Embedded IP-PBX (EIP)

IP-PBX Manual

March 13, 2008

Version: 1.09

Index

Packing List

You should find followings inside the packaging:

- Embedded IP-PBX *1
- AC100~240V/DC12V Adapters *1
- CD-Manual *1
- RJ-45 Cross-over Cable *1

Please contact the local distributor if there is any item missing.

1. Introduction

Thanks for purchasing of the Series Embedded IP-PBX. is based on SOHO and SME's communication needs to design. It has all features supported by traditional PBX, and it also support IP phones and other powerful features. Besides saving the communication between branches within the corporate, it can also save cost of other long-distance phone call. Moreover, the IP PBX can combine the information system and communication system of a corporate, providing more value to the corporate.

The setup of is very simple; it can be set up based on the manual, without the need of professional IT engineers. Please read through the manual before the setup.

2. Product Introduction

2.1 Product Introduction

IP-PBX is using IP packet exchange to fulfil what the traditional PBX has been doing. The design is based for enterprise-level phone system, it can bridge all branches within a global corporate, saving corporate cost and improving corporate efficiency.

an IP-PBX based on Embedded System infrastructure, making it – Low Cost, High Stability, Compact Size and Rich it in feature.

2.2 Physical interface

- **Specification**
  - Power: AC100V-240V, DC12V/1A,50/60 Hz
  - **Working Temperature**: 0°C~50°C (32°F~122°F) (Running)
  - **Humidity**: 5% to 90% non-condensing
  - **Safety**: FCC Part 15 Class B, CE Mark
  - **Telecommunication Certifcate**: FCC Part 68
  - **Dimension**: 260(Length) x 150(Width) x 43(High) mm
  - **Weight**: 1300g

- **System**
  - 4 * FXO **interface**
  - Max 120 Registered Users
  - 2 10/100Base-T interface (WAN, LAN)

- **Protocol**
  - Protocol: SIP
  - Support CODEC: G729a, GSM/MS-GSM, Speex, G711a, G711u

- **Call Function**
  - Call Transfer/Call Forward
  - UMS – Unified Messaging System
  - Pick up for same group
  - Call Limit
  - Conference Room
  - Call Queue
  - IVR
  - Personalized Prompt
  - DMZ
  - Call Forward to Mobile Phone
  - Built-in Web call support

- Voice Mail (Check Online or Send to email)
- Group Ringing, Rotation, Based on Call Volume, etc
- Billing Rate
- CDR (Export support)
- Customizable IVR through XML
- NAT Detection and Pass-through
- PBX Configuration backup and restore
- Support SIP/ENUM
- Support PPTP VPN
- Support CBCOM Encryption
- WEB Management
- Built-in Auto Provision Server for IP Phone

3 Hardware Installation

3.1 Front Panel

LED	
LAN	Both LED ON - 100Mbps One LED ON - 10Mbps
WAN	Both LED ON - 100Mbps One LED ON - 10Mbps
Power	LED ON – Power Supply On
Active	LED ON – PBX Software Running
ALARM	LED ON – Error in PBX
VPN	LED ON VPN Running
FXO 1 - 4	LED ON – FXO Module Ready LED Blinking – FXO Module in operation

3.2 Back Panel

Image 2: Back Panel

3.3 Network Connection

1. WAN Port

- Connect to External Network
- Support DHCP, Static IP and PPPoE

2. LAN PORT

- Connect to Internal network
- Support DHCP Server

3. Typical Network Setup

In case there is firewall in your corporate, please open the following ports for the IP-PBX:

- TCP: 22, 53, 80, 1723
- UDP: 53, 5060 (or other customized SIP port), 1194, 10000-20000

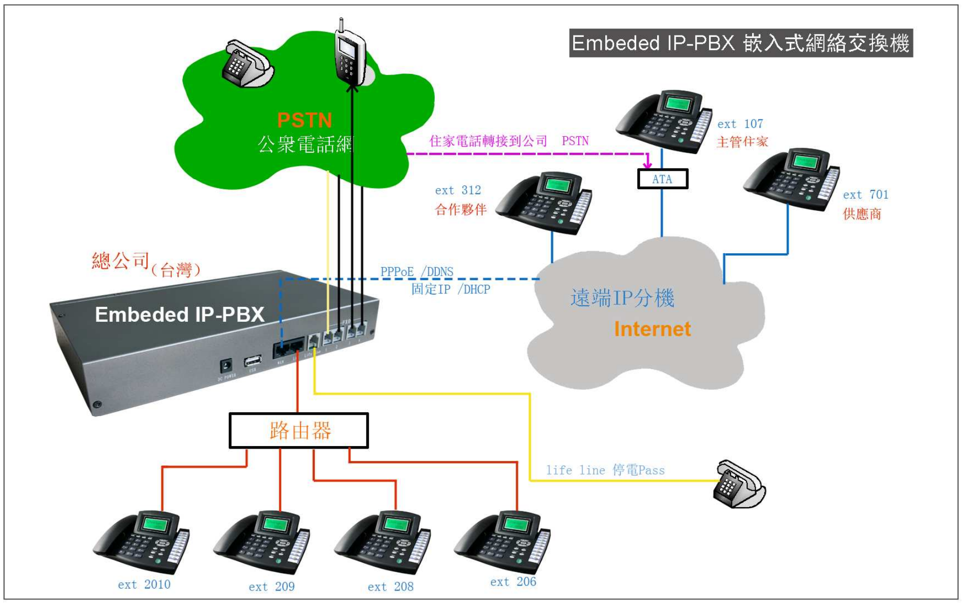


Image 3: Network Setup Example

Embedded IP-PBX support PPPoE broadband dial-up, if there is a dedicated broadband for the IP-PBX, please connect the IP-PBX directly to the ADSL Modem.

3.4 FXO Interface

FXO interface is used to connect the IP-PBX to traditional landline network (PSTN), making the corporate able the bridge up traditional and IP Phone network.

FXO-1 (The first port of FXO) has PSTN back-up support. When Embedded IP-PBX is out of power off, FXO-1 would be auto-connect to the Life-line port; user can connect a standard phone to Life-line port, and make call through FXO-1 port.

3.5 USB Interface

USB interface can be connected to an USB disk/flash drive. The USB disk can then be used to store voicemail, please refer to chapter 5 for details.

3.6 Reset to default setting

If your system has error and IP address is not known, you can reset the IP-PBX to default settings:

1. Unplug the power

2. Pressing the reset key on the front panel, and plug in the power. Wait for 30 seconds, and you should find the LEDs flashing for 3 times. You can then release the reset key, and the reset procedure is completed.

3. Wait for a minute, and the system would then be restarted, and LEDs would be flashing for 3 times again.

Some key default settings

WAN:192.168.139.3

LAN:192.168.1.1

Username: admin

Password:12346

Please be noted that:

1. All user information would be reset after reset procedure
2. This would restore the firmware and IP settings to default

4. Default Settings

4.1 Network setup

NIC	IP Address	Subnet Mask
WAN	192.168.139.3	255.255.255.0
LAN	192.168.1.1	255.255.255.0

Note: The network interface of IP-PBX does not support AUTO MDI/MDI-X, please use the cross-cable if you would like to connect the IP-PBX to a computer directly.

4.2 Administrator Management

IP-PBX provide WEB Management Interface.

Login Procedure: Inside your web browser (Internet Explorer for example), and enter IP-PBX IP address like following:

http://60.248.176.205

Default Username: admin

Default Password: 123456

Inside each page of WEB management interface, there is a help button. You can click the help if you need guidance.

4.3 User Management Console

Each IP-PBX user (extension) can login the IP-PBX to configure their own settings, and query their own call records.

