VOI-9200
SIP IP PBX

User Manual

Ver. 1.02 - 0809

Table of Contents

1. Introduction

1.1. Overview

The VOI-9200 IP PBX is an embedded Voice over IP (VoIP) Server with Session Initiation Protocol (SIP) to provide global virtual office IP extension phone connection for small-to-medium business (SMB) companys. Equipped with two FXS ports, two FXO ports, Ethernet LAN and WAN ports, VOI-9200 combines the telephony network and the data network into a manageable converged network.
VOI-9200 IP PBX works with various IP phones (Desktop, WiFi, Bluetooth, and DECT), VoIP gateways, and analog telephone adapters (ATA) to route calls among client phones, analog phones, and PSTN network. Call features such as conferencing, auto attendant, and voicemail can be seamlessly enabled to all phone devices. In addition, it also provides Internet access to all LAN devices through NAT router.
VOI-9200 IP PBX provides call control and media relay services to SIP clients, and it performs the following primary functions:

- SIP Registrar
- SIP Outbound Proxy with media relay
- SIP Gateways (FXO)
- SIP PBX for extension calls
- Auto attendant Interactive Voice Response (IVR)
- Voice Mail IVR
- Meet-Me Conferencing

VOI-9200 IP PBX has a built-in suite of voice applications for supplementary services. This lowers down the total cost of a converged network enabled by VOI-9200 IP PBX than building separated infrastructures for legacy telephony network and data network. In addition, with a web-browsable interface to the data network configuration and voice service provisioning, VOI-9200 brings the manageability of both networks together to facilitate administration locally or remotely.

1.2. Feature

Supported Standards
RFC 3261, RFC 3311, RFC 3515
RFC 3265, RFC 3892, RFC 3361
RFC 3842, RFC 3389, RFC 3489
RFC 3428, RFC 2327, RFC 2833
RFC 2976, RFC 3263, RFC 3264

SIP Registrar
- Static/Dynamic registration
- Configurable expiry time
- MD5 authentication
- Registration proxy to external registrars
- Configurable PBX Caller ID
- User profile
- Handle loose RFC-compliant phones
- Resilient message retry mechanism
- Seeding historical registrations

SIP Proxy
- Proxy server
- Call-based MD5 authentication
- NAT traversal for clients
- Outbound proxy with or without WAN
- Inter-proxy call hand-off

PBX Features
- Support call hold, call waiting, 3-way call conference with feature phones
- Built-in in-line call transfer
- Unconditional, unavailable, busy call forward
- Per-calling-number forward and rejection
- Group-based call pick-up
- Call-parking
- Multi-room meet-me conference
- Auto-attendant
- Voice mail system
- Call privilege grouping
- FXO interface for PSTN Inbound/outbound
- FXO disconnection tone detection
- FXO hunt group
- Caller ID detection
- Echo cancellation
- In-band/RFC2833/SIP-INFO DTMF translation
- Support 5 SIP trunk
- Intra-PBX stackable trunking over Ethernet
- FWD/Vonage account sharing for extensions
- Interoperable with Cisco CallManager, CCME, IOS SIP gateway, Unity CUE, 79XX, ATA
- Call admission control for wired/wireless phones
- Music on hold
- Direct line
- Outbound 900/0204 blocking

Auto Attendant
- Configurable Greeting
- Key to reach operator
- Timeout interval and timeout action
- Music on ringing extensions
- Forward to voice mail on no-answer

Voice Mail
- User PIN
- Multilingual
- Multi-folder archive
- Fast-forward/Rewind/Undelete
- MWI notification
- E-mail notification and attachment(Unified messaging)
- Personal reception on unavailability and busy
- Voicemail forwarding
- Reply call or new call in voicemail menu

Storage
- Support USB 2.0 storage media

Meet-me Conference
- Multiple rooms with configurable number and PIN
- Music on first dial-in party
- Hot key to leave conference

NAT
- Auto NAT discovery and traversal
- Built-in STUN client
- RTP proxy
- RTP port range designation

Relational Provision
- Logical partition/relation on users and trunks
- Logical provision on outgoing and incoming calling search scopes
- Rich dial-plan expressiveness through route patterns
- Object-oriented provisioning paradigm

Administration
- Web-based configuration
- Flat system event syslog
- Flat Call Detail Record (CDR)
- Extension status display
- TFTP server and TFTP repository maintenance
- Network Time Protocol time synchronization
- Real Time Clock setting
- DHCP server with multiple partitions, per-MAC IP binding, list of options
- Configurable time zone
- Firmware upgrade through Web interface

Network Management
- DHCP/PPPoE/Static IP on WAN
- LAN IP and netmask specification
- Firewalling on predefined services
- NAT for outbound traffic from LAN
- DNS and dynamic DNS
- QoS queuing mechanism for VoIP and data traffic

Maximum Capacity
- 30 extension registrations
- 30 voicemail accounts
- 10 concurrent calls with RTP
- 4 analog calls

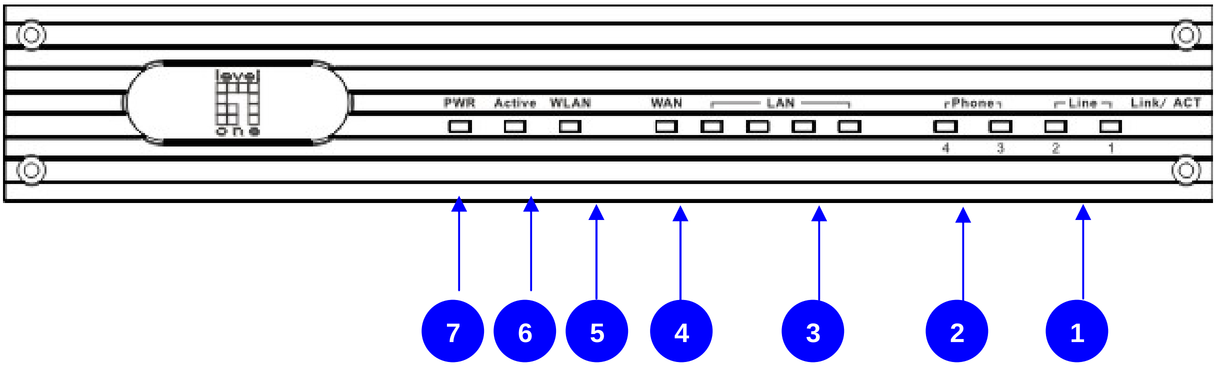
Voice
- Caller ID : Bellcore and ESTI
- Fax support: T.38 compliant
- Echo canceller: G.168-2002 compliant
- Seamless integrated with VOI-7000 / VOI-7100, VOI-800x

1.3. Package Content

- VOI-9200
- Power Adapter 12V, 1.66A
- Cat.5 Cable
- CD Manual

1.4. Physical

1.4.1. Front Panel

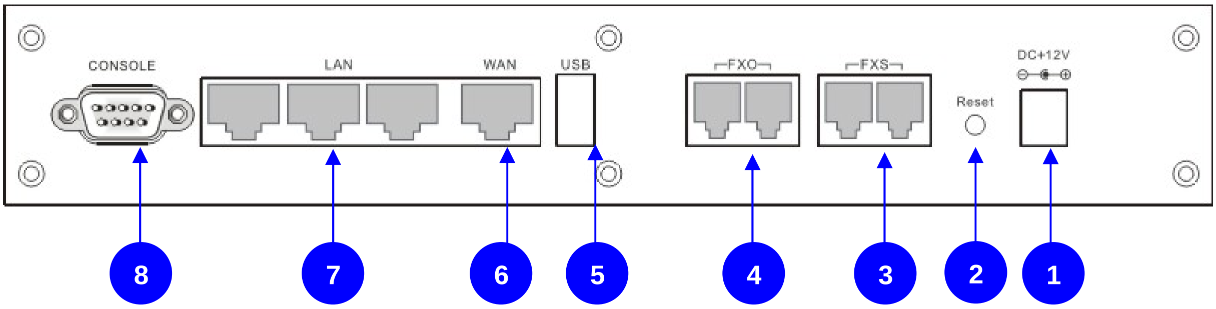


LED

1. Line	2-Port FXO
2. Phone	2-Port FXS
3. LAN	ON indicates LAN connection BLINK indicates LAN activity
4. WAN	ON indicates WAN connection BLINK indicates WAN activity
5. WLAN	Reserved *
6. Active	On Indicates the system is ready
7. PWR	ON indicates the unit is powered up

* WLAN is not applicable for the VOI-9200

1.4.2. Rear Panel

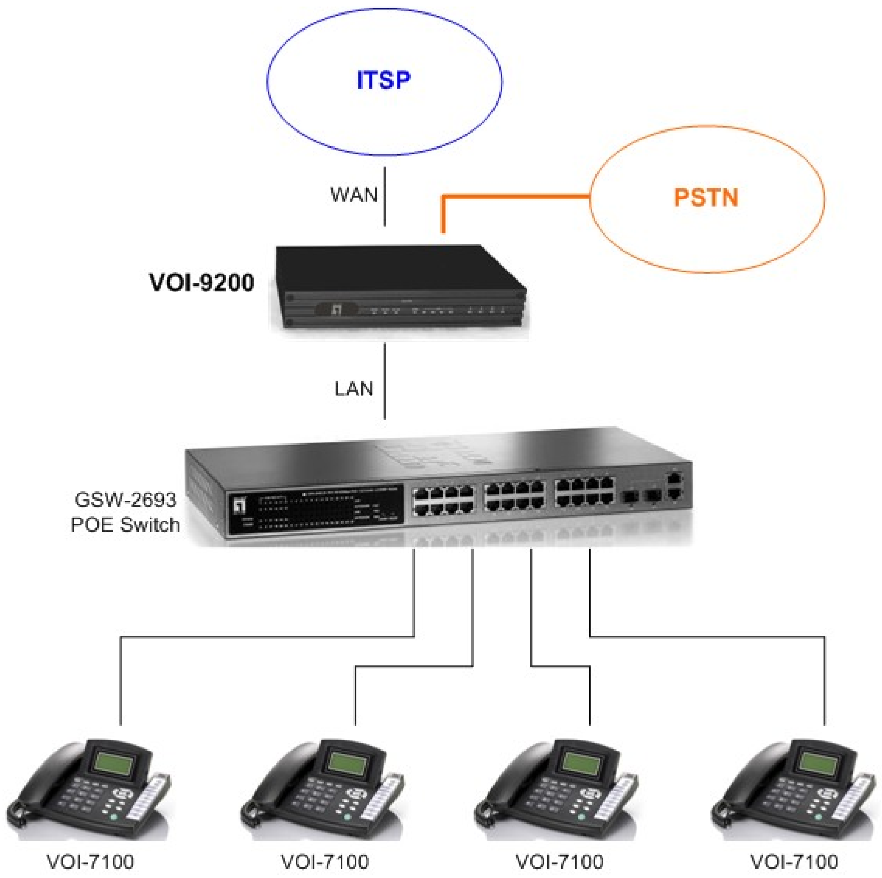


1. Power	12V DC, 1.66A
2. Reset	Hold the Reset button and release to reboot system Hold the Reset button for 10 seconds before release to restore whole system back to the factory default
3. FXO ports	2 FXO ports are for connection to PSTN lines, and numbered 1 and 2 from left to right.
4. FXS ports	2 FXS ports are for connection to the analog phone, and numbered 3 and 4 from left to right.
5. USB port	Connect to an external USB drive for backup internal system storage. Click the Backup icon in Web configurations and follow instructions to insert the USB connector of an external USB drive.
6. WAN port	Connect to a broadband ADSL/Cable modem or a WAN router.
7. LAN ports	4 LAN ports are for connection to PC or Laptop, extended IP Phones, or VoIP Gateways/ATA, etc.
8. Console port	RS-232 Serial connection for configuration by CLI

1.5. Default Settings

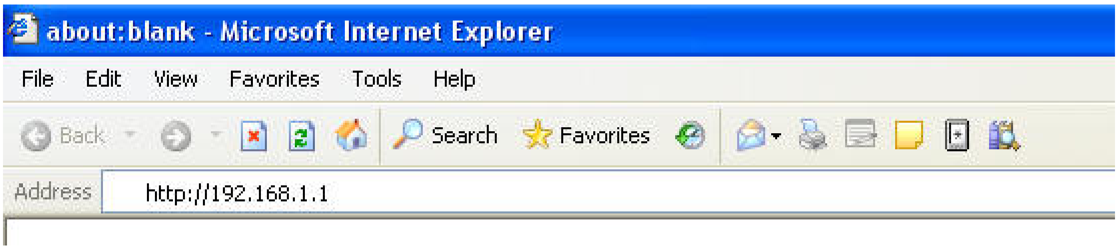
WAN IP	**192.168.0.1**
LAN IP	**192.168.1.1**
Username	**admin**
Password	**admin**

1.6. Connection Diagram



2. Web Based Management

This chapter illustrates how to login and configure system parameters with VOI-9200. The factory default of LAN IP address is 192.168.1.1 and WAN IP address is 192.168.0.1. You can connect your PC or laptop to access the web GUI through LAN port at https://192.168.1.1/, or through WAN port at https://192.168.0.1.



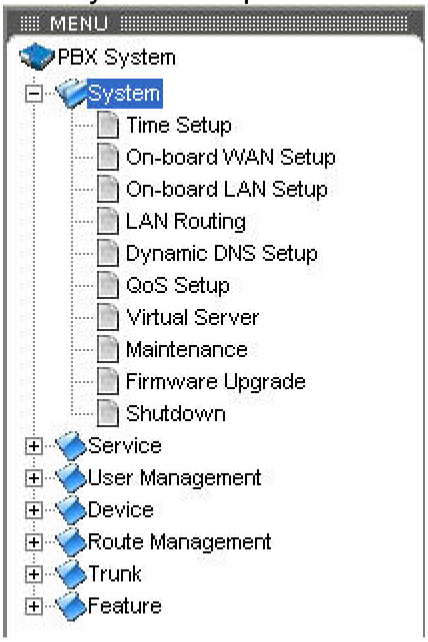
Once connected, the browser may warn about accepting a certificate, please accept it.

Then, enter the username and password (default is **admin**/**admin**) to login for PBX configuration. The PBX System with PBX status will be shown.

Note the password could be changed in the User Management page under user ID “admin”.

