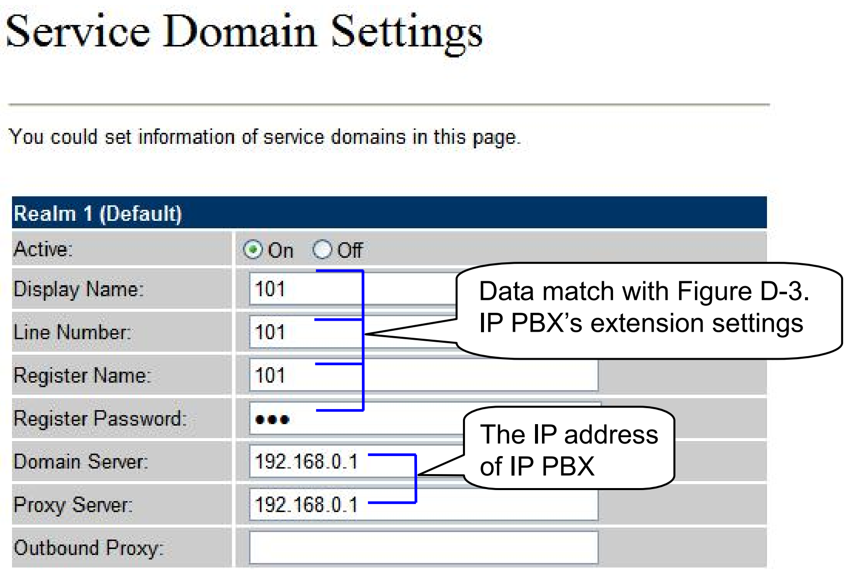


Figure D-5. Web page of VIP-154T

STEP 4:

Repeat the same configuration steps on VIP-254T-A, and check the machine registration status, make sure the registrations are completed.

STEP 5:

After both of devices have registered to IP PBX successfully, it could browse to “**Information -> PBX Extension Status**” page to show the registration status:

Figure D-8. Extension status

➢ **Test the Scenario:**

1. VIP-254T_B pick up the telephone
2. Dial the number: 100 shall be able to connect to the VIP-254T_A
3. Then the VIP-254_A should ring. Please repeat the same dialing steps on VIP-254_B to establish the first voice communication from VIP-254T_A

IP Phone make off-Net calls via Gateway

In the following samples, we’ll introduce VIP-154T and VIP-192 makes off-Net Calls (PSTN calls) via VIP-480FO applications.