1. Use web browser and enter the address of the IP-PBX (e.g. http://192.168.1.1/)

Enter the extension and the password

4.4 IP-PBX Configuration

The default SIP Port for IP-PBX is 5060. There are 10 default user accounts, ranged from 2001-2010, all of their default password is “123456”

5 IP-PBX User Guide

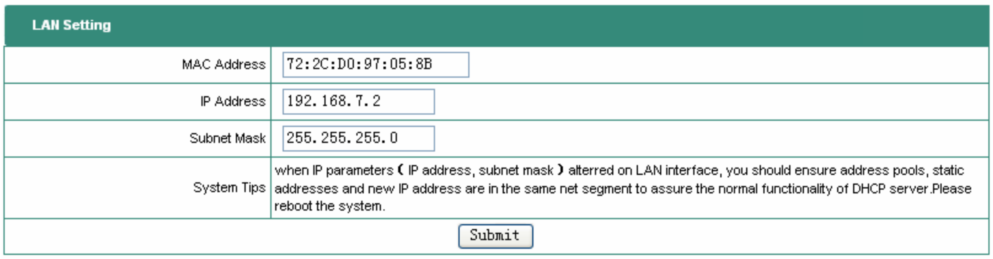
This chapter would explain the instructions of IP-PBX in detail.

5.1 Network Setup

Network Setup is the key for making IP-PBX work properly. Note that LAN and WAN port does not support AUTO MDI/MDI-X detection, so a cross-cable is required to connect directly to a PC. It is ok if both the IP PBX and your PC is connect to a switch.

5.1.1 LAN PORT Setup

LAN PORT is used to connect to local network. The LAN Port support DHCP Server; you can use the IP-PBX to assign IP address the IP phones connected to the same network. The default DHCP Server is disabled. 。



LAN Setting	
MAC Address	72:2C:D0:97:05:8B
IP Address	192.168.7.2
Subnet Mask	255.255.255.0
System Tips	when IP parameters (IP address, subnet mask) alterred on LAN interface, you should ensure address pools, static addresses and new IP address are in the same net segment to assure the normal functionality of DHCP server.Please reboot the system.
Submit	

MAC is physical address of the LAN PORT, you can modify it if necessary.

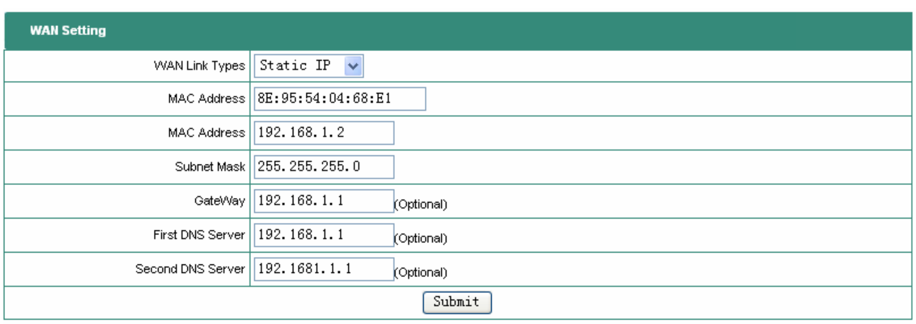
IP address is in format of xxx.xxx.xxx.xxx , like 192.168.1.1

IP netmask is used to partition network segment. Please make the netmask of the IP-PBX is the same as the netmask of other equipment in the same local network. If you are not clear about the net mask, please use the default setting.

5.1.2 WAN PORT

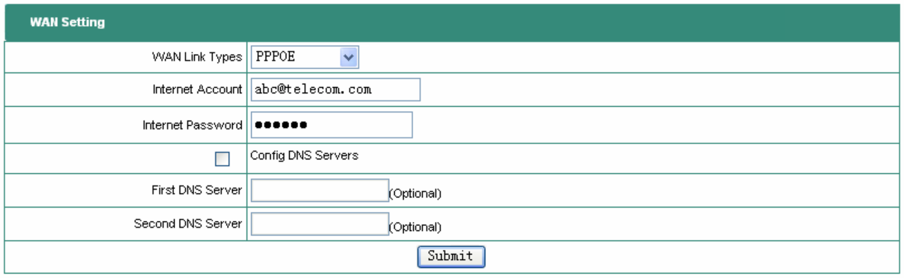
WAN PORT is the default route of the IP-PBX. This is mainly difference between WAN and LAN port. WAN PORT supports: DHCP, Static IP or PPPoE, you should choose the appropriate mode based on your network setup.

If you need to Static IP Mode, please refer to following settings:



WAN Setting	
WAN Link Types	Static IP
MAC Address	8E:95:54:04:68:E1
MAC Address	192.168.1.2
Subnet Mask	255.255.255.0
GateWay	192.168.1.1 (Optional)
First DNS Server	192.168.1.1 (Optional)
Second DNS Server	192.1681.1.1 (Optional)
Submit	

If you're connecting the IP-PBX directly to the ADSL, please use PPPoE mode, and fill up the broadband account username and password. You can add your preferred DNS server if necessary.



5.1.3 DHCP Server

IP-PBX support built-in DHCP Server, it can assign IP to the network equipment (e.g. IP Phone) on the same network segment.

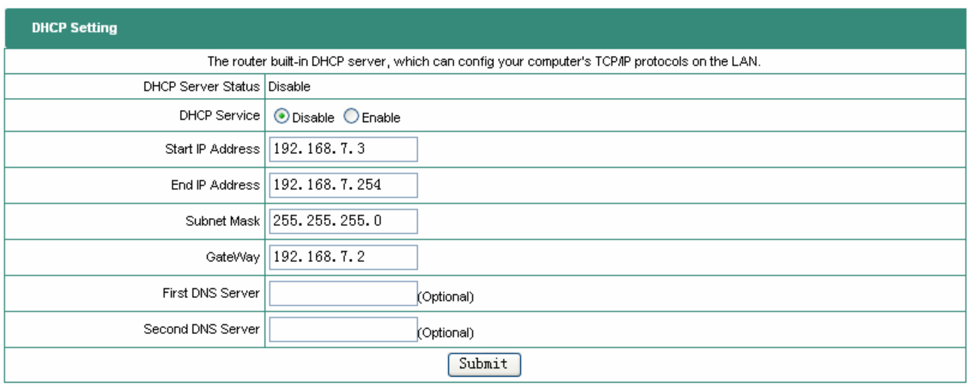
DHCP Server Status: Showing “Enable” if the DHCP Server is enabled.

Start IP Address/End IP Address: IP addresses range to be assigened (e.g 192.168.1.100 – 192.168.1.254)

Subnet Mask: This would be automatically filled. Not suggested to be changed unless necessary.

Gateway: The gateway when the network equipment is required to reach Internet. Usually is the router of the network.

DNS Server: The server responsible for domain name resolution.



5.1.4 DDNS (Dynamic DNS)

IP-PBX support DynDNS and cn99 Dynamic Domain

Server Provider: Provider for the Dynamic DNS Service, like DynDNS or cn99

Host Name: The hostname of Dynamic DNS

User Name: Username of the Dynamic DNS account

Password: Password for the Dynamic DNS account

5.1.5 VPN Setup

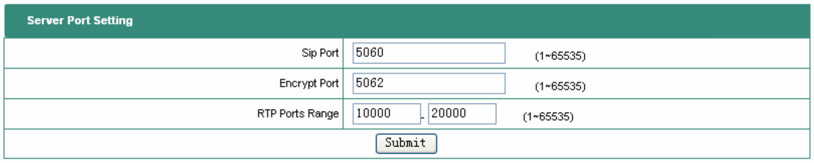
IP-PBX can be configured as a VPN server, or VPN Client.

5.2 System

This chapter would introduce some parameter setting of the IP PBX. Under “System” Menu, there is “SIP Port”, “Rate set”, “DMZ”, “Trust Host”, “Music on Hold”, “HotLines”, “Admin Account”, “USB Disk Setting”, “Voicemail Setting” and “Timezone”.

5.2.1 SIP Port

The SIP Port is used for communication with the terminals and the IP PBX; this port is used for signaling messages. The default port is 5060; you can set it between 1 and 65535. Default value is recommended to be used.