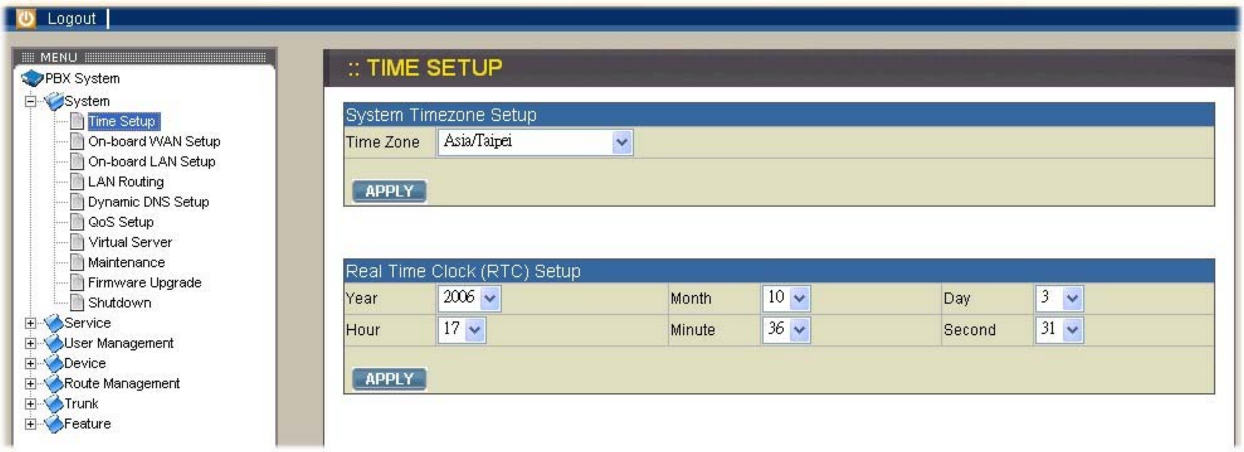
2.1. System

The System setup includes the following configurations



2.1.1. Time Setup

Select System→Time Setup, and you can see the current setting of time zone and real time clock. The Time Setup page allows to configure time zone and date for IP PBX.



System time zone setup

Choose the time zone for the IP PBX0, then click APPLY button to save the setting.

Real time clock (RTC) setup

Choose values of year, month, day, hour, minute, and second respectively, then click APPLY button to save the settings.

2.1.2. On-board WAN Setup

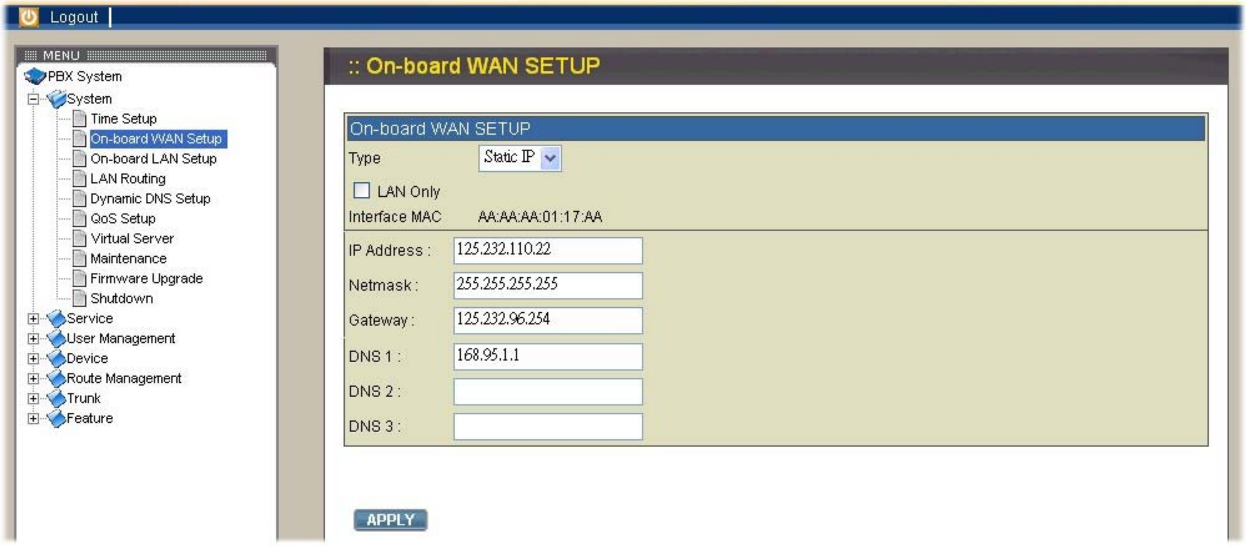
The WAN setup page allows administrator to configure WAN interface of VOI-9200. Select System→On-board WAN Setup, to display current setting of WAN interface. You can select one of three interface types among static IP, DHCP, and PPPoE. The default type of WAN interface is static IP and the default WAN IP is 192.168.0.1. If LAN only checkbox is checked, the WAN interface will be disabled.

Static IP

You can choose Static IP in Type list and configure the following information:

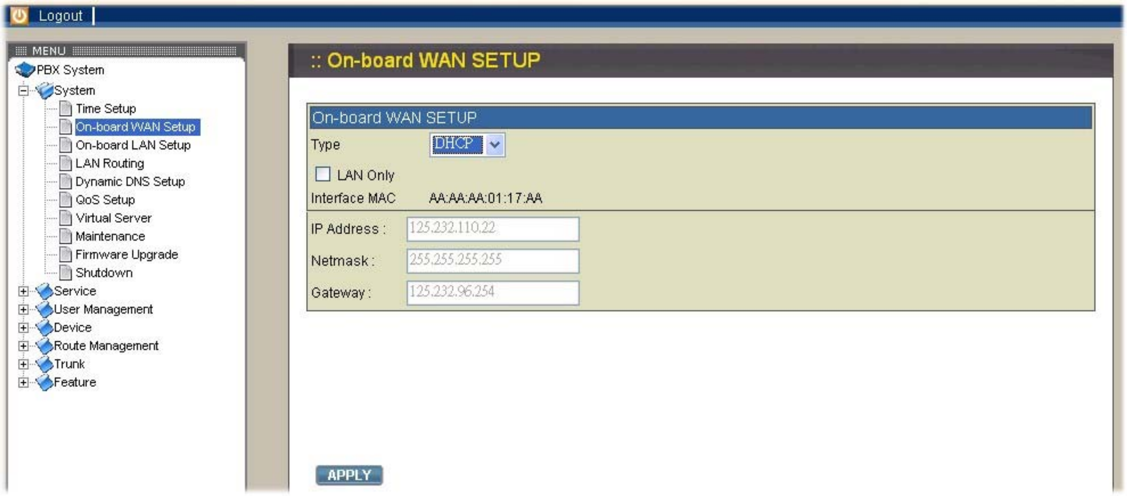
- IP address
- Network mask
- Default gateway IP address
- Primary and secondary DNS servers

Then click APPLY button to submit.



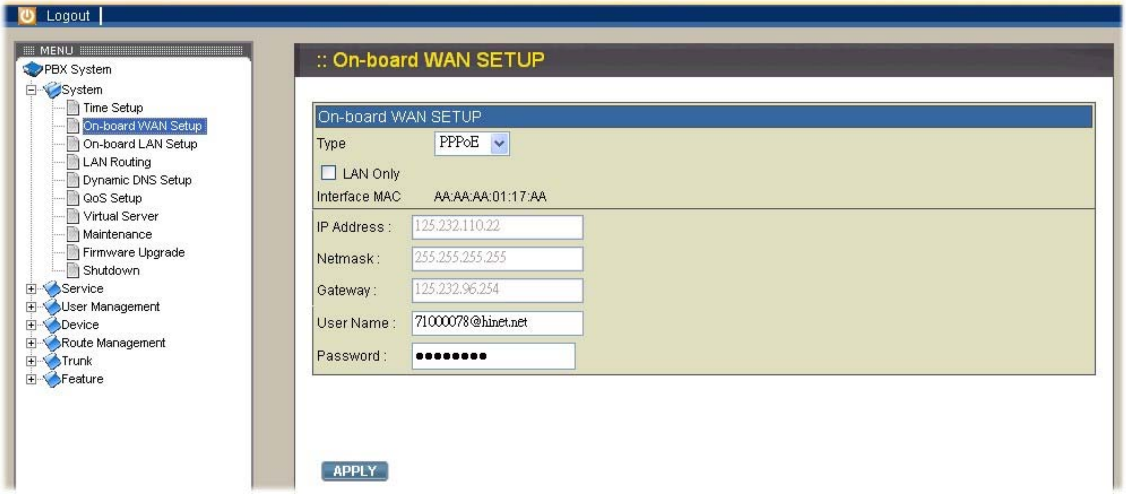
DHCP

You may choose DHCP in Type list and click APPLY button, and the acquired IP address, network mask, and default gateway information will be displayed when you revisit this page later.



PPPoE

Choose PPPoE in Type list and enter the username and password, then click APPLY button to save your input. The PPPoE dialing will start right away. When the connection is active, the page will show the acquired IP address, network mask, and default gateway information. There will also be a Disconnect button to disconnect connection when desired. Please ask your ISP if you aren't sure the Username and Password.

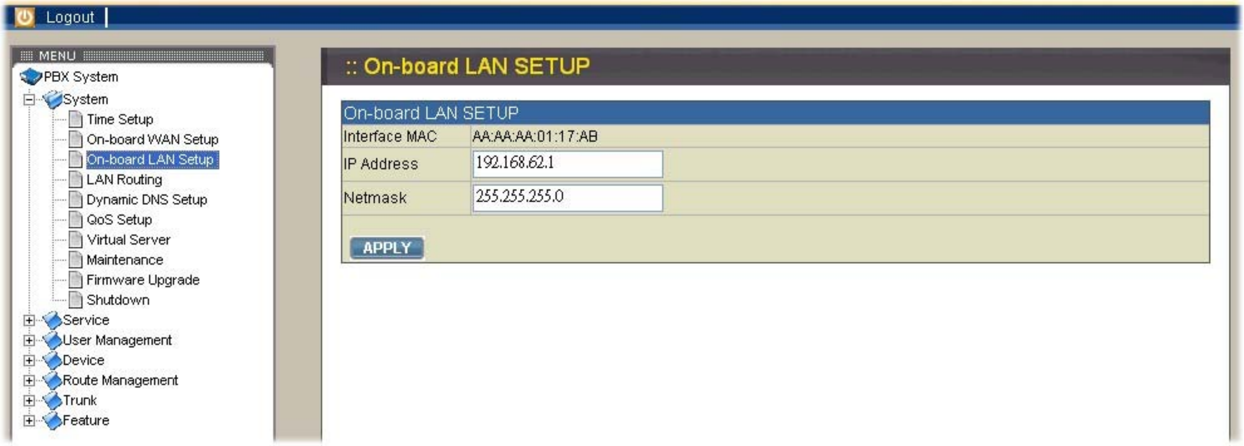


Checkbox: LAN Only

Check checkbox LAN Only to disable WAN IP setting but allow the configuration of default gateway and primary/secondary DNS servers.

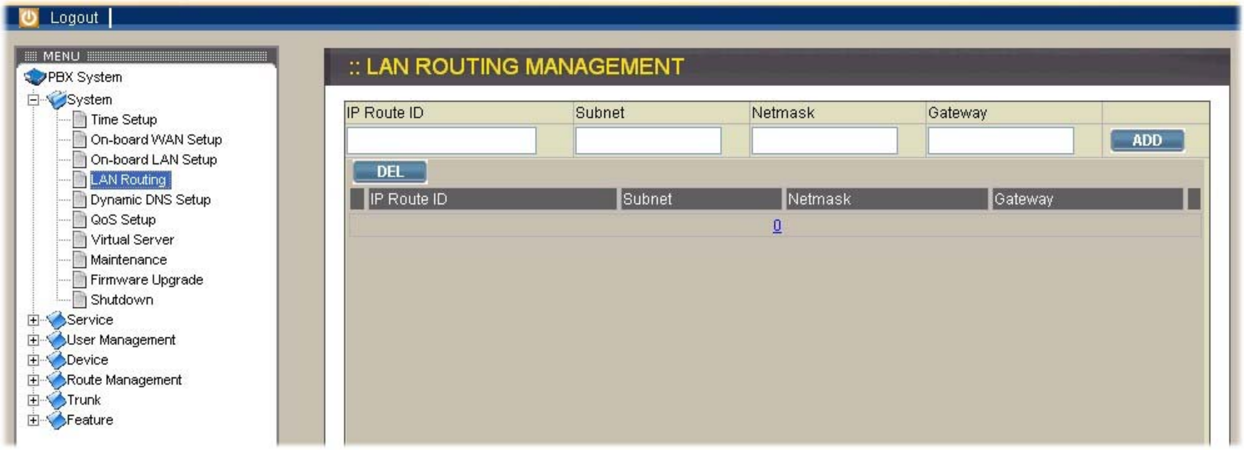
2.1.3. On-board LAN setup

The LAN setup page allows administrator to configure LAN network interface for VOI-9200. Select System→On-board LAN setup to display the current setting of LAN interface. The default LAN IP address is 192.168.1.1. You can enter your desired IP and mask then click APPLY button to save. Note that VOI-9200 IP PBX at default assigns IP addresses for LAN devices via DHCP server and translates those addresses into its WAN IP address for access beyond the LAN subnet. As a result, modifying the system LAN IP subnet must also change DHCP pool and LAN Routing (if any) accordingly. Besides, IP PBX service must be restarted.



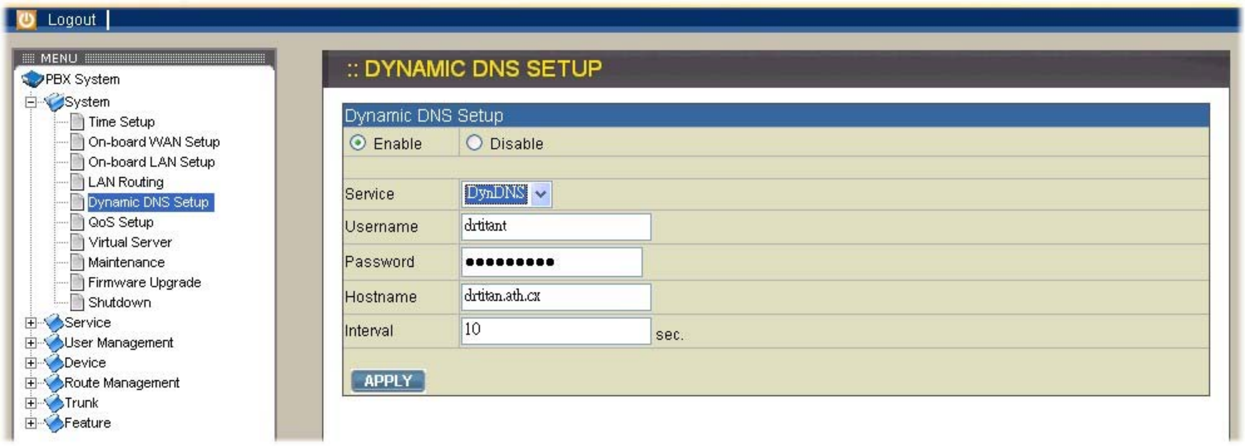
2.1.4. LAN Routing

To enable static routing among LAN subnets, enter the IP address, netmask, and the gateway IP address for the IP PBX. It is important to assure that the given gateway IP address is in the same IP PBX's LAN network. Each subnet requires an entry even multiple subnets share the same gateway, unless masking does the same. The examples are adding IP Route IDs net1 and net2 with parameters 192.168.128.0/255.255.255.0, 192.168.129.0/255.255.255.0, shared gateway 192.168.1.254 respectively. Or, IP Route ID net1n2 with 192.168.128.0/255.255.254.0 and gateway 192.168.1.254 would do the same. Added routes enable routing immediately after clicking APPLY button, however, IP PBX service needs to be restarted to activate calls from designated LAN subnets.