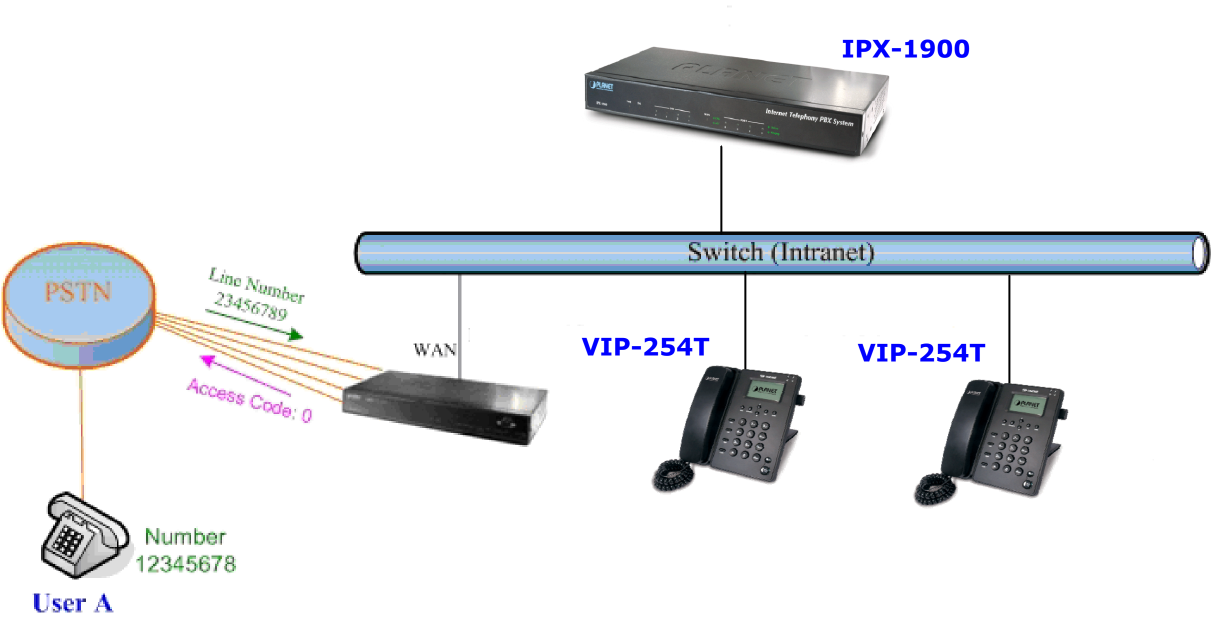


Figure D-9. Installation example with VIP-480FO

➢ Machine Configuration:

STEP 1:

Please refer to the first sample and let VIP-154T and VIP-192 register to IP PBX.

STEP 2:

Please log in IP PBX via web browser and browse to "**IP PBX Setup → User Extensions Setup**" configuration menu to add four accounts for VIP-480FO using.

Figure D-10. Add accounts for VIP-480FO

STEP 3:

Browse to "**IP PBX Setup → Attendant Extension**" configuration menu. Assign an attendant number which inexistence extension in Extension List and the sample configuration screen is shown below:

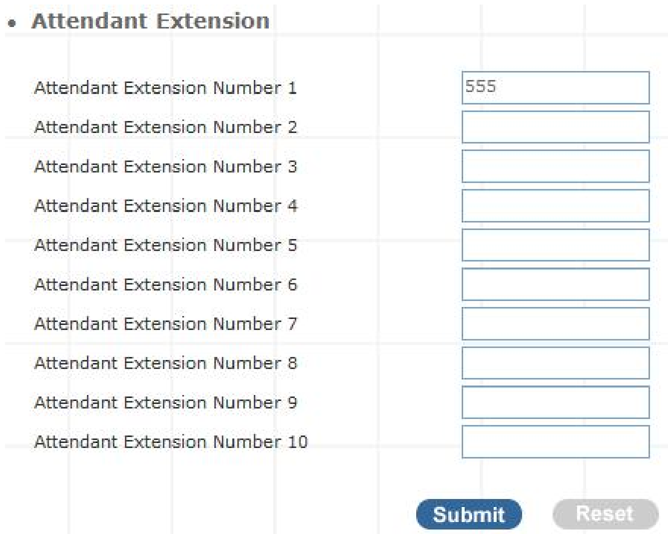


Figure D-11. Assign an attendant number

Pressing the "**Submit**" button for activate the configuration.

STEP 4:

Browse to "**IP PBX Setup → Trunk Management → Gateway Trunk**" configuration menu. Fill in the IP address of VIP-480FO for connecting with VIP-480FO by peer-to-peer mode, and press the "**Insert**" button for activate the configuration.

- **Gateway Trunk Setting**

Add Gateway trunk Gateway trunk Max is 10

IP	Port	Action
192.168.0.12	5060	Insert Change

Figure D-12. Add a Gateway trunk for connecting with VIP-480FO

STEP 5:

Browse to "**IP PBX Setup → Trunk Management → Trunk Group**" configuration menu. Add a Trunk Group for making off-Net calls via VIP-480FO.

- **Trunk Group Setting**

Add New Grop Name Add

Group Name List Trunk Group Max is 10

Group Name	Group Number	Number	Action
VIP-480FO	0	192.168.0.12:5060	Edit Delete

Figure D-13. Add Trunk Group number for grabbing the FXO ports of VIP-480FO

STEP 6:

Please log in VIP-480FO via web browser and browse to "**Advance Setup → VoIP Setup → VoIP Basic**" configuration menu. Insert the account/password information and set up the hunting function. The sample configuration screen is shown below:

Port Number / Password Setting(MAX 20 digit) :

No.	Number	Reg	Account	Password	Register Status	Reason
1	200	☑	200	•••	Success	OK
2	201	☑	201	•••	Success	OK
3	202	☑	202	•••	Success	OK
4	203	☑	203	•••	Success	OK