Note: In case the port is changed, you have to change to the SIP port setting on ALL terminals (like IP Phone, ATA) as well.

5.2.2 Rate Set

Rate set is used for measuring call rate, each call have a call record, recording its charge of call. The charge is calculated based on the call rate set here. Call rate is based on the call prefix, for example:

Rate Prefix: 0

Rate: 15 (Minute)

Unit (Unit: Seconds): 60

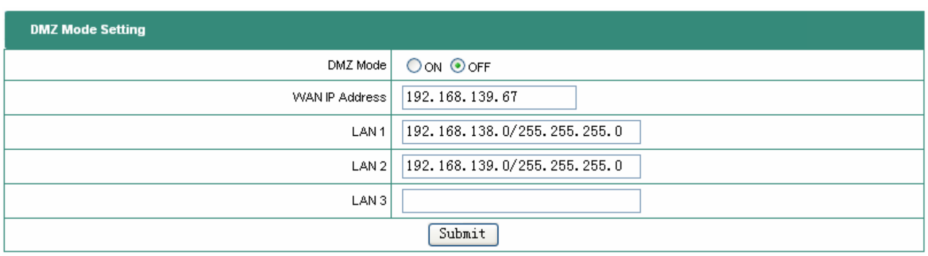
When an extension call a number start with 0 (like 0010086), the call is be charged for 15 cents per 60 seconds. Less than 60 seconds would be counted as 60 seconds.

Besides, the matching rule is choosing the rule has longest matching prefix. If there are rules of prefix 0 and prefix 00, when you make a call of 0010086, the rule of prefix 00 would be chosen.

If there is matching rule, then the call would not be charged.

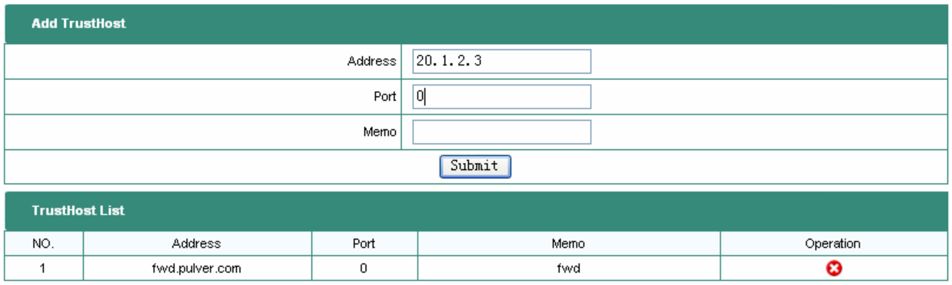
5.2.3 DMZ

DMZ is the short form of "demilitarized zone", it is used to solve problem of LAN device not able to be contacted from Internet.



5.2.4 Trust Host

Calls from Trust Host can be accepted with authorization. If trust host port is 0, that means the all port from the trust host can be accepted, otherwise please enter the SIP port of the trust host. Memo is for user's own use, it be left empty.



Add TrustHost

Address	20.1.2.3
Port	0
Memo	
Submit	

TrustHost List

NO.	Address	Port	Memo	Operation
1	fwd.pulver.com	0	fwd	

Trust host is normally used under following situations:

1. Integration with FXO gateway
2. Integration with other IP PBX
3. Integration with other SIP Server

In short, the Trust Host is should set whenever the incoming call does not require authorization.

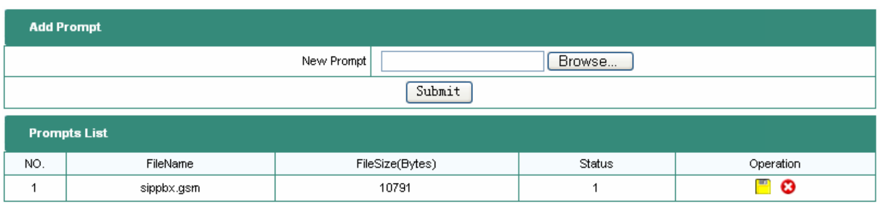
5.2.5 Music on Hold

Music on hold is used when the incoming call is on hold. It can be used on following situation:

1. When incoming call connecting to the operator, but the operator is busy
2. When incoming call is put on the Queue and waiting to be proceed.
3. When incoming is on transfer.

You can choose any music as the On Hold music. On Hold music has to be saved a .gsm format. Note that mp3 file is not supported. Please ask the distributor for the file conversion tool in order to convert the mp3 file to this .gsm format.

1. Enter the "Auto Attend"->"Voice Prompt"



Add Prompt

New Prompt	Browse...
Submit	

Prompts List

NO.	FileName	FileSize(Bytes)	Status	Operation
1	sippbx.gsm	10791	1	

2. Click "Add Prompt", select the gsm file you would like to use

5.4 Switchboard

When someone calls the switchboard of the PBX, switchboard will direct the call either to a customizable voice prompt, or to a certain extension/phone number. The configuration is made inside this switchboard settings.

5.4.1 Prompt

You can set the voice prompt of the switchboard under this menu.

1. Under “Add Prompt”, select the voice-prompt file you would like to use; Click submit to upload
2. In case you have multiple voice-prompt, click file icon to choose the one to be used actively. The status of the actively used voice prompt should be “1”

5.4.2 Operator

When a caller needs to query the extension of a user or need help, they can call the operator of the IP PBX.

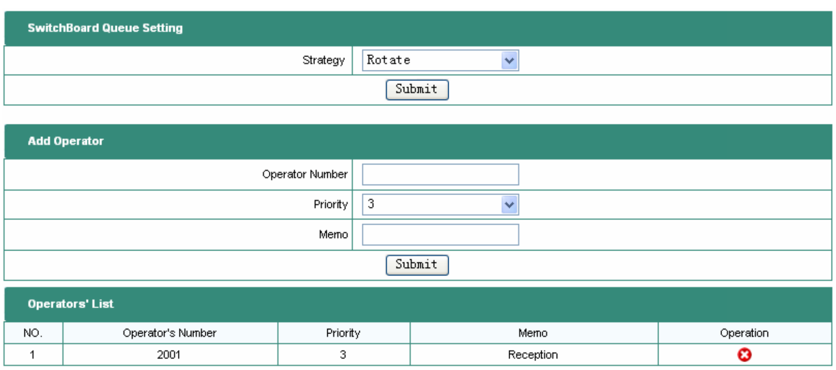
Strategy: When there is more than one operator, there are different strategies available for use

- Rotate: Operators will be ringed in rotation, ordered by the “priority” of the operators. Each operator would be ringed for 20s before ringing next one.
- Ring-all: Ring all the operator’s phone
- Random: A random operator’s phone would be ringed

To add an operator:

1. Under the “Operator Number”, enter the extension of user or the phone number chosen to be operator
2. Enter the priority of the operator. Refer to the above strategy for use of priority. Leave it default normally
3. Memo is an optional field for mnemonic use

Note that not only a extension but any phone number (e.g. mobile phone) can be used a operator.



5.4.3 Auto Attend

PBX can response to the customer when there no one in the office (e.g. after office hour). Different voice-prompts can be used on different hours.