2.1.5. Dynamic DNS Setup

A dynamic WAN IP address causes difficulty for inbound connections from remote clients for IP PBX systems. A popular work-around is to adopt domain names provided by a DynDNS server and run a DynDNS client on or behind the gateway router (or IPPBX). It is required to apply an account first and create a hostname in the account before configuration. Select System→Dynamic DNS Setup, check Enable box, and enter the account infomations and refresh interval to activate a DynDNS client. The client then uses Username and Password to access its account and update the Hostname with the latest WAN IP address at the DynDNS server in Interval seconds periodically.



Enable DynDNS

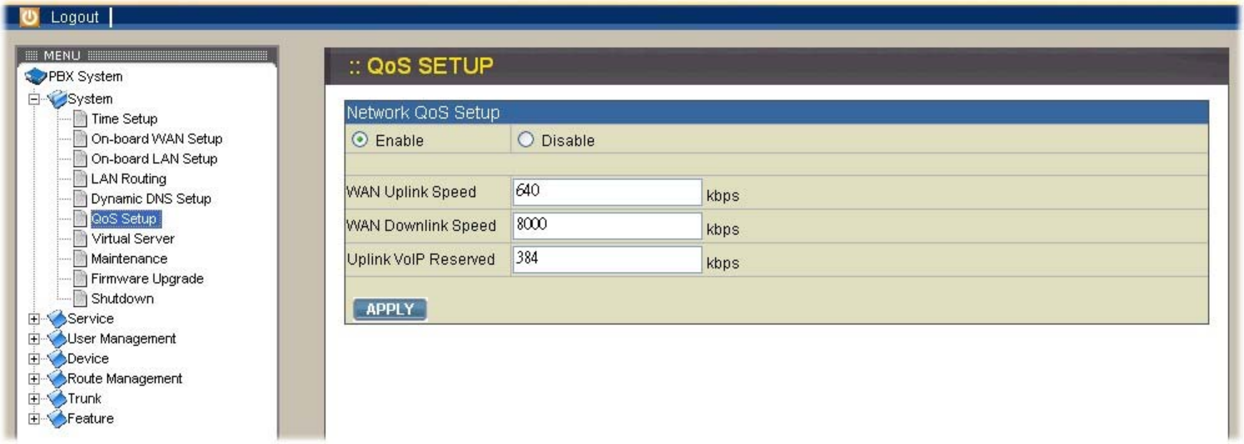
Select Enable button, enter the Username, Password, Hostname, and Interval, and then click APPLY button. Typical hostname has a form of <hostname>.dyndns.org. The refresh interval can be from 60 to 600 seconds depending on the volatility of WAN IP assignment. For example, you can visit http://www.no-ip.com to apply an account with your own username and password and acquire a hostname, like VOI-9200.no-ip.org, named by yourself.

Disable DynDNS

Select Disable button, and then click APPLY button.

2.1.6. QoS Setup

To assure the bandwidth reserved for the outgoing and incoming VoIP traffic overriding regular data traffic, you can select System→QoS Setup to access the QoS Setup page which offers three parameters to characterize the WAN link. These parameters must be correctly given according to the actual WAN transmission speed. By default QoS is disabled.



Enable QoS

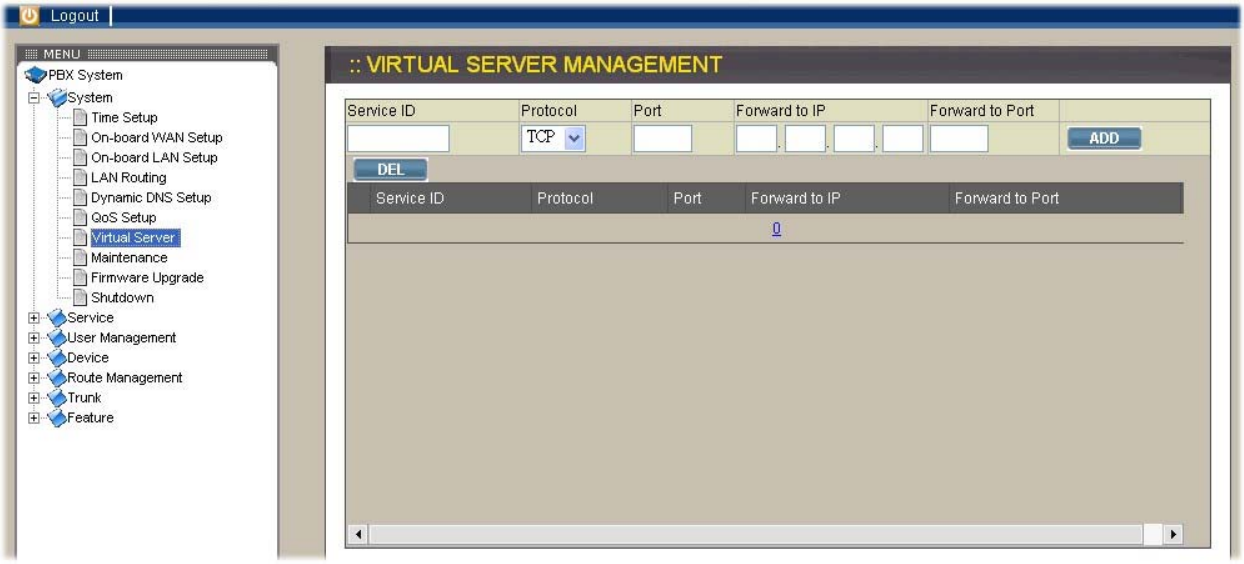
Select Enable button and enter the values of WAN Uplink Speed, WAN Downlink Speed, and Uplink VoIP Reserved (bandwidth) respectively, and then click APPLY button. For example, a 2M/256K ADSL program, the maximum WAN uplink and WAN downlink speed are 256 kbps and 2048 kbps individually. The Uplink VoIP reserved could be, say, 192 out of the total 256 kbps to allow 2 concurrent G.711 calls.

Disable QoS

Select Disable button, and then click APPLY button.

2.1.7. Virtual Server

If you want to access any device behind LAN of VOI-9200 IP PBX from WAN, you need to select System→Virtual Server to configure port mappings. In virtual server page, Service ID names the service; Protocol and Port specify the TCP or UDP port number(s) on WAN IP which will be forwarded to the Forward to port of Forward to IP in LAN. For example, 192.168.1.5 is a mail server to be seen from outside, one should configure TCP port 25 to be forwarded to 192.168.1.5 port 25. In other words, if the WAN IP of VOI-9200 is 192.168.0.1, you can type http://192.168.0.1:25 in your browser to access the mail server of which IP is 192.168.1.5.



2.1.8. Maintenance

Select System→Maintenance to enter the IP PBX maintenance page to get relative records of system operation.

Storage Backup

To backup internal main storage, click the BACKUP icon and follow instructions to insert the USB connector of an external USB drive. There are two checkboxes for removing either CDR or Voicemails after backup. After a confirmation of the insertion, backup starts a few seconds later if the external USB drive is accessible and has enough available capacity. After a successful backup, you can find a new folder created on the external USB drive. Note that whether the backup is successful or not, the external USB drive must be removed.

SIP UA
When you click **SIP UA** tab, it will show a new listing of SIP registration status for each client. All fields shown in the window are explained as below.

Extension/Trunk ID: show the extension or trunk ID which are also used as SIP account for registering at VOI-9200 IPPBX

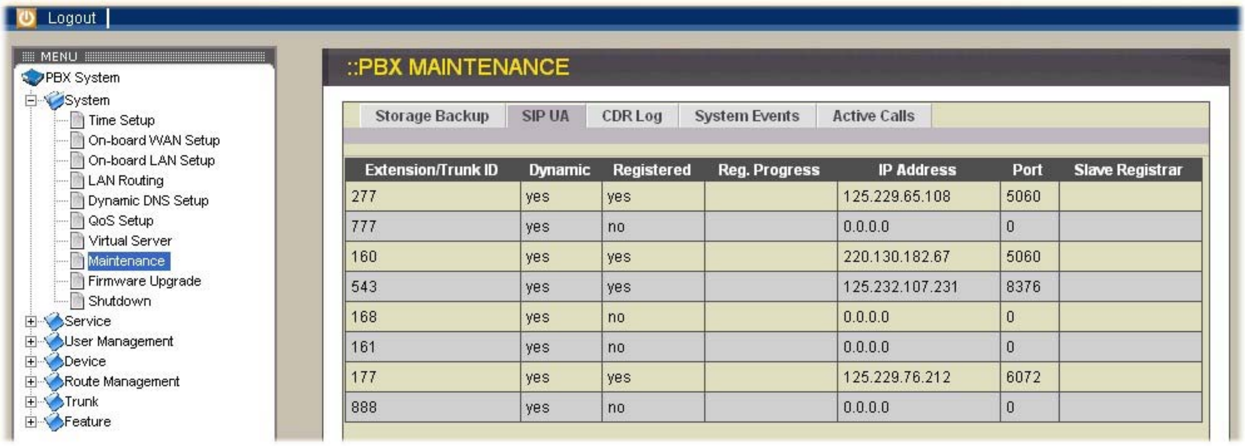
Dynamic: reveal the listed IP address of the corresponding extension/trunk ID is dynamic or specified

Registered: indicate whether the extension/trunk ID registers successfully or not

Reg. Progress: display the response code and message if registration has been attempted but not success so far

IP address: present the IP address of the corresponding extension/trunk ID

Port: designate which port to be used for SIP connection between VOI-9200 IPPBX and the SIP device who registered as the corresponding extension/trunk ID



CDR Log
Click CDR Log icon to show the Call Detail Record (CDR) listing each call record in detail. You can download the complete CDR file by clicking the Get File icon. The following describes all fields in CDR page.

Calling Number: the extension belonging to the caller

Dialed Number: the complete number that the caller dialed practically

Caller ID: the configured caller ID of the caller

Dest. Interface: the interface of VOI-9200 that this call passed through

Start Time: the time when this call was made

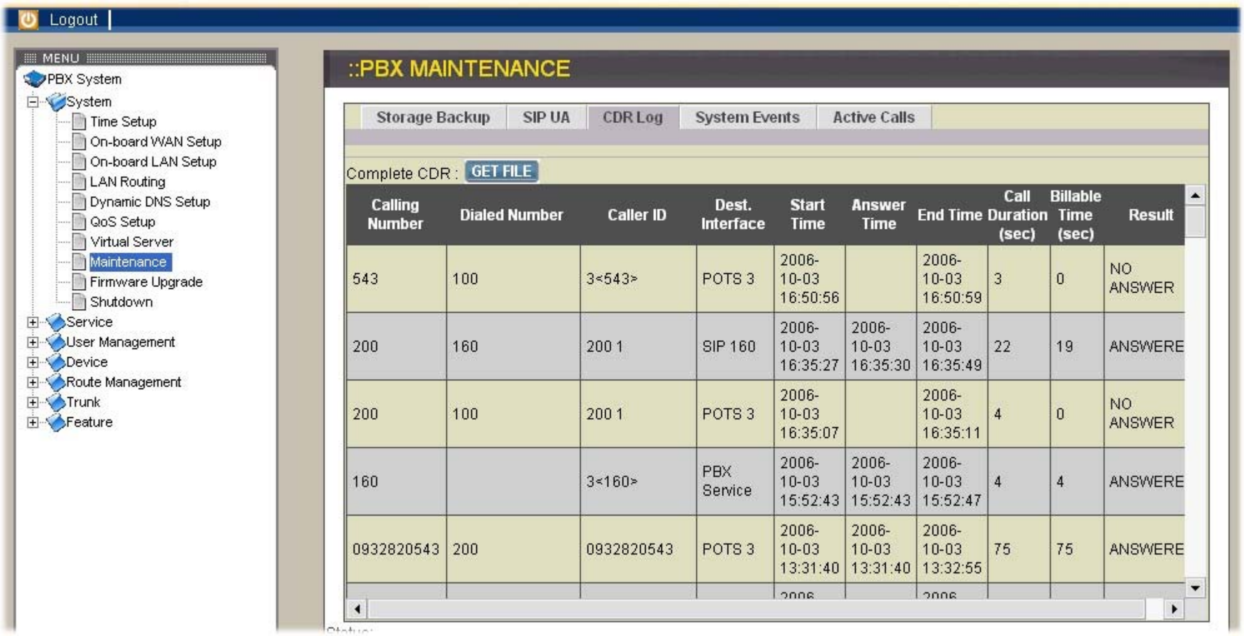
Answer Time: the time when this call was hanged up

End Time: the time when this call was hanged up

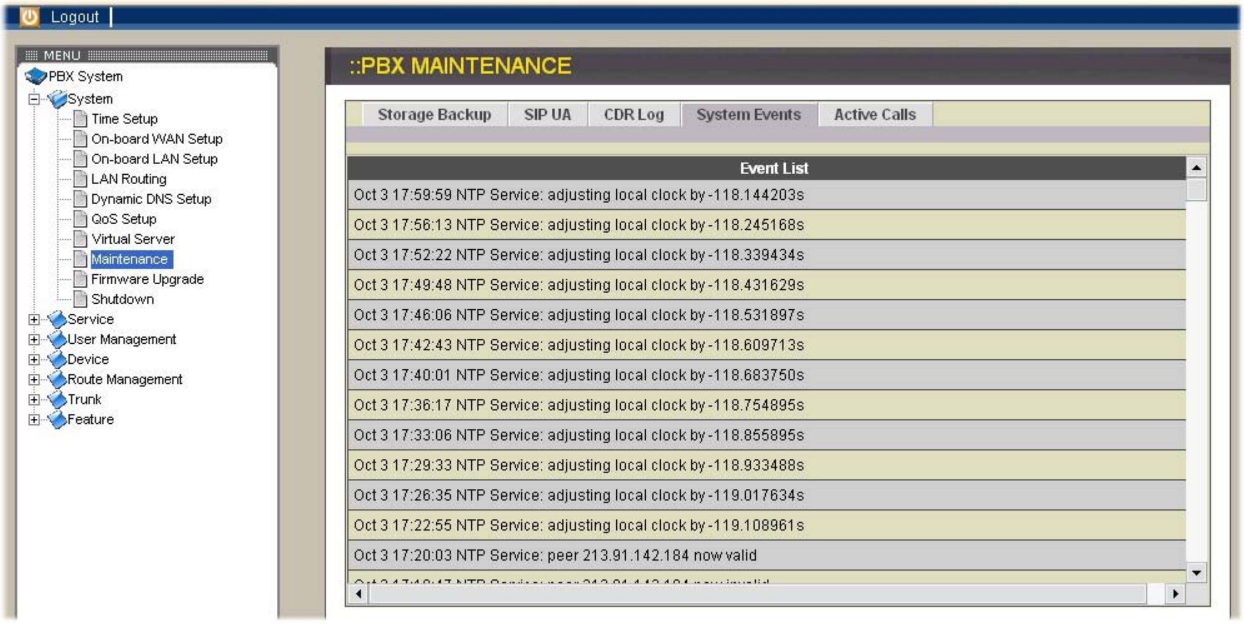
Call Duration (sec): the duration from Start Time to End Time

Billable Time (sec): the duration from Answer Time to End Time

Result: show that the call is answered or not



System Events
Event log includes reported events from system services including: NTP, DNS, DHCP, and PPPoE. You can click System Events icon to see complete records.