Figure D-14. Set up the number of FXO ports of VIP-480FO

SIP Hunting Table :

No.	Hunting Member
1	☑ Port 1 ☑ Port 2 ☑ Port 3 ☑ Port 4
2	☑ Port 1 ☑ Port 2 ☑ Port 3 ☑ Port 4
3	☑ Port 1 ☑ Port 2 ☑ Port 3 ☑ Port 4
4	☑ Port 1 ☑ Port 2 ☑ Port 3 ☑ Port 4

Figure D-15. Set up the Hunting Member of FXO ports

SIP Proxy Setting :

Domain/Realm	192.168.0.1
SIP Proxy Server	192.168.0.1/5060 ☐ use net2phone
Register Interval(seconds)	900
SIP Authentication	◉ Enable ○ Disable
Outbound Proxy Server	0.0.0.0/0

Figure D-16. Set up the Proxy Server IP address for register to IPX-1900

STEP 7:

Browse to "**Dialing Plan**" configuration menu. Add an Incoming Dial Plan (no.1x) for redirect the PSTN outgoing calls to FXO ports.

Incoming Dial Plan: (maximun 50 entries, maximun length of prefix digits is 16 digit, maximun length of number is 20 digit):

Item	Incoming no.	Length of Number	Delete Length	Prefix no.	Destination telephone port	Operation
1	1x	2 ~ 20	0	None	1	
		~				ADD

DELETE Inbound Dial Plan	From To

Figure D-17. Add an incoming dial plan

STEP 8:

Browse to "**Port Status**" configuration menu. Fill in the auto attendant number **555** to all of ports. (Where 555 is the auto-attendant number of IP PBX)

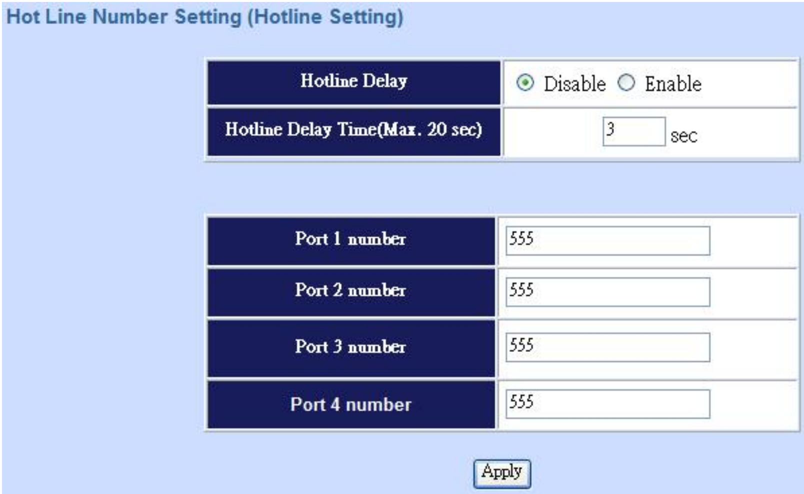


Figure D-18. Hot Line to auto-attendant of IPX-1900

STEP 8:

After all of devices have registered to IP PBX successfully, the **Extension Status** page will show the registration status:

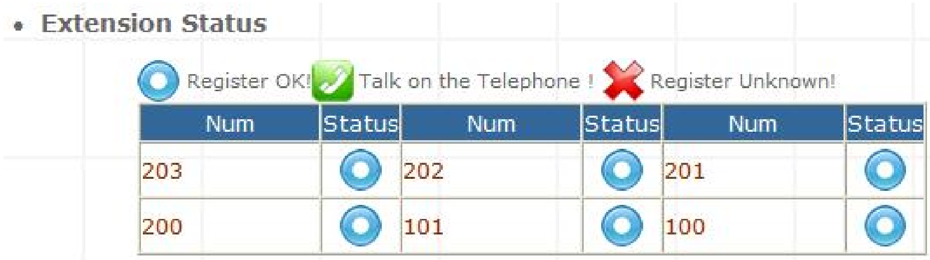


Figure D-19. Extension status page with Phone and Gateway registered

- **Test the Scenario:**

1. VIP-154T pick up the telephone
2. Dial the number: **0** will hear the dial tone, and dial the number: 12345678. This call will hunt the FXO port of VIP-480FO and shall be able connect to the User A.
3. Then the telephone of User A will ringing, User A can pick up the handset and talk with VIP-154T.
4. Both VIP-154T and User A hang up the calls.
5. User A pick up the telephone and dial the number: 23456789 should be able to connect to the Auto Attendant System of IP PBX.
6. The User A will hear the prompts, and dial the extension number: 100 shall be able connect to the VIP-192.
7. Then the VIP-192 should ringing, and it to pick up the call then talk with User A.