There are two ways for Auto Attend: Prompt or Phone

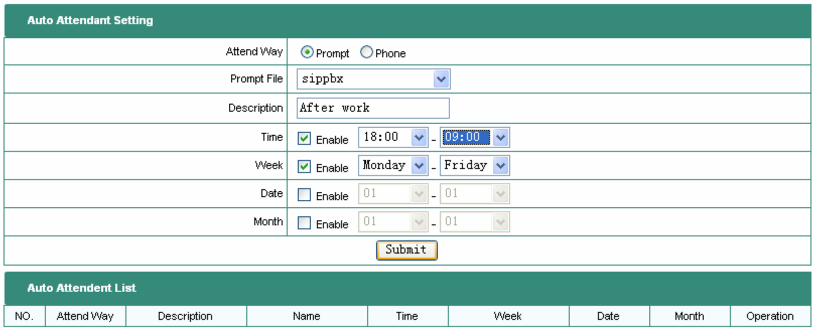
Phone: When a phone calling in, it will be forwarded to certain extension or phone (e.g. mobile phone)

1. Choose "Phone" as the Attend Way

2. Enter the Phone Number to be used, can be an extension or a phone number

3. Choose the time range to use this Auto Attend. ***If none of the fields (time, week, date, months) are enabled, this Auto Attend will be active for all the time.***

4. Click Submit when completed



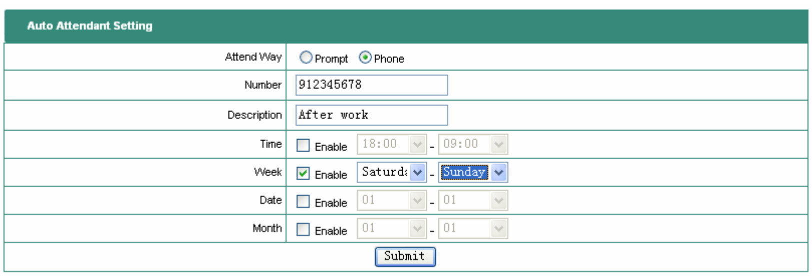
Prompt Auto Attend: When a phone calling in, a prompt will be used to response.

1. Choose "Prompt" as the "Auto Attend Way"

2. Choose a prompt to be used. The prompt can be customized. Please go to "Switchboard"->"Prompt" to add new prompts.

3. Choose the time range to use this Auto Attend. ***If none of the fields (time, week, date, months) are enabled, this Auto Attend will be active for all the time.***

4. Click Submit when completed



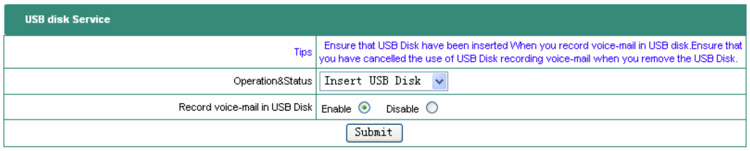
5.2.7 USB-Disk Setting

Our IP PBX supports using USB Disk to extend voicemail storage.

1. Plug in USB-Disk
2. Under "System"->"USB-Disk Setting"->"Operation and Status", choose "Insert USB Disk"
3. Select "Enable" for the "Record voicemail in USB Disk"
4. Submit for update.

If you would like to disable the voicemail storage extension, and remove the USB-Disk:

1. Under "System"->"USB-Disk Setting"->"Operation and Status", choose "Remove USB Disk"
2. Select "Disable" for the "Record voicemail in USB Disk"
3. Submit for update.
4. Unplug in USB-Disk



5.2.8 Voicemail Setting

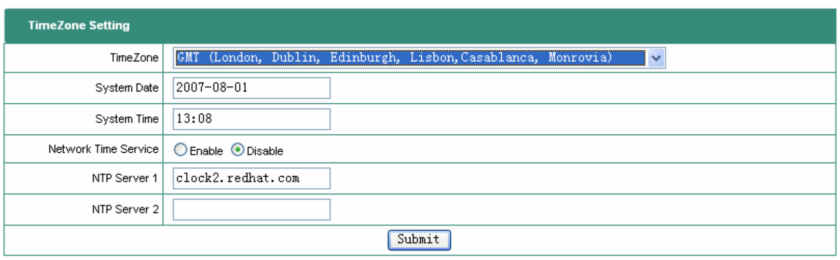
IP PBX supports forwarding the voicemail to user through email. You can set the outgoing mail server under this page.

- Mailbox - Mailbox Name (e.g. myippbx@netvigator.com)
- User - User Name of the Mailbox if authentication is required (e.g. myippbx@netvigator.com)
- Password - Password of the Mailbox if authentication is required
- SMTP Server - SMTP Server name (e.g. smtp.netvigator.com)

Click “submit” and Reboot the system to take effect.

5.2.8 Time zone

Time-zone and time setting can be configured under this page. Incorrect time setting would affect the CDR records. NTP server is supported.



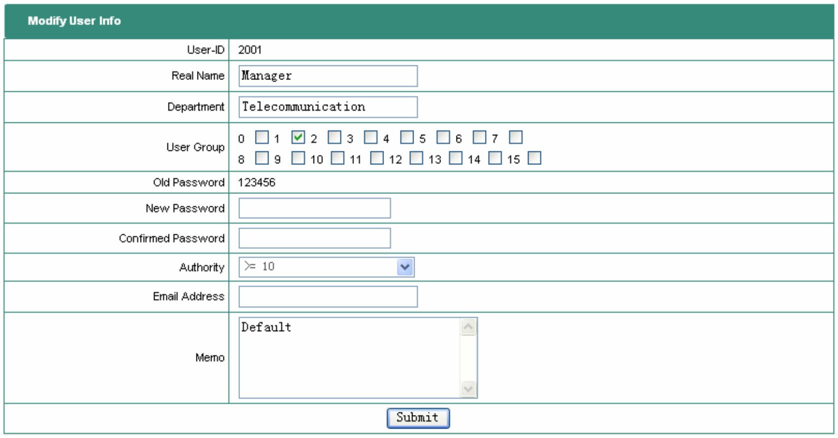
5.3 Users

This chapter we will introduce the user management of the IP PBX, including adding extension, user management and call records

User-ID (Same as Extension Number) should be a number less than 32 digits. It is recommended to have planning on the extension numbering, for example, different departments should use extension starting with a certain prefix.

A general user would have following attributes:

- User-ID: A number less than 32 digits
- Password: A 32-digit of number or alphabet, used to authenticate the user. This have to be set on the terminal device (e.g. IP Phone or ATA)
- Authority: It is a number between 0 and 15. Different call numbers has different authority (refer to the Incoming call and Outgoing call settings), only users of higher or equal authority can make such call.
- Group: Each user belongs to one or more groups. User of same group can pick up call for each other, by pressing “8” when a phone of the group rings. We support 16 groups.

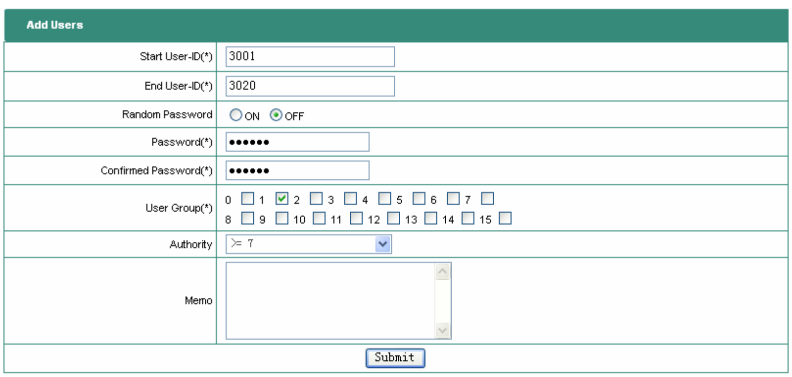


5.3.1 Add User

You can add a user under this page, based on the attribute mentioned above. Note that adding user would fail if the number of users exceeds license limitation.

5.3.2 Add Users

You can add a range of users under this batch mode. For the password, you can use a same password for all accounts, or let the system to generate random passwords.



5.3.3 Bindings (Group Ringing)

You can bind several extensions to a Binding ID. When a phone rings, the other phones of same binding ID would ring together. This is called as Group Ringing.

- User-ID: Extension Number
- Binding-ID: An arbitrary number

Note: An extension is allowed to be bind with several Binding ID

5.3.4 Delete User

User can be deleted one by one, or deleted in a range.

Note that information of deleted user cannot be recovered.

5.3.5 Information Update

Information of user can be updated through this page.