Active Calls
Click Active Calls icon to display the active call status.

2.1.9. Firmware Upgrade

Select System→Firmware Upgrade and the version of the running PBX firmware could be found in the page. To upgrade current firmware, you need to locate a release file obtained from the vendor and then click UPGRADE icon. Note that the filename of firmware should not be changed; otherwise system will refuse to upgrade it.

2.1.10. Shutdown

By selecting System→Shutdown, you can shutdown the machine after clicking Yes icon. If you have checked the checkbox Reboot after shutdown, the system will reboot after shutdown. Please press and release the hardware reset button quickly to reboot system in case of unsuccessful software rebooting.

2.2. Service

This chapter describes configurations for various services provided by VOI-9200.

2.2.1. NTP Service

Select Service→NTP Service to specify a NTP server for network time synchronization. You can enable or disable NTP service at any time.

Enable NTP service

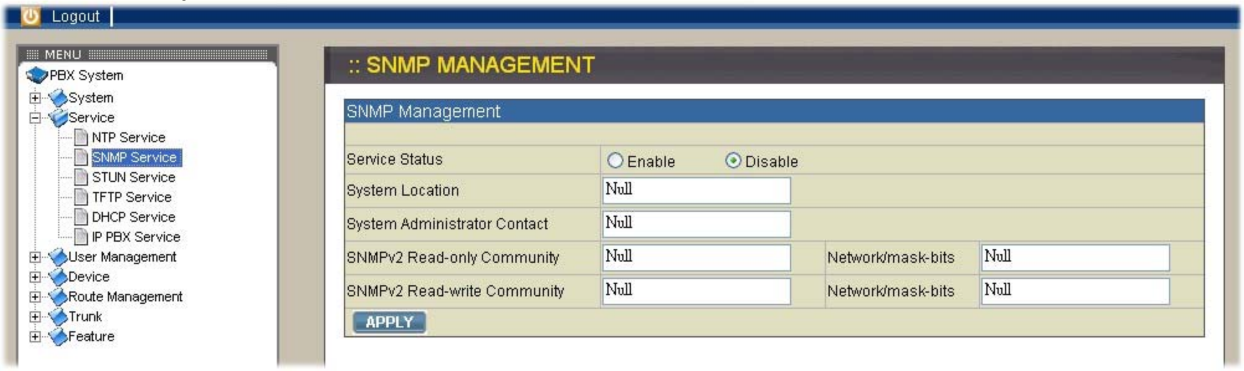
Select Enable button and then enter the fully qualified domain name (FQDN) or the IP address of a NTP server.   Click APPLY icon to save the change.

Disable NTP service

Select Disable button and then click APPLY icon.

2.2.2. SNMP Service

Select Service→SNMP Service to specify Simple Network Management Protocol (SNMP) parameters for networking status retrieval. You can enable or disable SNMP service at any time.



Enable SNMP service

Select Enable button then enter the values of System Location, Administrator Contact, read-only community, and finally click APPLY icon to save the changes. For example, you can key in the values of snmpserver.xxx.com, irving@xxx.com, and public in turn.

Disable SNMP service

Select Disable button and then click APPLY icon.

2.2.3. STUN Service

Select Service→STUN Service to specify a Simple Traversal of UDP through NATs (STUN) server for NAT traversal. You can enable or disable STUN service at any time.

Enable STUN service

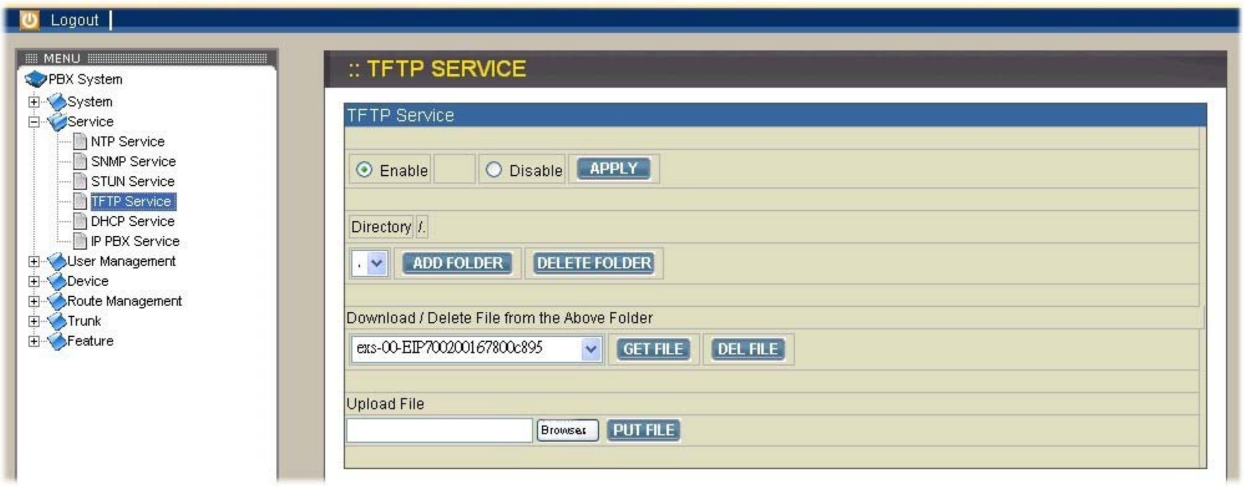
Select Enable button then enter the fully qualified domain name (FQDN) or the IP address of a STUN server. You have to click APPLY icon to save the change.

Disable STUN service

After selecting Disable button, you can enter the fully qualified domain name (FQDN) or the static IP address of the external WAN interface if needed and then click APPLY icon. Usually this address refers to as the static WAN IP address if there is a NAT device between the IPPBX and Internet. If the WAN port of IPPBX directly connects to Internet or it is unused, leave the address blank.

2.2.4. TFTP Service

Select Service→TFTP Service to show current status of TFTP service in this page. You can enable or disable TFTP service at any time.



Enable TFTP service
Select Enable button, and then click APPLY icon. Afterward you are able to do file management, for example, upload files into or download files from IP PBX through TFTP service.

Change Directory
Choose the desired directory in Directory list. By default, the root directory is /tftpboot. Initially, you might not be able to change directory, since there is no other folder created under /tftpboot yet. Current directory is shown in gray on the right side field, for instance, as /tftpboot at the beginning.

Add Folder
Choose a directory under which you want to add a new folder then click Add Folder icon. Enter a name of the new folder in the pop-up window, say, myfolder, then click OK icon. The new folder is then created accordingly, say, /tftpboot/myfolder.

Delete Folder
Choose a directory you would like to delete from Download list then click Delete Folder icon. The folder you just deleted shall disappear from the list.

Download File
Choose a directory in Download list from where you would like to download a file, and select a file in this directory from Download file in the above folder list. Subsequently click Get File icon to download the file.

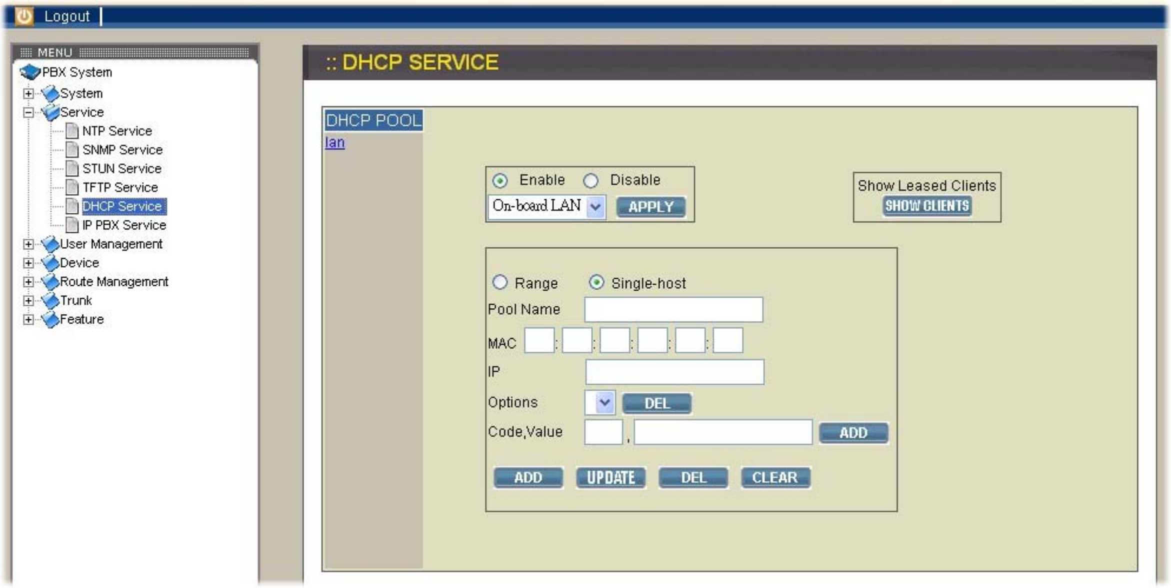
Delete File
Choose a directory in Download list where you would like to delete some file and select the file from the list. Click Delete File icon to delete the file.

Upload File
Choose a directory in Download list where you would like to upload some file, and click Browse icon to locate the file in the local storage. Click Put File icon to upload the file. Now, the file you just uploaded should appear in current directory and is displayed in the Download file in the above folder list.

Disable TFTP service
Select Disable button, and then click APPLY icon.

2.2.5. DHCP Service

Select Service→DHCP Service to display current status of DHCP service in this page. You can enable or disable DHCP service at any time.



Enable DHCP service

Select Enable button, and then click APPLY icon. Afterward, you can configure more DHCP settings in this page.

Add DHCP pool

Click <Add new> icon from the left panel. At first, specify a pool ID which must have an alphabet initial in the ID text field. Secondly, check Single host checkbox if the binding is intended for a specific host only then give the MAC address of the host right below. Thirdly, enter the DHCP range of addresses available for lease in the two Range text fields; Start and End. If Single host checkbox is checked, the End text field will be grayed out. Optionally, DHCP options could be configured by entering an option code in Code text field and the option value in Value text field then click Add icon. The just added DHCP option shall be displayed in the Options list. Follow the same steps to add more DHCP options. To delete an option, choose it from the Options list and click Delete icon. Last, click Save icon to commit changes. You should be able to see the newly added DHCP pool displayed in the DHCP Pool panel on the left side.

Edit DHCP Pool

Click the link of the pool ID listed in the DHCP Pool panel, and enter desired settings shown on the right side. Click APPLY icon to save the changes.

Delete DHCP Pool

Click the link of the pool ID which you want to delete from the DHCP Pool panel. Click Delete icon to delete it. The just deleted pool ID shall disappear from the DHCP Pool panel.

Show Clients

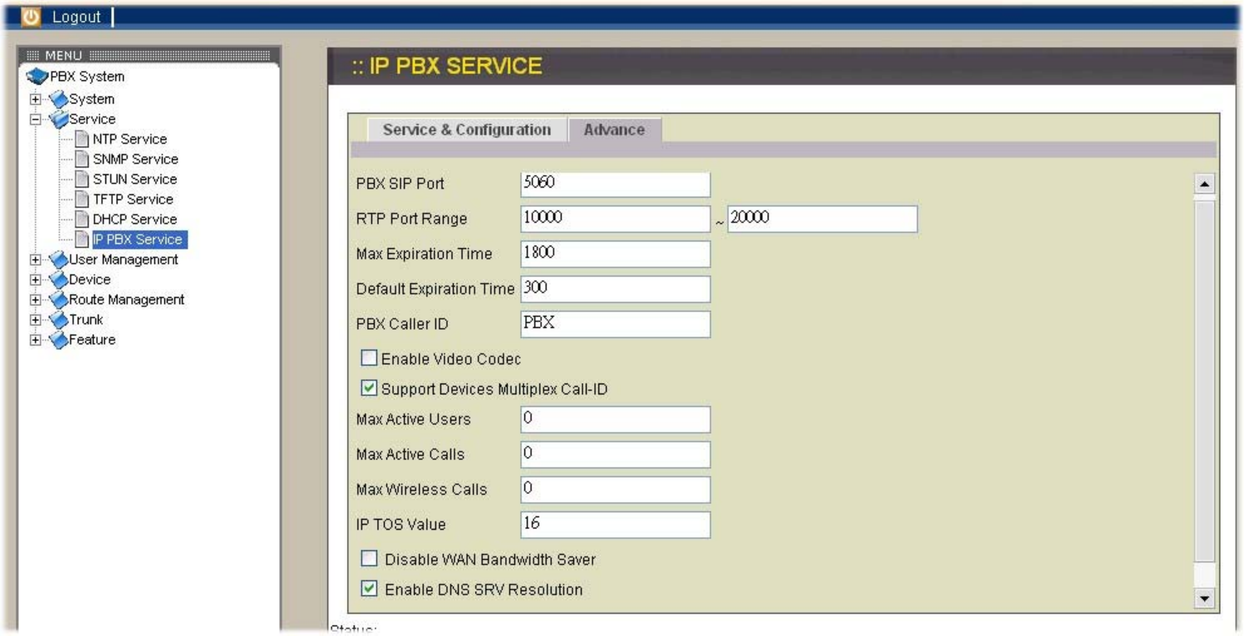
Click the Show client icon to list all leased LAN IP addresses and client details in a new pop-up window.

Disable DHCP service

Select Disable button then click APPLY icon.