IP Phone make external SIP Proxy calls via SIP Trunk

In the following samples, we'll introduce VIP-154T and VIP-192 makes SIP Proxy calls via SIP Trunk applications.

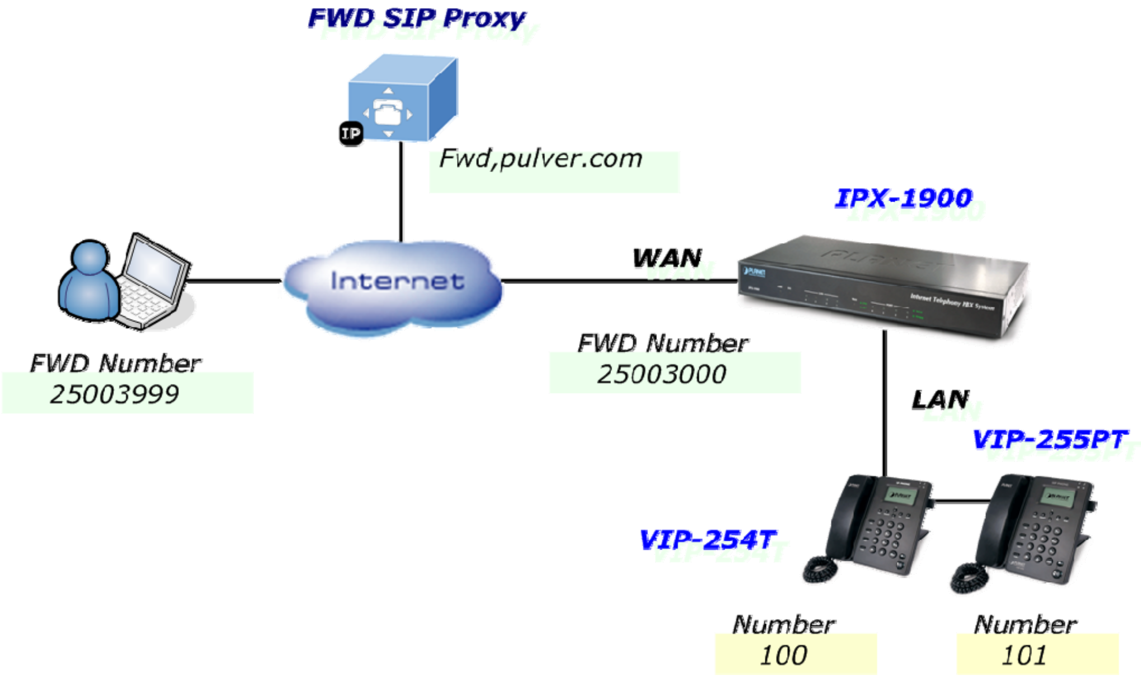


Figure D-20. Installation example with FWD SIP Prxoy

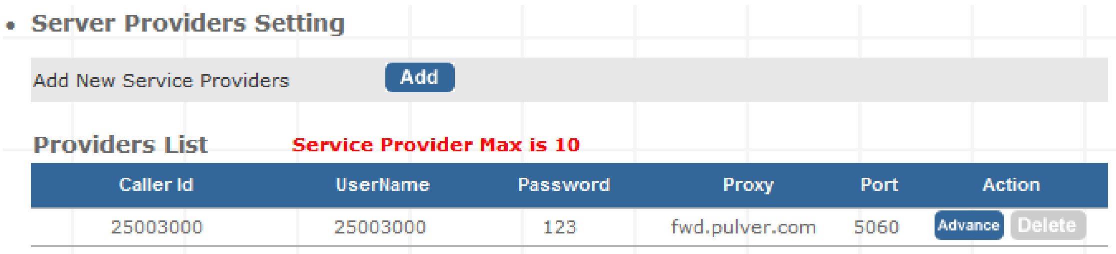
- **Machine Configuration:**

STEP 1:

Please refer to the first sample and let VIP-254T and VIP-255PT register to IP PBX.