5.3.6 Function Setting

We offer four functions for each user

1. **Call Forward**
    a. Forward All: All incoming call would be forwarded instantly to the forward number
    b. No Answer Forward: Incoming call would be forwarded after ringing for 40 seconds
    c. Busy Forward: Incoming call would be forwarded when the user is busy
    d. Offline Forward: Incoming call would be forwarded when the user is offline.

    Note on Forwarding to an External Phone Number

    For external phone number (e.g. Phone Number), please make sure a FXO outgoing prefix is added in front of the number. For example, you would like to bind your mobile phone (98354409) to this extension. On your IP PBX's "Outgoing Calls through FXO", Prefix '9' is needed to call through FXO. In this case, you should put "998354409" as the forward number.

2. **Find Me (Same as Follow Me)**

    When enabled, the incoming call would be forwarded if the phone is not answer for 40 seconds. The call would be forwarded to first Find Me number, and ring for a number of seconds (depends on the settings). If it is not answer again, the number would be forwarded to next number in the same fashion. There is 5 find me number in total for each user.

    Note on "Find Me" with an External Phone Number

    For external phone number (e.g. Phone Number), please make sure a FXO outgoing prefix is added in front of the number. For example, you would like to bind your mobile phone (98354409) to this extension. On your IP PBX's "Outgoing Calls through FXO", Prefix '9' is needed to call through FXO. In this case, you should put "998354409" on the "Find Me" List.

3. **Binding User-ID**

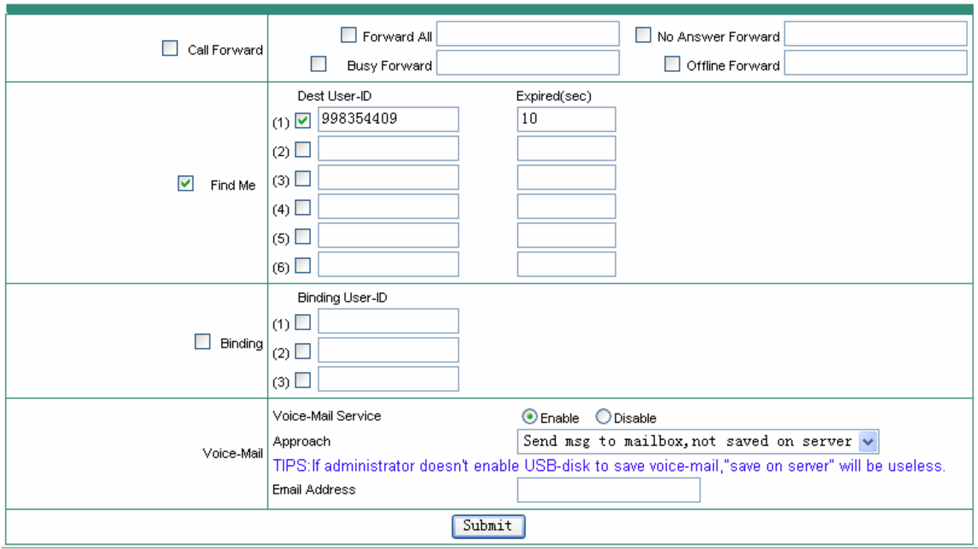
    Our IP PBX can bind several extensions or external phone number (e.g. Mobile Phone) together; all these number would ring together if there is incoming on this extension. If any one of the binding phone picks up the call, the other phones would stop ringing.

    Note on Binding an External Phone Number

    For external phone number (e.g. Phone Number), please make sure a FXO outgoing prefix is added in front of the number. For example, you would like to bind your mobile phone (98354409) to this extension. On your IP PBX's "Outgoing Calls through FXO", Prefix '9' is needed to call through FXO. In this case, you should put "998354409" on the Binding List.

4. **Voicemail**

    Voicemail can either be saved on server, or can be forwarded to a certain email. Please set the "System"->"Voicemail" setting for outgoing mail server address.



Note:

1. ***Only one of the three functions can be used***
2. ***Remember to click the checkbox to enable the function***
3. ***For external phone number, make sure the phone number has the prefix defined in the dialing rule. For example, if 0 is used to dial PSTN, making the external number has the prefix 0.***

5.4 Checking Voicemail

1. To enable voicemail notification, please enable the “Subscribe for MWI” for the terminal (IP Phone or ATA)
2. Under “User Management>>Function Setting”, Enable the Voice Mail
3. The Voicemail indicator on the terminal would turn on when there is a voice mail. User can press “1604” to check the mail. The IVR will prompt user to enter the extension number and the password. Then press “1” to listen the voicemail.

   Note:

   - If there is USB-Disk plugged in, each extension can leave 5 voicemail of 60 seconds.
   - If there is a USB-Disk plugged in, each extension can leave 50 voicemail of 60 seconds

Using our IP Phone and ATA, user can define soft-key to check the voicemail.

5.5 Outgoing Call

Our IP PBX provides way for outgoing calls: Outgoing call through FXO and Outgoing call through VoIP:

5.5.1 Outgoing call through FXO (Landline Call)

Our IP PBX supports 4 FXO port, you use connect these FXO port to telephone landlines (your local phone service provider).

FXO Group Settings

We allow the FXO port to be grouped into different FXO groups. An FXO port has to be put in one of the groups in order to be used. After setting the FXO Group, you need to setup a dialing rule of that FXO Group. If more than one FXO ports are put in the same FXO Group, the IP PBX will automatically use the free one.

FXO Groups

FXO Group NO.	Group 1	Group 2	Group 3	Group 4	Group 5	Group 6	Group 7	Group 8
FXO1	☑	☐	☐	☐	☐	☐	☐	☐
FXO2	☐	☑	☐	☐	☐	☐	☐	☐
FXO3	☐	☑	☐	☐	☐	☐	☐	☐
FXO4	☐	☐	☑	☐	☐	☐	☐	☐

Submit

Current Outgoing Rules List Via FXO [Add Outgoing Rules via FXO]

NO.	Authority	FXO Group NO.	Prefix	Strip Bits	Append Prefix	Operation
1	>= 0	Group1	9	1		

Outgoing Rules via FXO

Click the "Outgoing Rules via FXO" link below the FXO Group table to add a rule. Inside the rule settings:

- Authority: Only user of equal or higher authority can make outgoing call of this rule (refer to the user settings)
- Group No: The FXO group no that would be used by this rule matches the dialing number (Refer to the FXO Group setting)
- Dial Prefix: When the number dialed matches this prefix, this rule would be used (e.g. 9)
- Dial Strip bit: Number of digit to strip on the number to be dialed out on FXO
- Append Prefix: Number(s) to be put in front of the dialed number
- Extern User Control: Whether to let external incoming call to use this rule to dial out

Example:

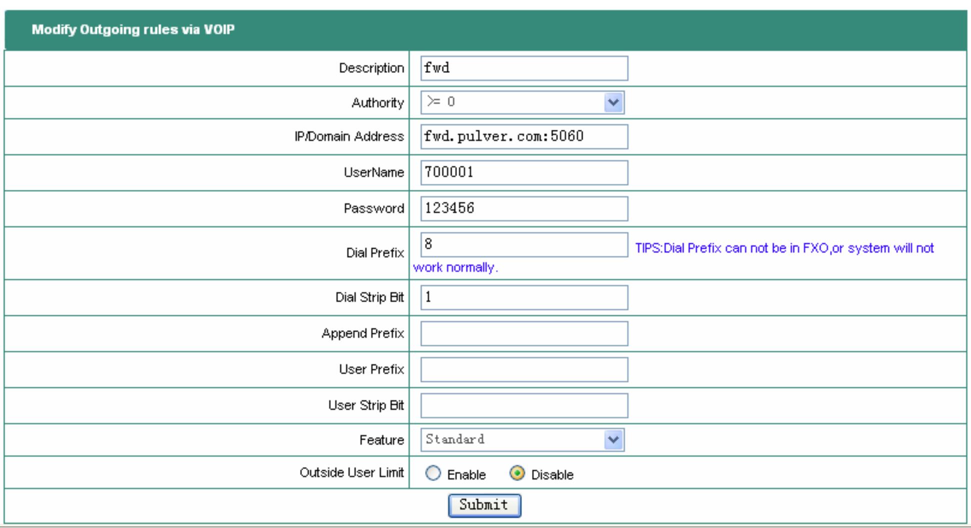
Dial number with prefix 9 would use this rule, 1 digit would be stripped and 00852 would be appended. For example, a user of authority 5 dial 929110000, 9 will be stripped and 00852 would be appended; the actual number dialed out would be 0085229110000. If there is more than one FXO port grouped into group 2. IP PBX will automatically choose the one available.