2.2.6. IP PBX Service

Select Service→IP PBX Service to specify IPPBX global parameters in Advance subpage. Besides, you can also reload, backup, or restore IPPBX configuration and restart IPPBX service in Service & Configuration subpage.



IPPBX global parameters

Global SIP settings, call records, and status of clients could be configured in the first half of Advance page.

PBX SIP Port

The UDP port where the SIP service listens on

RTP Port Range

The range of RTP ports used by the IPPBX for media transporting

Max Expiration Time

Maximum available duration of SIP registration

Default Expiration Time

Default available duration of SIP registration

PBX Caller ID

The default Caller ID if that of an incoming call is unknown

Enable Video CODEC

A checkbox for enabling video CODEC of the IP PBX if there are video clients registering to the system

Support Devices Multiplex Call-ID
A checkbox for forcing discrimination of SIP tags. Do this only when there is such a client device in the system and other devices supporting the same feature. Otherwise, one may find the special device only got registered with this option but other clients or even SIP trunks failed due to such change.

Disable WAN Bandwidth Saver
A checkbox for disabling the logic to attempt low-bit-rate over WAN

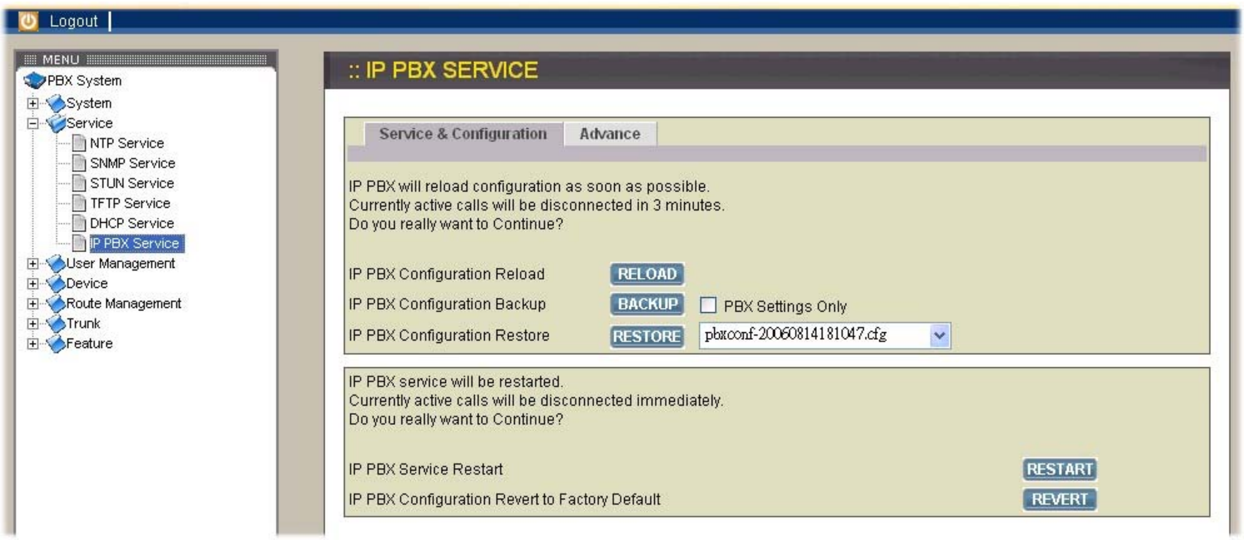
Enable DNS SRV Resolution
A checkbox for enabling DNS SRV resolution for phones or remote SIP trunks not yet registered

Max Active Users
Maximum number of actively registered clients

Max Active Calls
Maximum number of concurrent calls

Max Wireless Calls
Maximum number of calls made by explicitly specified wireless extensions

Note that IP PBX service must be reloaded to activate changes in Service & Conf page.



Reload IPPBX configuration
By clicking **RELOAD** icon, IP PBX will reload configuration immediately when there is no active call. Current active calls will be retained up to 3 minutes. It is one of the most useful functions for the IP PBX to reload to activate all configuration changes.

Backup IPPBX configuration
By clicking **BACKUP** icon, IP PBX archives and encrypts current configuration into a time-stamped backup file under /tftpboot. To secure configuration files, it is suggested that the files be download to a local host by the TFTP service once in a while. Note that the filename of the configuration file should not be changed; otherwise, it will be rejected when you want to restore it.

Restore IPPBX configuration
Select a configuration backup file from the list and click **RESTORE** icon, and then IP PBX will restore the recorded settings in the file as current settings. Remember to reload to activate configurations.

Restart IPPBX service
Click **RESTART** icon, and the IP PBX service will completely restart. All active calls will be disconnected immediately. This function should be rarely required unless the LAN setting has been changed or the operating service is found abnormally without problematic configuration.

Revert IPPBX configuration
Click REVERT icon, and IP PBX will erase current settings and revert configuration back to the factory default. Note the reversion affects IPPBX services only but not the rest system services, such as DHCP, TFTP, QoS, NTP, STUN, etc. The backup IPPBX configuration files under TFTP remain intact after reversion, so that one can restore to a specific time if a backup file had been generated at that time.

To completely revert the whole system back to the factory default, press the hardware reset button and hold for 10 seconds before release. Since this will wipe out everything generated by the user, remember to download configuration files under /tftpboot/ to a local host in case they may be used later. However, all system interfaces and services must be configured from scratch again.

2.3. User Management

This chapter describes the procedures to configure IP telephony part for the IP PBX. Usually, it is required to reload configuration in order to make new configuration effective. Before reloading IP PBX, you need to configure in sequence the user group, user, device, route, route group, and trunk pages as follows.

2.3.1. Usergroup

A usergroup is a logical grouping of users and their privileges. For instance, one could belong to multiple usergroups in an IP telephony network, such as Sales, Marketing, Administration, Accounting, and Engineering, etc. Each usergroup associates with a set of PBX features and call routing scopes. In other words, users in the same usergroup share the same usage privilege of PBX features and final destinations.

The Usergroup Management page allows the administrator to manage usergroups. By selecting User→Usergroup, one can add, edit, or delete usergroups. Note that you have to reload to activate changes.

1. Add Usergroup

Click <ADD> icon from the panel
Enter settings shown in Table 2-1.
Click APPLY icon.
Now, you should be able to see the newly added usergroup displayed in the Groups panel.

2. Edit Usergroup
Click the usergroup link in Groups panel.
Edit settings shown on the right side.
Click APPLY icon.

3. Delete Usergroup
Click the usergroup link in Groups panel.
Click Delete icon.
The deleted usergroup shall disappear from the Groups panel.

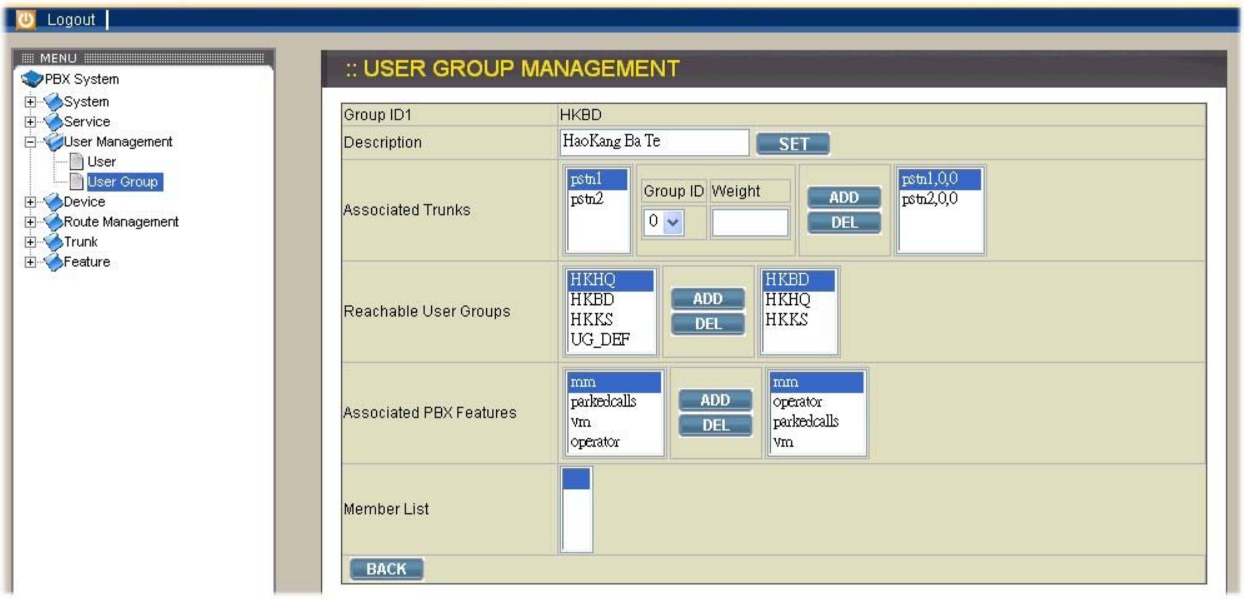
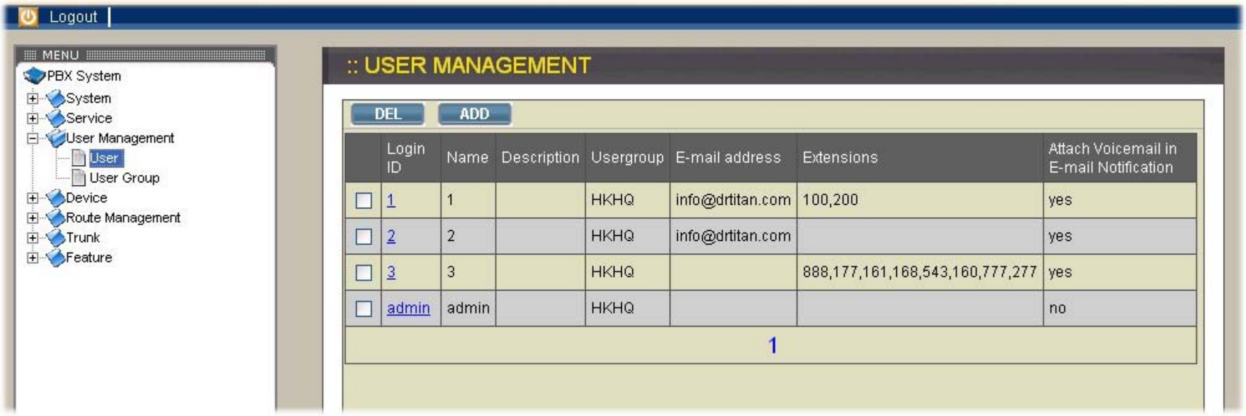


Table 2-1 Usergroup Configuration Settings

Field	Description
Group ID	A unique name of this group composed of alphabets, numbers, and underscore but without spaces; 32 characters maximum
Description	Arbitrary description info
Associated trunks	Select outbound SIP trunks and PSTN trunks accessible by this usergroup. Note the order matters the hunting sequence in run time. ☞ There is no SIP trunk and PSTN trunk initially. Come back later to revise selection once trunks have been created.
Reachable usergroups	Select other usergroups reachable from this usergroup. By default, only users in the same usergroup can be reached one another. ☞ There is no usergroup initially. Come back later to revise selection once more usergroups have been created.
Associated PBX features	Select PBX features enabled to this usergroup. Here vm stands for voice mail, mm for meet-me conference, parkedcalls for call parking, operator for operator service. ☞ Most features have to be configured to function correctly. Remember to examine the settings of selected features before activating current configuration.
Member list	Show the users belonging to this group.

2.3.2. User

A user is a logical entity in IP telephony which associates extensions with a usergroup. It also propagates its attributes such as e-mail and voicemail PIN to extensions. Usually a user refers to a real person who has a name and e-mail; however, one can always create virtual users to associate with public extensions. For example, extensions in lobby, tea room, or meeting room, etc.



The User Management page allows the administrator to manage users in the IP telephony network. Select User Management→User, one can add, edit, and delete users. Note that you have to reload to activate changes.

1. Add User

Click <ADD> icon from the panel.
Enter settings shown in Table 2-2.
Click APPLY icon.
The newly added user should be displayed in the Users panel on the left side.

2. Edit User

Click the link of the user to edit from the Users panel.
Edit settings shown on the right side.
Click UPDATE icon.

3. Delete User

Click the link of the user to delete from the Users panel.
Click Delete icon to confirm deletion.
The deleted user shall disappear from the Users panel.
☞ A user can be deleted only when no extension is associated with it.