STEP 2:

Browse to "**IP PBX Setup → Trunk Management → SIP Trunk**" configuration menu. Add a new Service Provider account for registering to FWD SIP Proxy.



• Server Providers Setting

Add New Service Providers Add

Providers List Service Provider Max is 10

Caller Id	UserName	Password	Proxy	Port	Action
25003000	25003000	123	fwd.pulver.com	5060	Advance Delete

Figure D-21. Add a Service Provider account

STEP 3:

Browse to "**IP PBX Setup → Trunk Management → Trunk Group**" configuration menu. Add a Trunk Group for making external SIP Proxy calls.

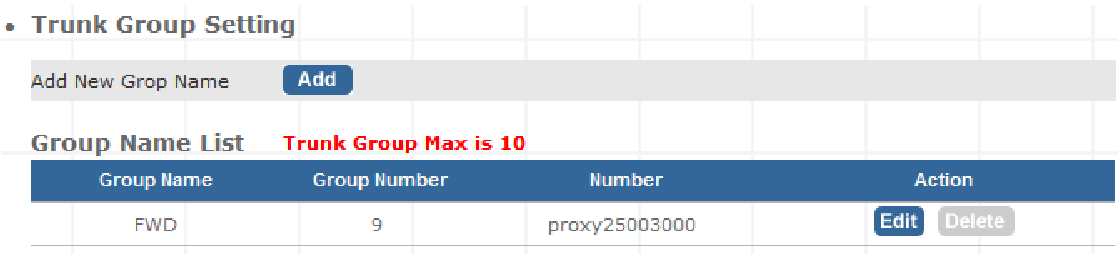


Figure D-22. Add Trunk Group number

STEP 4:

After the SIP Trunk has registered to FWD SIP Proxy successfully, the **Service Provider Status** page will show the registration status:

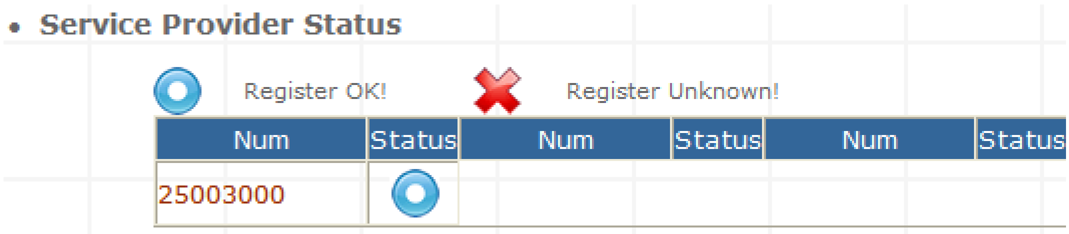


Figure D-23. Service Provider status page

- **Test the Scenario:**

1. VIP-154T pick up the telephone
2. Dial the number: **9** will hear the dial tone, and dial the number: 25003999. This call shall be able connect to the User B.
3. Then the softphone of User B will ringing, User B can answer the call and talk with VIP-154T.
4. Both VIP-154T and User B hang up the calls.
5. User B pick up and dial the number: 25003000 should be able to connect to the Auto Attendant System of IP PBX.
6. The User B will hear the prompts, and dial the extension number: 100 shall be able connect to the VIP-254T.
7. Then the VIP-254T should ringing, and it to pick up the call then talk with User B.