5.5.2 VoIP Outgoing Call (Outgoing SIP Trunk)

Our IP PBX is based on standard protocol SIP, so we can adding third-party SIP account for outgoing call. We can also make outgoing landline call through the third-party's platform.

There are two ways for VoIP outgoing calls

1. Adding an account with the username and password of the third-party's SIP platform
2. Let the third-party SIP server make our IP PBX as a Trust Host, then no username and password is required for authentication.



- Dial Prefix – Number matching this Prefix would go through this Outgoing Rule. (e.g. 9)
- Dial Strip Bit – The Number of Digits to remove from the dialed number. (e.g. 1)
- Append Prefix – The Prefix to added to the beginning if the rule is used. (e.g. 00852)

For example, when User dials 929001100, and there is a Rule of Dial Prefix is "9", Dial Strip Bit is "1" and Append Prefix is "00852". The actual dialed number would be "0085229001100"

- User Prefix – This is effective only when No Authentication for the Outgoing SIP Trunk. User Prefix is the Prefix to added to the Caller number
- User Strip Number - This is effective only when No Authentication for the Outgoing SIP Trunk. User Strip Number is the number to remove from the Caller Number

For Example, when a Extension of "274" makes a Call, and it matches with a Rule of User Prefix is "290011" and

User Strip Number is "1". One digit would be removed from "274" and "290011" would be added. So, the Caller Number sending to the SIP Outgoing Trunk should be "29001174"

5.6 Incoming Call

Similar to Outgoing Call, we provide incoming call through VoIP and Incoming call through FXO.

5.6.1 Calls from FXO (Incoming PSTN Trunk)

You can configure how the system should handle the incoming calls from each FXO port. The incoming call either be routed to switchboard, calling queue, certain extension or to a conference room.

Calls from FXO	
FXO 1(FXO1)Redirect to：	2001
FXO 2(FXO2)Redirect to：	Switchboard
FXO 3(FXO3)Redirect to：	Switchboard
FXO 4(FXO4)Redirect to：	Switchboard
Update	

5.6.2 Calls from VoIP

Similar to Outgoing to VoIP, we can allow third-party SIP call to call in our IP PBX. The incoming VoIP call can be routed to any number, including the switchboard, certain extension or calling queue.

5.6.3 Add Access Number

For example, If you would like make number "700" as the Voicemail Checking, or mapping the number "3001" as extension "2003", You can use this function to do the mappings.

Add Access Number	
Access Number	700
Redirect into	My Own Voicemail
Memo	
Submit	

Current Access Number List

NO.	Access Number	Redirect into	Memo	Operation
1	3001	2003		

Access Number: The number user would call

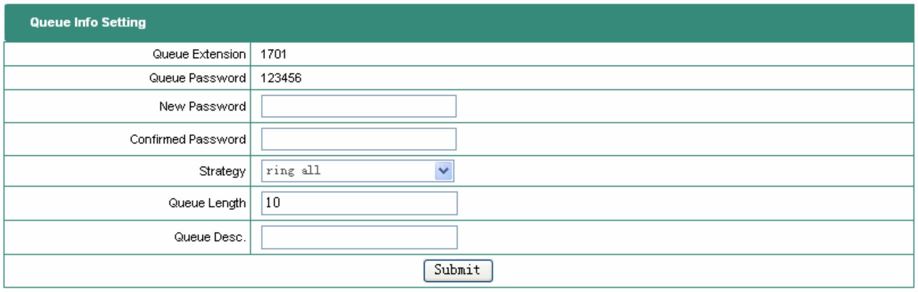
Redirect Into: The number the IP PBX would redirect to. It can also be any arbitrary numbers.

Memo: Anything that help use to comment on this number. Can be left blink

5.7 Advanced Setting

5.7.1 Queue Settings

Operator and PBX can put the incoming call into several Queues. Sit (Call handler of a Queue) of a Queue would ring to notify them to handle the call. If no Sit is available to handle, the incoming call would be hold till there is available handler. An extension can become a Sit of the queue using IVR (default Hotline: 1600).



Queue Info Setting	
Queue Extension	1701
Queue Password	123456
New Password	
Confirmed Password	
Strategy	ring all
Queue Length	10
Queue Desc.	
Submit	

Strategy

Rotation: Each sit would ring in rotation

Random: The system would random find a set to pick up the call

Fewest Frequency: The sit picking up the least number of call would ring

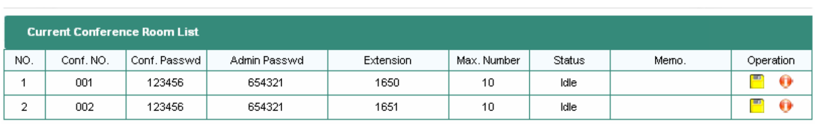
Least Time: The sit handling call least number would ring

Ring all: All sits in the queue would ring, the one picking up first would handle the call

5.7.2 Conference Room

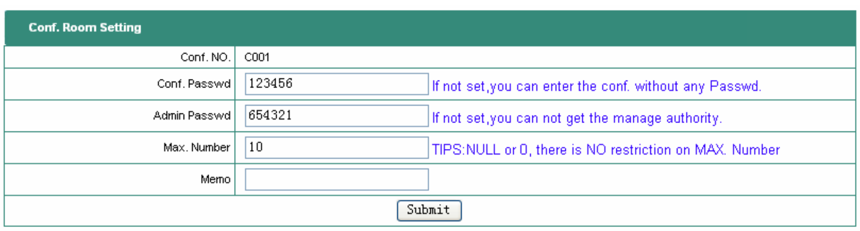
Our system provide two IP PBX conference Room

- Conference Room Password: To authorize user trying to join the conference.
- Max Number: Maximum number of user allowed to join the conference



Current Conference Room List

NO.	Conf. NO.	Conf. Passwd	Admin Passwd	Extension	Max. Number	Status	Memo.	Operation
1	001	123456	654321	1650	10	Idle		
2	002	123456	654321	1651	10	Idle		



Conf. Room Setting		
Conf. NO.	C001	
Conf. Passwd	123456	If not set,you can enter the conf. without any Passwd.
Admin Passwd	654321	If not set,you can not get the manage authority.
Max. Number	10	TIPS:NULL or 0, there is NO restriction on MAX. Number
Memo		
Submit		

Under Operation Page, administrator can:

1. Allow a user to listen and speak
2. Put a user as listen-only
3. Kick a user out to Conference Room

Using IVR to control the Conference Room

After joining a Conference Room, a user can press '*' to get IVR prompt.

Normal user:

1. Press '1' to mute (All other user in the conference cannot listen what you then say). Press '1' again to unmute
2. Press '2' to obtain the administrator authority. IVR will prompt user to enter the Conference Room Administrator password

Administrator:

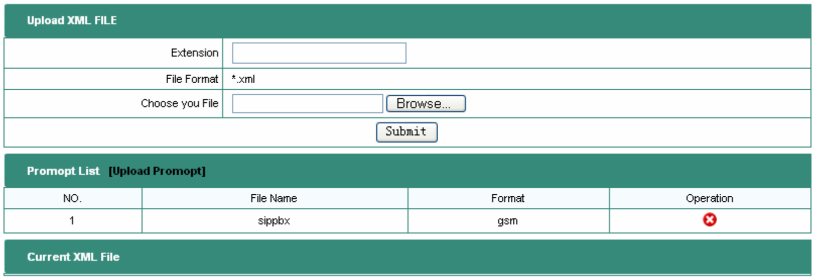
1. Press '1' to mute like normal user
2. Press '2' to lock the current status of Conference Room, so no new user would be allowed to join the conference room.
3. Press '3' to kick a user out the conference room
4. Press '4' to mute all users of the conference room

5. Press '5' to unmute all users of the conference room
6. Press '6' to invite new user to the conference room. You can either invite an extension of the IP PBX or invite an external phone through PSTN.