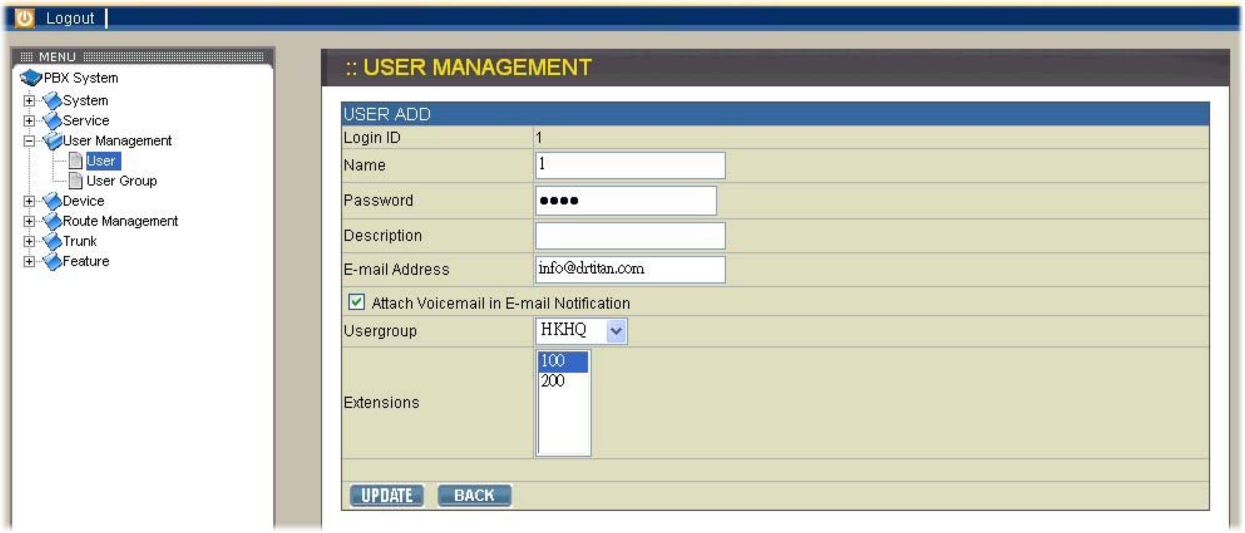


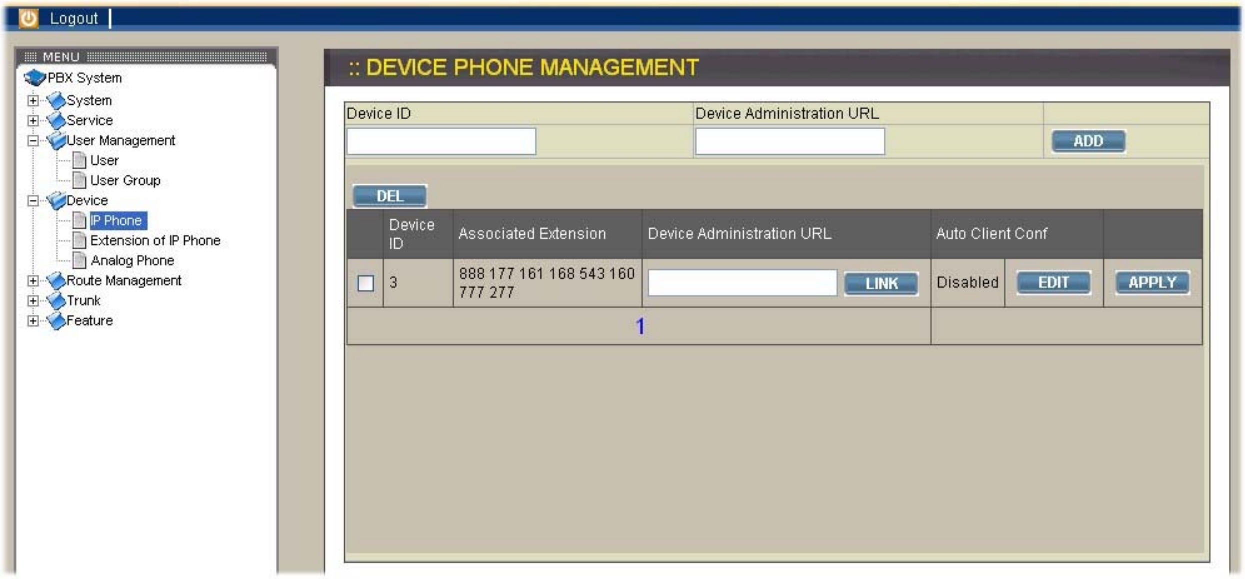
Table 2-2 User Configuration Settings

Field	Description
Login ID	A unique name of this user composed of alphabets, numbers, and underscore but without spaces; 32 characters maximum ☞This is the ID for the user to access the IPPBX Web GUI for management.
Password	The password for the user to access IPPBX Web GUI for management
Name	The name of the user, either a real or a virtual one, e.g. Alice Lee or Conference Room
Description	Arbitrary description info
E-mail address	An e-mail address of the user for voicemail notification
Attach voicemail in e-mail notification	Check if to enclose the voice message received in the notification e-mail as an attachment or not.
Associated usergroup	Select the usergroup this user belongs to. ☞ There is no usergroup initially. Come back later to revise selection if no appropriate usergroup could be chosen now.
Associated extensions	List extensions associated with this user.

2.4. Device

A device could be an IP phone, gateway, analog telephone adapter, or even another IP PBX, etc. It has one or more extensions to be registered to the IP PBX.

The Device Phone Management page lets the administrator to create accounts for device extensions. Note the same account information has to be programmed into the device through the configuration interface. Select Device→IP Phone, one can add, edits, and deletes devices. Note IP PBX service must be reloaded to activate changes.



2.4.1. IP Phone

Add Device

Click <ADD> button from the panel.

Enter settings shown in Table 2-3.

Click APPLY button.

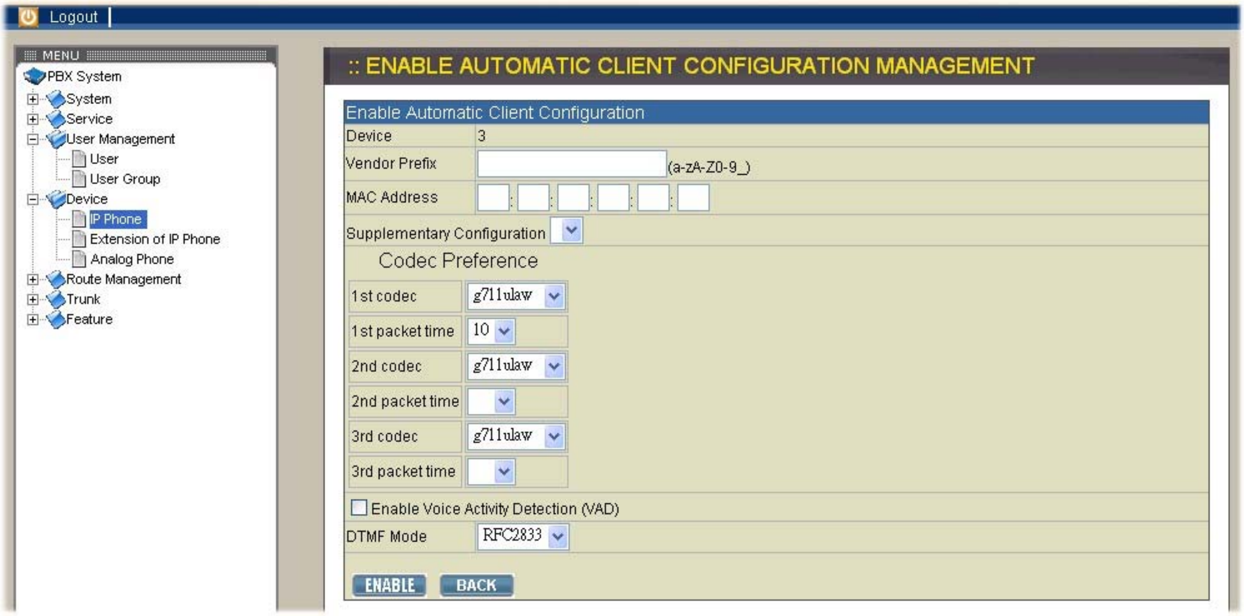


Table 2-3 Device Configuration Settings

<table>
<tr><th>Field</th><th colspan="2">Description</th></tr>
<tr><td>Device ID</td><td colspan="2">A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.</td></tr>
<tr><td>Device administration URL</td><td colspan="2">(Optional) Administration URL of the device.</td></tr>
<tr><td rowspan="7">Enable Automatic Client Configuration</td><td colspan="2">(Optional) For phones support auto-config, check to specify the MAC address and audio preferences of the phone. Each field is explained as followings. Note that for phones using HTTP for auto-config, DHCP setting needs a new option 151 with a value of http://<IP PBX LAN IP>/tftpboot/. No extra setting if the phone uses TFTP for auto-config.</td></tr>
<tr><td>Vendor prefix</td><td>Specify if provided by phone</td></tr>
<tr><td>MAC address</td><td>MAC address of the device</td></tr>
<tr><td>Supplementary configuration</td><td>Specify if provided by phone</td></tr>
<tr><td>Codec preference</td><td>Preference order of codecs and packet times by phone</td></tr>
<tr><td>Enable VAD</td><td>VAD on phone</td></tr>
<tr><td>DTMF mode</td><td>DTMF mode used by phone</td></tr>
</table>

2.4.2. Extension of IP Phone

The newly added device should be displayed in Devices panel. The following shows steps to add extensions for the new device.

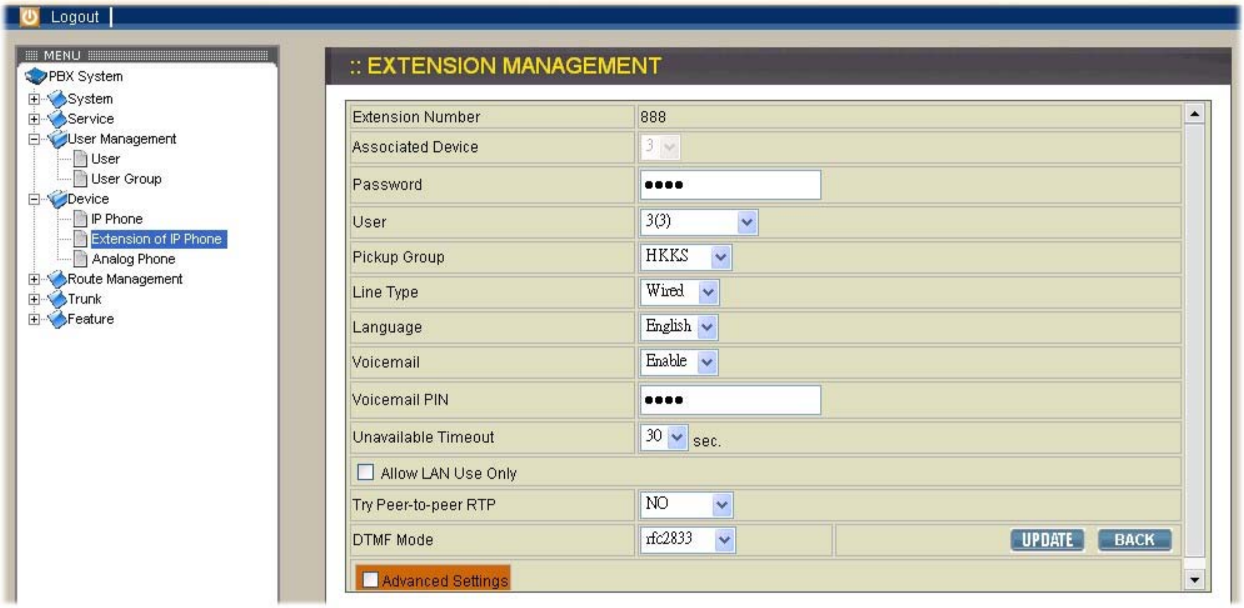
Click link of the device shown in the **Devices** panel.
Click **Extension** of **IP Phone** button to the extension management page.
Click <ADD> button from the panel.
Enter settings shown in Table 2-4
Click APPLY button.

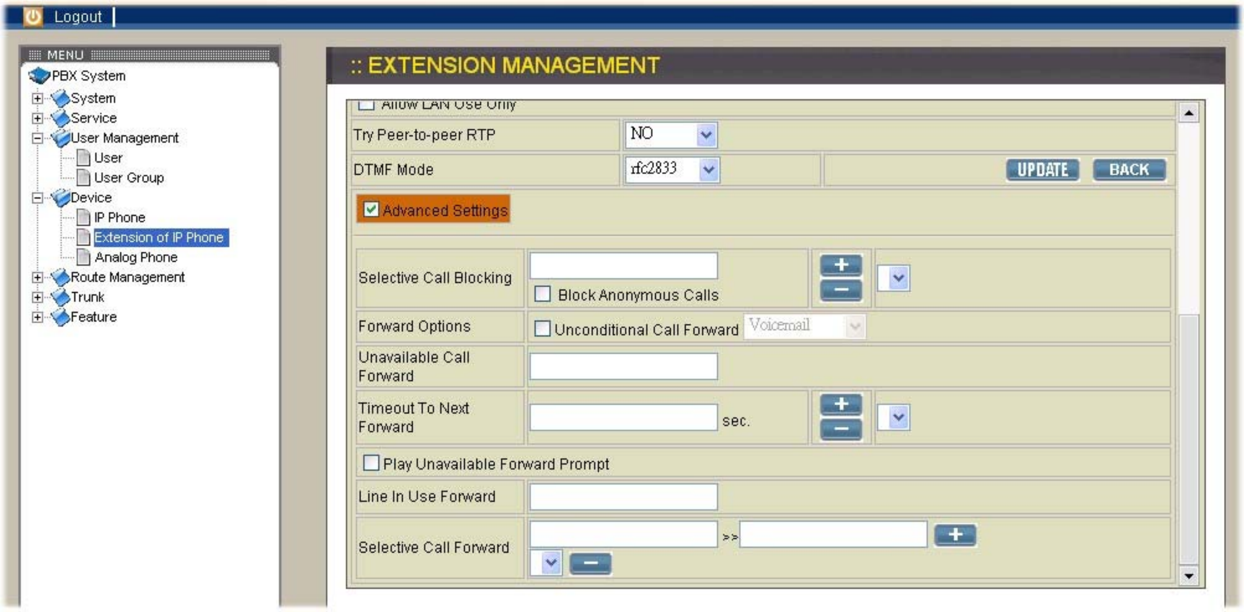
	Extension Number	Associated Device	Pickup Group	Unavailable Timeout	Line Type	User	Voicemail Enable	Language	Allow LAN Use Only	DTMF Mode	Try Peer-to-peer RTP	Rejected Caller
☐	888	3	HKKS	30	wired	3(3)	yes	en	no	rfc2833	NO	
☐	177	3	HKHQ	30	wired	3(3)	no	en	no	rfc2833	NO	
☐	161	3	HKHQ	20	wired	3(3)	no	en	no	rfc2833	NO	
☐	168	3	HKHQ	20	wired	3(3)	no	en	no	rfc2833	NO	
☐	543	3	HKHQ	20	wired	3(3)	no	en	no	rfc2833	NO	
☐	160	3	HKBD	20	wired	3(3)	no	en	no	rfc2833	NO	
☐	777	3	HKHQ	30	wired	3(3)	yes	en	no	rfc2833	NO	
☐	277	3	HKHQ	30	wired	3(3)	no	en	no	rfc2833	NO	

The newly added extension of this device should be displayed in Extensions panel on the left side. More extensions could be added by repeating the last 3 steps above.

Edit Device
Click device link in Devices panel.
Edit settings shown on the right side.
Click APPLY button.

Extensions associated with this device could be modified by the following steps.
Click Extensions button.
Click extension link in Extensions panel.
Edit settings and Advanced Settings shown on the right side.
Click UPDATE button.





Delete Device
To delete one or more extensions associated with a certain device, follow steps below.
Click device link in Devices panel.
Click Extensions button.
Click extension link in Extensions panel.
Click Delete button.

The deleted extension shall disappear from the Extensions panel.
Once a device has no extension, it can be deleted.
Click device link in Devices panel.
Click Delete button.
The deleted device shall disappear from the Devices panel.