Appendix F

IPX-1900 Series Specifications

Product	Internet Telephony PBX System
Model	IPX-1900
Hardware	
LAN	1 RJ-45 (10/100Base-TX, Auto-Sensing/Switching)
WAN	1 RJ-45 (10/100Base-TX, Auto-Sensing/Switching)
Standards and Protocol	
Call control	SIP 2.0 (RFC3261) , SDP (RFC 2327), Symmetric RTP
Registration	Max. 300 nodes / SIP IP phones/ ATA / FXO gateways
Calls	Max. 60 concurrent calls
Voice CODEC Support	G.723, G.726, G.729, G.711, GSM, iLBC
Voice Processing	DTMF detection and generation In-Band and Out-of-Band (RFC 2833), (SIP INFO) Supports password authentication using MD5 digest
PBX features	Auto Attendant (AA) Interactive Voice Response (IVR) Records IVR via IP Phone Voicemail Support (VM) Voicemail Send to E-mail Call Detailed Record (CDR) User Management via Web Browsers Web Firmware Upgrade Backup and Restore Configuration file Call/Pickup Group Displays 300 Registered User's Status: Unregistered / Registered / On-Call Displays 60 Registered Trunk's Status: Unregistered / Registered Fax Support using G.711 Pass-Through or T.38**
Call features	Caller ID Call Group Call Hold Call Waiting Call Transfer Call Forward (Always, Busy, No Answer) Call Pickup Call Park Call Resume Music on Hold Three-way conference with feature phones (VIP-254T series, VIP-255PT, 351PT and ATA series: VIP-156/ 157/ 158 / 161W)

Internet Sharing	
Protocol	TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, NAT, DHCP, PPPoE, DNS
Advanced Function	NAT/Bridge mode, DHCP server, Static Route, DMZ, Virtual Server, Port Trigger, Packet / URL Filter, UPnP, DDNS, SNMP, Ping test
Network and Configuration	
Connection Type	Static IP, PPPoE, DHCP
Management	HTTP Web Browser
LED Indications	System: 1, PWR WAN: 1, LNK/ACT LAN: 4, LNK/ACT Line: 4, In-Use/Ringing
Environment	
Dimension (W x D x H)	340 x 159 x 40 mm
Operating Temperature	0~40 degree C, 0~90% humidity
Power Requirement	12V DC
EMC/EMI	CE, FCC Class B
Remark: T.38 support is dependent on fax machine, SIP provider and network / transport resilience	

Appendix G

IPX-1900 Module Card Specifications

IPX-19FO Card Technical Specifications

Signaling	Loop Start / DTMF
No. of channels	2
Interface Connectors	2 RJ-11 2-pin modular jacks
AC Impedance Selection	600 Ω /900 Ω / Global Impedance / Sixteen Impedance for selection.
Receive Frequency Response	Low –3 dBFS Corner, FILT = 0 , 5 Hz Low –3 dBFS Corner, FILT = 1, 200 Hz
Return Loss	≥ 25 dB , 300–3.4 kHz, all ac terminations
digital gain/attenuation adjustment	–16.5 to 13.5 dB
Tranhybrid Balance	≥ 20 dB, 300–3.4 kHz, all ac terminations
Polarity Reversal Detection	Support Battery Reversal Detection

IPX-19FS Card Technical Specifications

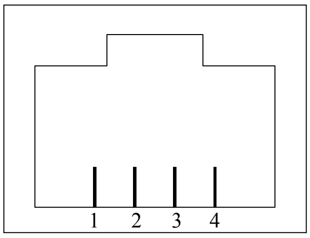
Signaling	Loop Start / DTMF
No. of channels	2
Interface Connectors	2 RJ-11 2-pin modular jacks
Line Impedance	600 Ω 900 Ω
Return Loss	Min 30db, 200 Hz to 3.4 kHz
Gain/Attenuation	Digital Programmable from mute~6db
Metallic to Longitudinal Balance	Min. 40 db, 200 Hz to 3.4 kHz
DC Loop Current Accuracy	29mA nominal
Ring Voltage	44Vrms Nominal
Ringing Tone	16.667Hz, 20Hz /30Hz / 40Hz/ 50Hz / 60Hz
REN	5
Pulse metering	12k/16k hz

Note: The IPX-19SL module card signaling same as IPX-19FO/FS specifications.

FXO Port Pin Assignments

The FXO Telephony Interface has 2 RJ-11C/W modular jacks. The following diagram and table show the assignments of the pin for the R-J11 port.

RJ-11pin	Signal
1	
2	Tip
3	Ring
4	



FXS Port Pin Assignments

The FXS Telephony Interface has 4 RJ-11C/W modular jacks. The following diagram and table show the assignments of the pin for the RJ-11 port.

RJ-11 pin	Signal
1	Not connected
2	Tip
3	Ring
4	Not connected