Note: The IP PBX only responsible to invite, but it wouldn't report whether the invitation is succeeded or not. If the user does not pick up the invitation call, the PBX would call the user again. IP PBX would given invitation after 3 time trial.

5.7.3 Upload IVR Flow

User can define their own flow of IVR through XML. Please refer the Appendix for the details



5.7.4 P2T List (web call)

5.7.5 Network Parameters

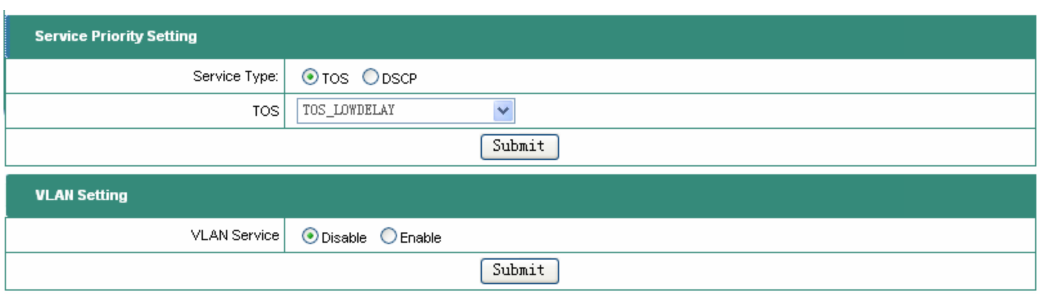
Our IP PBX supports two modes of service priority setting: TOS (Type of Service) and DSCP (Differentiated Service Code Point)

Service Priority Setting

Service Type: TOS

- TOS_LOWDELAY
- TOS_THROUGHPUT
- TOS_RELIABILITY
- TOS_MINCOST

TOS_LOWDELAY is recommended value for the TOS, many routers (including any linux router routers with 2.4 kernels) will give priority to these packets, improving voice quality



Service Type: DSCP

- DSCP Value to be used to mark the voice packet

VLAN Service

You may separate your PC network from your VoIP network using VLAN; this can prevent PC network data from affecting the voice quality of the Voice Network.

Our IP PBX support VLAN, and at the same, you would need a managed switch supporting VLAN as well. Accepted practice is to make a Voice VLAN and Data VLAN. On the IP PBX, enable the VLAN Service, and enter the VLAN ID for the Voice VLAN.

5.7.6 Update System Prompt

System Prompt is those voice prompts user heard during IVR, like Voicemail or Conference Call. These prompt are in English by default, but Update to different language or voice is possible through the web interface.

Procedure to Update to English Prompts

1. "Advanced Setting"->"Update System Prompt"
2. Select Prompt (Obtained from the Vendor) and Submit
3. Wait the submit and transfer completes
4. Reboot the system to take effect.

5.8 Call Record

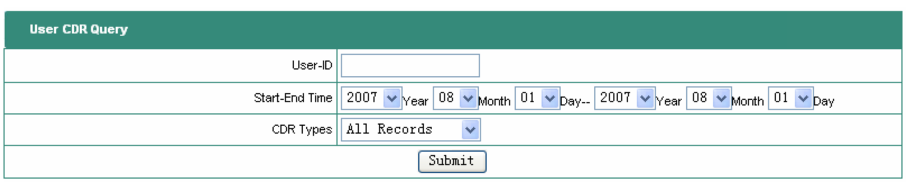
Our system would keep call record within 2 months. You can analyze these call history. If the history would like to be kept, please export the Call Record every 2 months

5.8.1 System Call Records

You can list out the call records of the whole system.

5.8.2 User Call Record

You can list out the call records of a user within particular period.



5.9 IP Voice Broadcast

People in Office (especially Reception) will find it convenient if there is a broadcast system to make an annoucement, notify some colleague for a meeting, or looking for a not-in-seat colleague. Our IP PBX offers a broadcast system when it is used with our IP Phone (VOB820, VOB821, VOB822 or VOB823).

To use the broadcast

1. Use any Phone registered to the IP PBX, press 1800 (using an IP phone). When the call is connected, you can speak to the phone and make the broadcast.
2. All IP Phones that support IP Broadcast in Group 0 would change to Speaker mode automatically; otherwise the IP Phone would ring instead. Voice would be broadcast through those IP Phones' speaker phone.
3. All other non-ACETECH in Group 0 would rings instead of changing to Speaker mode automatically. We suggest you putting only ACETECH IP Phone in Group 0 for deployment of this feature.

Q&A

Q: Number of FXO Ports is not enough for company's use

A: You can add a SIP VoIP FXO Gateway, and connect it together with the IP PBX. Configure the FXO Gateway to forward all incoming to the switchboard.

Q: No response on incoming call from external FXO Gateway

A: Please make sure the external FXO gateway is put to Trust host List (System >> Trust Host)

Q: How to record the Greeting/Welcome message ?

A: Please follow the below steps

1. Press "1605", and you should hear voice-prompt
2. Press "0" to record Welcome Message, and you should hear the current Welcome Message.
3. Start recording after you hear a Beep sound, and press '#' key when you finished.
4. Then, Press '1' to review your recorded message, Press '2' to Record again, Press '3' to Save.

Q: How to configure personal Voice Prompt?

A: Please follow the below steps

1. Press 1604 into voice-prompt
2. Enter username and password
3. Press '0' to enter voice recording mode
4. Press '1' to record normal voice-prompt
5. Press '2' to record voice-prompt on busy

Note: Make sure the Call Waiting is disabled in order to test the voice-prompt on busy

Q: Voicemail Left but User cannot find it

A: Please make sure the extension has the voicemail option enabled, and set the Voicemail "saved on server"

1. Login as Administrator
2. Select "Users"->"User List"
3. Click the "Functional Setting" of the User you would like configure
4. Make sure the "Voice-Mail Service" is "Enable" and Approach is "Send msg to mailbox, and saved on server"
5. Click Submit if any change is made

Q: Voicemail Left but No email notification is sent

A: Please make sure your email account is properly set and the SMTP setting is set properly under "System"->"Voicemail Setting"

The IPPBX need to connect to a outgoing mail server (SMTP) to send mail.

Set the SMTP Setting

1. Login as Administrator
2. Select "System"->"Voicemail Setting", and set the followings

Mailbox - Mailbox Name (e.g. myippbx@netvigator.com)

User - User Name of the Mailbox if authentication is required (e.g. myippbx@netvigator.com)

Password - Password of the Mailbox if authentication is required

SMTP Server - SMTP Server name (e.g. smtp.netvigator.com)

3. Click “submit” and Reboot the system

Set the Personal Email Account (per user)

1. Login as Administrator
2. Select "Users"->"User List"
3. Click the "Functional Setting" of the User you would like configure
4. Make sure the "Voice-Mail Service" is "Enable" and "Email Address" is set to mailbox (e.g . yourname@gmail.com")
5. Click Submit if any change is made

Q: How to clear the Call Record History

A: Goto “CDR”->”CDR Export”. Select the Month you would like to clear the Call Record History, and click “Submit”. You can then Click delete (The icon with a cross sign) to clear the record.

Q: The FXO port of IP PBX will keep busy even if the call is disconnected.