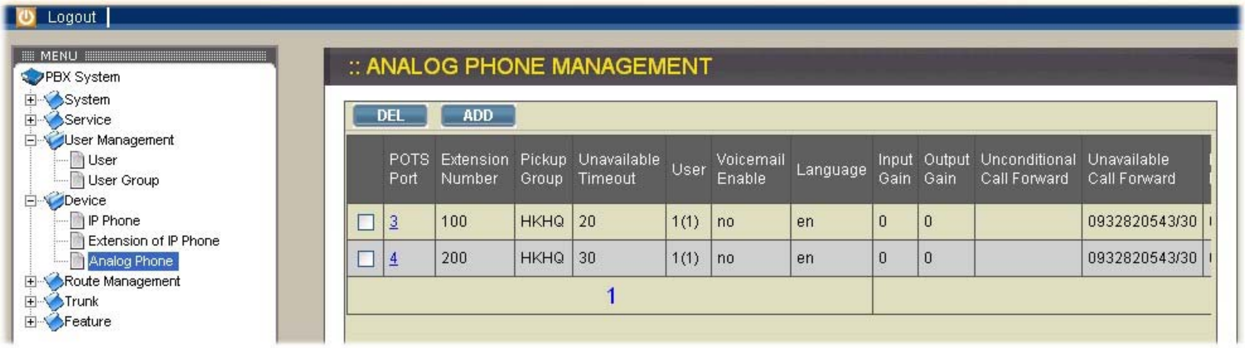
Table 2-4 Device Extension Configuration Settings

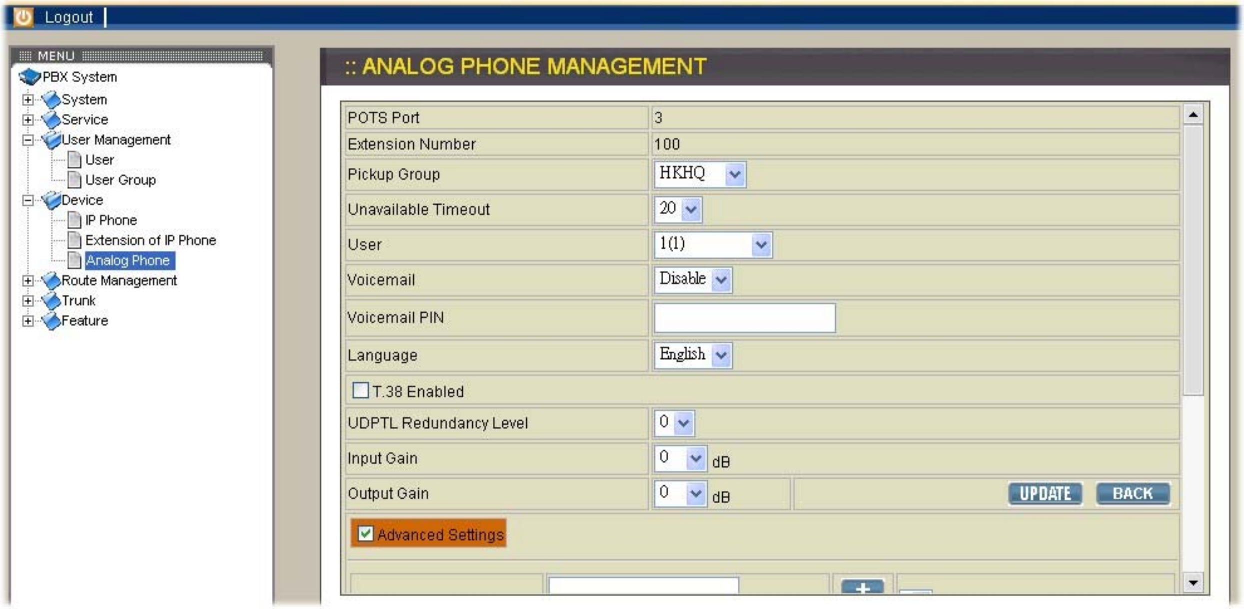
Field	Description
Extension number	A unique line number composed of digits only, e.g. 101; 32 digits maximum. This is the login ID on the device configuration side.
Password	Password of this extension. Same password must be configured on the device side as well.
Pickup group	The pickup group the extension belongs to. Extensions in the same pickup group can call *8 to pick up a call in ringing state.
Unavailable timeout	Timeout for ringing before a call is answered.
Line type	Specify the type of connection, wired or wireless, of the client with the extension.
User	Select the user this extension associates with. ☞ There may not be appropriate users to select initially. One can come back later once the expected user has been added.
Voicemail	Select enable to allocate voicemail account for the extension.
Voicemail PIN	PIN to access voicemails. This is mandatory if above voicemail option is enabled.
Language	Preferred language for system instructions heard from the extension.
Allow LAN use only	Check to reject registration and calls from WAN in a SIP ID same as the extension number. I.e., this extension must be on LAN.
Disable NAT Traversal	Respond with the device based on IP addresses given in the SIP messages instead of exploiting the received source IP address for NAT traversal.
DTMF Mode	Choose preferred DTMF mode for this extension. Currently supported types include RFC2833, SIP INFO, and in-band tone. It must match configuration on the device side.
Try peer-to-peer RTP	If checked, IP PBX will attempt to notify the two peers in a conversation to try peer-to-peer RTP transmission. This is suggested as long as phones support INVITE or UPDATE method during a connected call to save the resource of IP PBX. However, only SIP INFO DTMF mode phones should enable this since other DTMF modes require IP PBX being RTP relay server to support in-line transfer.
Unconditional call forward	(Optional) Check voicemail as default destination or enter a number to which incoming calls are forwarded unconditionally. The number could be an extension or a PSTN number with appropriate outbound prefix.
Unavailable call forward	(Optional) Enter a number to which incoming calls are forwarded when not answered. The number could be an extension or a PSTN number with appropriate outbound prefix.

Line in use forward	(Optional) Enter a number to which incoming calls are forwarded when the extension is busy. The number could be an extension or a PSTN number with appropriate outbound prefix.
Selective call blocking	(Optional) Check Block anonymous calls to block all calls without a Caller ID; one could also explicitly list numbers to block by entering one or more calling numbers and click Add button. Use Remove button and Remove all button to cancel blockings.
Selective call forward	(Optional) Unconditional call forwarding according to the calling number. Enter one or more calling numbers and a forwarding number, and click Add button. E.g., forward only calls from 101 and 102 to a cellular number, while let the rest enter the voice mail by default. Note that extensions must be separated by commas. Use Remove button or Remove all buttons when some forwardings are no longer required.

2.4.3. Analog Phone (PC#1, PC#2 only)

Connect an analog phone to an FXS port and configure the properties of each port. Detailed settings are described as in Table 2-5





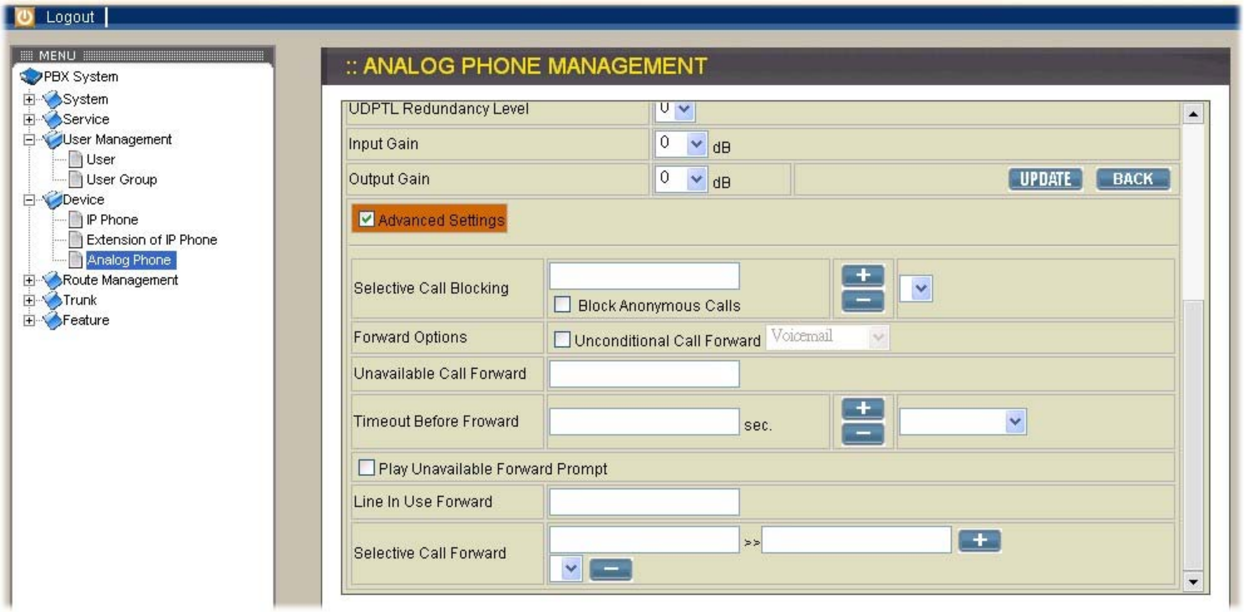


Table 2-5 FXS Extension Configuration Settings

Field	Description
POTS port	FXS port index
Extension number	A unique line number composed of numeric digits only, e.g. 101, with maximum up to 32 digits.
Pickup group	The pickup group to which the extension belongs.
Unavailable timeout	Timeout for ringing before a call is answered.
User	Select the user this extension associates with. ☞ There may not be appropriate users to select initially. One can come back later once the expected user has been added.
Voicemail	Select enable to allocate voicemail account for the extension.
Voicemail PIN	PIN to access voicemails. This is mandatory if above voicemail option is enabled.
Language	Preferred language for system instructions heard from the extension.
Input/Output gain	Voice amplification or attenuation in dB scale to adjust input/output volume.
Unconditional call forward	(Optional) Check voicemail as default destination or enter a number to which incoming calls are forwarded unconditionally. The number could be an extension or a PSTN number with appropriate outbound prefix.
Unavailable call forward	(Optional) Enter a number to which incoming calls are forwarded when not answered. The number could be an extension or a PSTN number with appropriate outbound prefix.
Line in use forward	(Optional) Enter a number to which incoming calls are forwarded when the extension is busy. The number could be an extension or a PSTN number with appropriate outbound prefix.
Selective call blocking	(Optional) Check Block anonymous calls to block all calls without a Caller ID; one could also explicitly list numbers to block by entering one or more calling numbers and click Add button. Use Remove button and Remove all button to cancel blockings.
Selective call forward	(Optional) Unconditional call forwarding according to the calling number. Enter one or more calling numbers and a forwarding number, and click Add button. E.g., forward only calls from 101 and 102 to a cellular number, while let the rest enter the voice mail by default. Note that extensions must be separated by commas. Use Remove button and Remove all button accordingly when some forwardings are no longer required.

2.5. Route Management

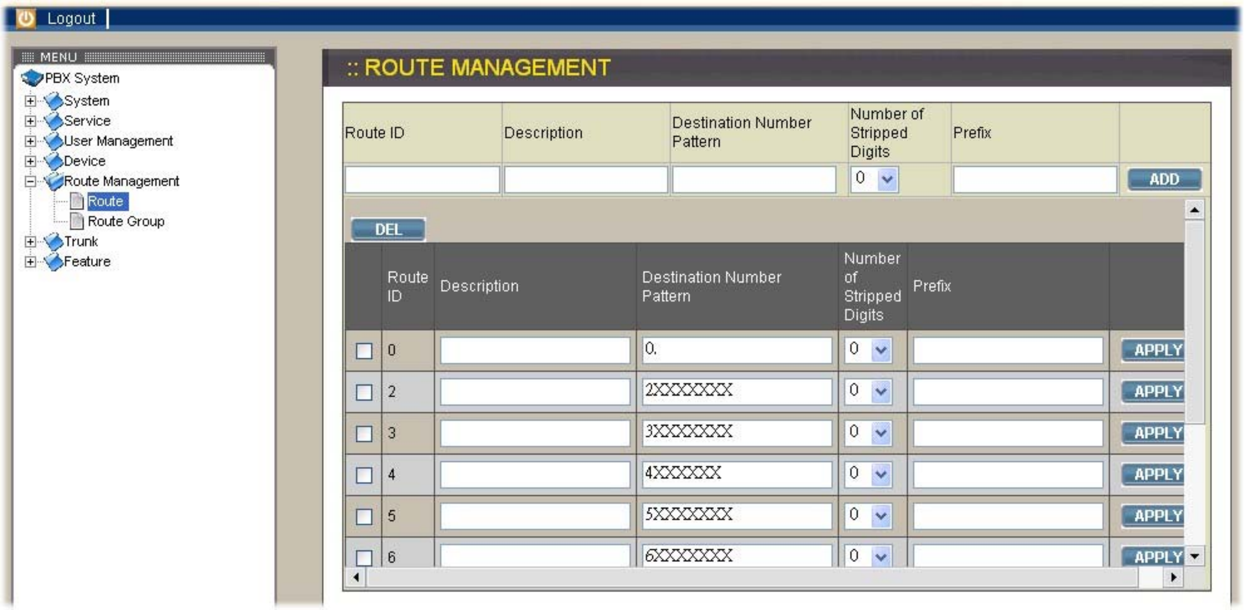
A route is a destination number pattern for outbound call matching. A pattern consists of digits 0-9, “*”, “#”, digit set, and wildcard characters like “.”, “X”, “Z”, and “N”. Table 2-5-1 explains digit set and wildcard characters.

Table 2-5-1 Digit Set and Wildcard Characters for Route Patterns

Expression	Description
[<digits>]	Match any single digit listed explicitly. For example, digit set [13579] match odd digits.
. (dot)	Match any digit in any length. Usually given in the end of a pattern to include all numbers matched a specific prefix.
X	Match any single digit from 0 to 9.
Z	Match any single digit from 1 to 9.
N	Match any single digit from 2 to 9.

2.5.1. Route

By selecting Route Management-> Route, the administrator can add, edit, and delete routes in the Route Management page. IP PBX service must be reloaded to activate changes.



Add Route
Click <ADD> button from the left panel.
Enter settings shown in Table 2-5-2
Click APPLY button.
The newly added route should be displayed in Routes panel on the left side.

Edit Route
Click the link of the route to edit from the Routes panel.
Edit settings shown on the right side.
Click APPLY button.

Delete Route
Click the link of the route to delete from the Routes panel.
Click Delete button.
The deleted route shall disappear from the Routes panel.