A: Probably there is some problem on Busy Tone detection. Please follow the below (Only support after Version 080206)s

1. Call the IP PBX from the FXO port
2. Under the Auto-Answer of the IP PBX, press 1850, and you should hear three beep sounds

Wait till the LED light of the FXO stop blinking or after 1 minute.

Hotline Table

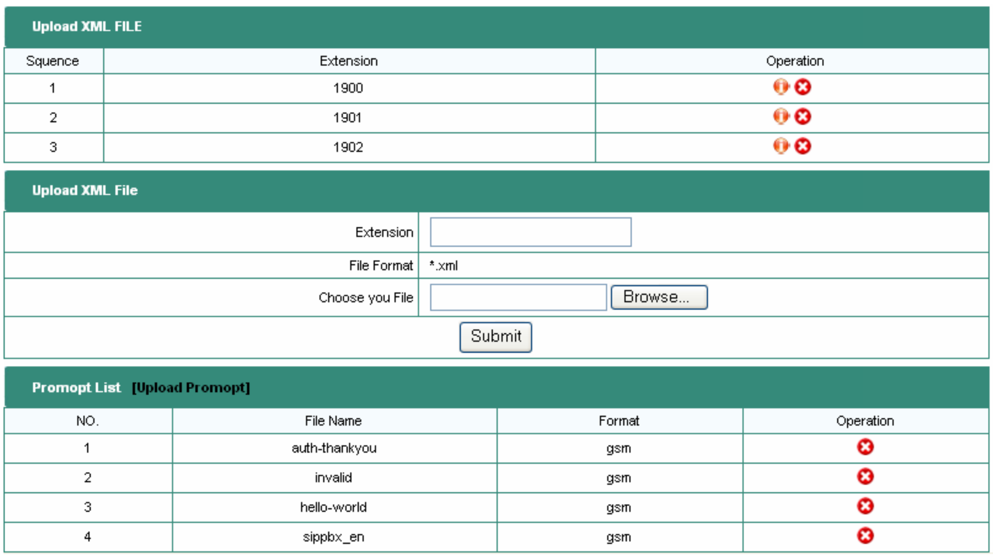
Hotline	Function	Details
***8**	Pickup Phone for the same group	
Transfer		During conversation, press transfer key on IP Phone, and press the extension to forward
112	Switchboard	Switchboard IVR
117	CID Reader	Show the Caller ID of caller
1600	Queue Manage Hotline	
1601	Operator Setting Hotline	
1602	Functions Setting Hotline	Configure Call Transfer, Call Forward, Number Binding, etcs
1603	My Own VoiceMail	Listen and manage your own Voicemail Account
1604	Voicemail Hotline	Listen and manage anyone Voicemail Account (Need to enter UserID and the Password)
1605	Record Prompt	Record Welcome Message, Unavailable Message, Busy Message, etc
1650	Conference Room1	Default Password: 123456
1651	Conference Room2	Default Password: 123456
1701	Queue1	
1702	Queue2	

1703	Queue3	
1704	Queue4	
1800	IP Broadcast	Broadcast to all phones in the Group 1
1850	Auto Busy Detect	Calibrate the Busy Tone Detection

Appendix A Customize IVR Using XML

IP PBX User can use a XML file to describe the IVR flow to be used in the IP PBX. Three steps can customize the IVR Flow – Edit the XML, Record the Voice Prompts, and Upload the XML

1. Create a XML File
   XML is used to describe the flow of the IVR. You can use a simple text editor like Notepad to edit a XML file.
2. Upload the Prompt Files required by the IVR. The Prompt should be load under "Switchboard"->"Prompts" page.
3. Upload the XML File, and binding the IVR Flow described by the XML with an Extension. So, user call that extension would enter the IVR Flow described by the XML File.



How to Edit the IVR XML File ?

The best way to learn is learnt by example. Here are some examples for the IVR XML File, from simple to more complicated one.

1. **Play a prompt file and wait for a single key input.**

   <Prompt File="sippbx" Timeout="40" Repeat="3" String="0"> </Prompt>

   The above XML File would make IP PBX to play to prompt file "sippbx.gsm", and wait 40 seconds for user input (Timeout="40"); If there is no user input, the IP PBX would repeat the prompt for at most 3 times (Repeat="3"). The IP PBX would stop playing the prompt if user press a key.
   Note that no ".gsm" extension is needed on specifying the prompt file.
2. **Play a prompt file and wait for more than one key**

   <Prompt File="sippbx" Timeout="40" Repeat="3" String="1"> </Prompt>

   The only difference between this IVR XML File and Example a is that, String="1" for this File. That means IP PBX is now expecting for a set of keys instead of one. User is expected to press '#' key to let

PBX know the input is completed

3. **Playing different Prompts based on Input Key**

```
<Prompt File="sippbx" Timeout="40" Repeat="3" String="0">
    <Case DTMF="1">
        <Prompt File="en" Timeout="40" Repeat="3" String="0"></Prompt>
    </Case>
    <Case DTMF="2">
        <Prompt File="cn" Timeout="40" Repeat="3" String="0"></Prompt>
    </Case>
</Prompt>
```

The IP PBX would play the "sippbx.gsm" Prompt file, and wait for a single key. If '1' is pressed, The IP PBX would play the Prompt file "en.gsm". If '2' is pressed, the IP PBX would play the Prompt file "cn.gsm". If no input is pressed, IP PBX would close the call after 40 second Timeout.

4. **Call Forward**

```
<Goto Exten="1701" />
```

The IP PBX would Forward the call to Extension 1701 (Queue).

5. **Interactive IVR**

First of all, we have need to introduce three new commands - Replay, GoBack and End Syntax

Replay – Replay the previous prompt

GoBack – Go Back to Upper Menu

End – End the Call

```
<Prompt File="sippbx" Timeout="20" Repeat="2" String="0">
<Case DTMF="2">
          <Prompt File="invalid" Timeout="20" Repeat="2" String="0">
           </Prompt>
</Case>
<Case DTMF="1">
        <Prompt File="hello" Timeout="20" Repeat="2" String="0">
        <Case DTMF="3">
                <Prompt File="thankyou" Timeout="0" Repeat="0" String="0"/>
                <End/>
        </Case>
       <Case DTMF="4">
                <Goto Exten="117"/>
       </Case>
        <Case DTMF="5">
                <GoBack/>
       </Case>
       </Prompt>
</Case>
<Case DTMF="*">
        <Replay/>
</Case>
</Prompt>
```

The IP PBX would play prompt file "sippbx.gsm", and wait for an input.

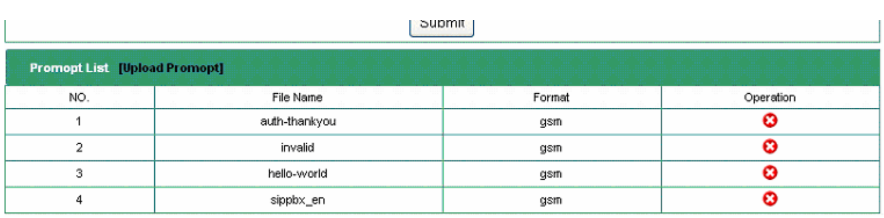
If User pressed '2', the IPPBX would play the prompt file "invalid.gsm".

If User pressed '1' the IPPBX would play the prompt file "helloworld.gsm", and wait for another key.

If User then pressed '3', the IPPBX would play the prompt "thankyou.gsm", and End the call; If User pressed '4', the IPPBX would forward the call to Extension "117"; If User pressed '5', the IPPBX would go back to Upper menu.

If User pressed '*', the IPPBX would replay the previous prompt.

Note: Please make sure all the prompt files used by the IVR has been uploaded to the IP PBX. User can check in the "Upload XML File" page, just like below:



Submit

Promopt List [Upload Promopt]

NO.	File Name	Format	Operation
1	auth-thankyou	gsm	✖
2	invalid	gsm	✖
3	hello-world	gsm	✖
4	sippbx_en	gsm	✖