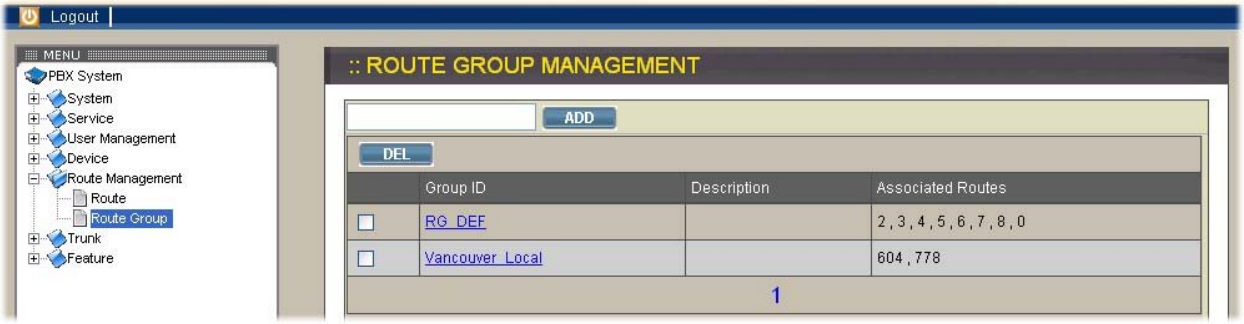
Table 2-5-2 Route Configuration Settings

Field	Description
Route ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description info.
Destination number pattern	A destination number pattern consisting of digits, digit set, and wildcard characters, e.g. 9NXXXXXX matches any 7-digit called number starting from a digit larger or equal to 2 and with an extra prefix digit 9.
Number of stripped digits	Number of leading digits to be stripped from the original dialed number when matches this route. Using 9NXXXXXX as an example route pattern with number of stripped digits equal to 1, dialing 95270001 will be stripped to be 5270001 when it actually got dialed out.
Prefix	A sequence of digits to be prefixed to the final dialed number after stripping. Using 9NXXXXXX as an example route pattern with number of stripped digits equal to 1 and prefix 1408, dialing 95270001 will be 14085270001 when it actually got dialed out. A special prefix character “w” could be used for PSTN trunks to pause 0.5 second during dialing. Say, 4 leading consecutive “w” result in 2 seconds delay before dialing.

2.5.2. Route Group

A route group routes into a logical superset of route patterns. Such abbreviation simplifies the association of multiple routes with a trunk, say, a PSTN line. A route could be included in various routegroups and a routegroup could contain one single route only.

By select Route Management -> Route Group, the administrator can add, edit, and delete routegroups in the Route Group Management page. IP PBX service must be reloaded to activate changes.



Add Routegroup

Click <ADD> button from the left panel.
Enter settings shown in Table 2-5-3
Click APPLY button.
The newly added routegroup should be displayed in Groups panel on the left side.

Edit Routegroup

Click the link of the routegroup to edit from the Groups panel.
Edit settings shown on the right side.
Click SET button.

Delete Routegroup

Click the link of the routegroup to delete from the Groups panel.
Click Delete button.
The deleted routegroup shall disappear from the Groups panel.

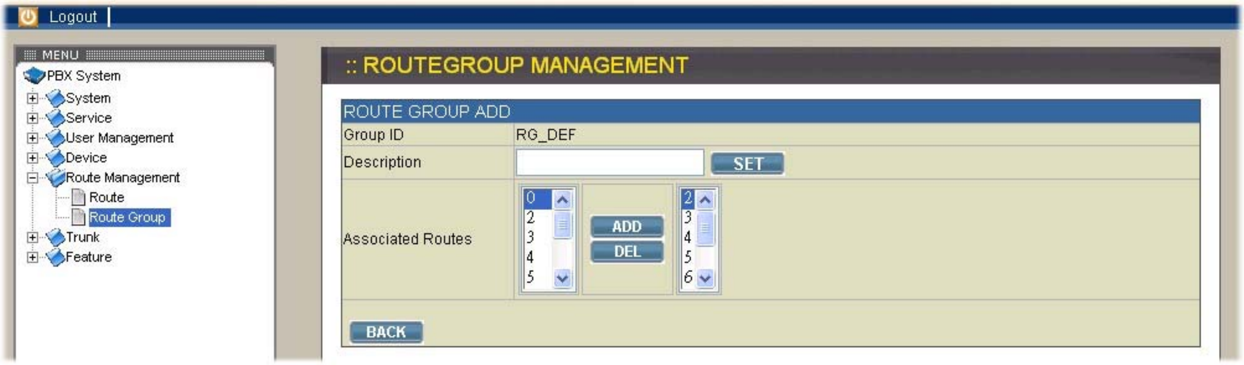


Table 2-5-3 Routegroup Configuration Settings

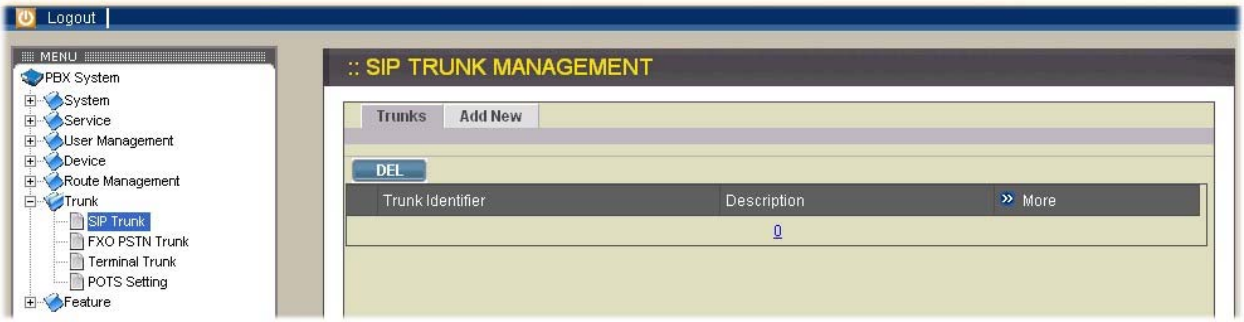
Field	Description
Group ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description info.
Associated routes	Select routes belonged to this routegroup. Click arrow icons to add or remove a route to/from the routegroup. The right box lists currently selected routes. Note the order of selected routes is important since it decides which route would be matched first for an outgoing call. ☞ There may not be appropriate routes to select initially. One can come back later to revise it once the expected routes are added.

2.6. Trunk

A SIP trunk refers to a SIP account on a remote call routing or gateway device. A practical example is an account at an Internet Telephony Service Provider (ITSP) where a call is routed to a SIP client or off-ramped to an analog subscriber via PSTN. One could also build SIP trunk to a remote IP PBX to reach its extensions and PSTN ports.

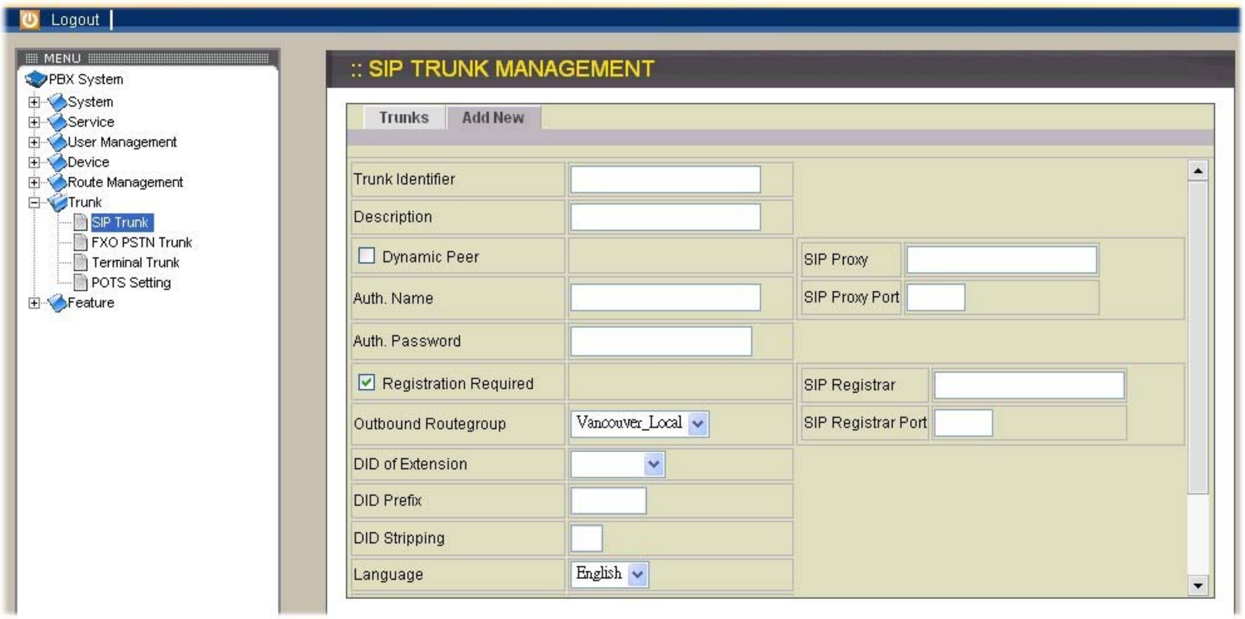
2.6.1. SIP Trunk

The SIP Trunk Management page allows the administrator to configure SIP trunks used by VOI-9200 IP PBX. Select Trunk -> SIP Trunk, and one can add, edit, and delete SIP trunks. Note IP PBX service must be reloaded to activate changes.



Add SIP Trunk

Click <Add new> button from the left panel.
Enter settings shown in Table 2-6-1
Click APPLY button.
The newly added SIP trunk shall be displayed in Trunks panel on the left side.



Edit SIP Trunk
Click the link of the SIP trunk to edit from the Trunks panel.
Edit settings shown on the right side.
Click Save button.

Delete SIP Trunk
Click the link of the SIP trunk to delete from the Trunks panel.
Click Delete button.
The deleted SIP trunk shall disappear from the Trunks panel.

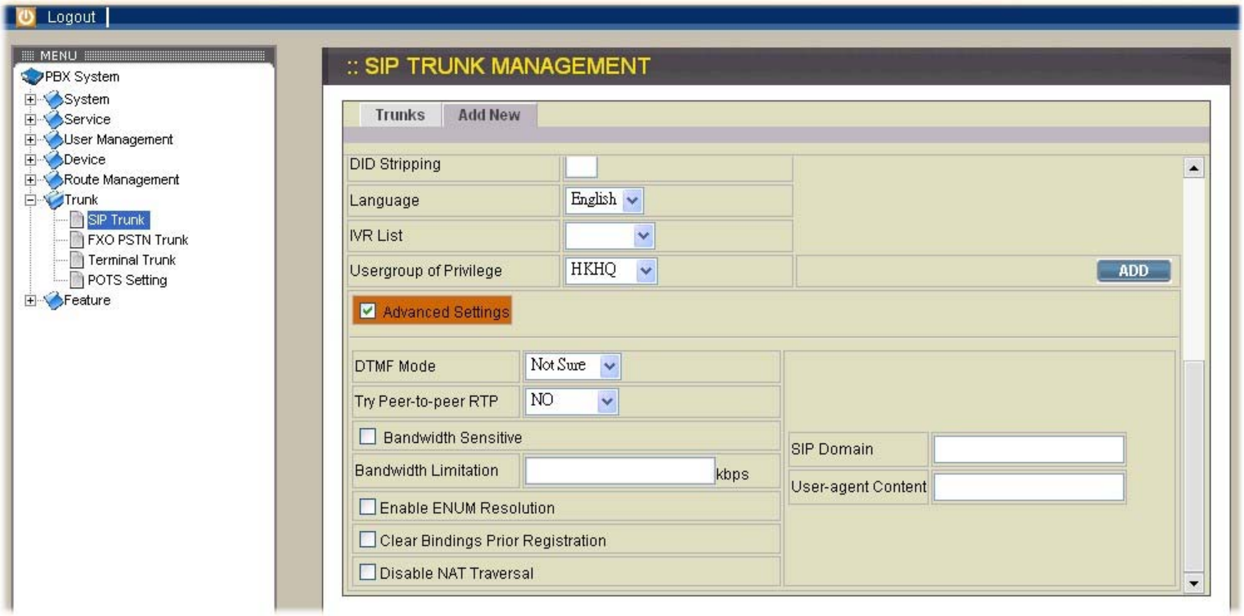


Table 2-6-1 SIP Trunk Configuration Settings

Field	Description
Trunk identifier	A unique number consisting of digits only. Usually give the phone number issued by the ITSP for consistency.
Description	Arbitrary description info.
Block all calls	Check this to create a dummy blocking trunk. Outbound calls matching routegroup associated with this trunk will be blocked.
Dynamic peer	Check if the trunk is a passive trunk which means the registration will be from a dynamic remote peer. Typical application is to accept registration from an IP PBX at a remote site with dynamic IP address. Once the remote IP PBX registers, calls from local to remote can be made reversely over the trunk.

Field	Description
Dynamic peer	Check if the trunk is a passive trunk which means the registration will be from a dynamic remote peer. Typical application is to accept registration from an IP PBX at a remote site with dynamic IP address. Once the remote IP PBX registers, calls from local to remote can be made reversely over the trunk.
User Agent Content	Override default User-Agent header content.
SIP domain	Specify the SIP domain used by the proxy and registrar. If not specified, IP address will be used as the domain by default.
SIP proxy IP or FQDN	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP proxy, which usually refer to an ITSP SIP server.
SIP proxy port	
SIP registrar IP or FQDN	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP registrar, which usually refer to an ITSP SIP server (same as proxy).
SIP registrar port	
Registration required	Check if registration to a registrar is required to activate the trunk. This is true for a remote IP PBX or an ITSP account, however, may be not required in case of a SIP gateway.
Trunk password	Give password used for authentication on the remote SIP proxy or registrar. Usually this is given by the ITSP.
Enable ENUM resolution	Enable ENUM resolution with suffix e164.arpa and e164.org. Appropriate DNS server setting under WAN interface configuration should be given accordingly. If ENUM resolution fails, the call will fall back to a regular outbound SIP trunk call based on the specified proxy info.
Clear bindings prior registration	Send a binding clearing request to the proxy before each registration request.

Field	Description
Outbound routegroup	Select a routegroup to associate routes with this trunk. Outbound calls match included route patterns could employ this trunk to hop onto a remote SIP domain. ☞ There may not be appropriate routegroup to select initially. One can come back later to revise it once the expected routegroup has been added.
DID of extension	When enabled DID, choose an extension to be an unconditional destination for incoming calls to this trunk. The number of the SIP trunk is therefore regarded as the direct line number of the extension.
Language	Preferred language for system instructions heard from the trunk.
Usergroup of privilege	When disabled DID, select a usergroup whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this trunk. ☞ There may not be appropriate usergroups to select initially. One can come back later once the expected usergroup has been added.
Disable NAT Traversal	Respond with the proxy based on IP addresses given in the SIP messages instead of exploiting the received source IP address for NAT traversal.
DTMF Mode	Choose preferred DTMF mode for this trunk. Currently supported types include RFC2833, SIP INFO, and in-band tone. It must match configuration on the server side.
Try peer-to-peer RTP	If checked, IP PBX will attempt to notify the two peers in a conversation to try peer-to-peer RTP transmission. This is suggested as long as phone and ITSP side support INVITE or UPDATE method during a connected call to save the resource of IP PBX. However, only SIP INFO DTMF mode should enable this since other DTMF modes require IP PBX being RTP relay server to support in-line transfer.
Bandwidth sensitive	Indicate the trunk is over a bandwidth-sensitive link, e.g. across Internet. If checked, a limit must be specified for call admission.

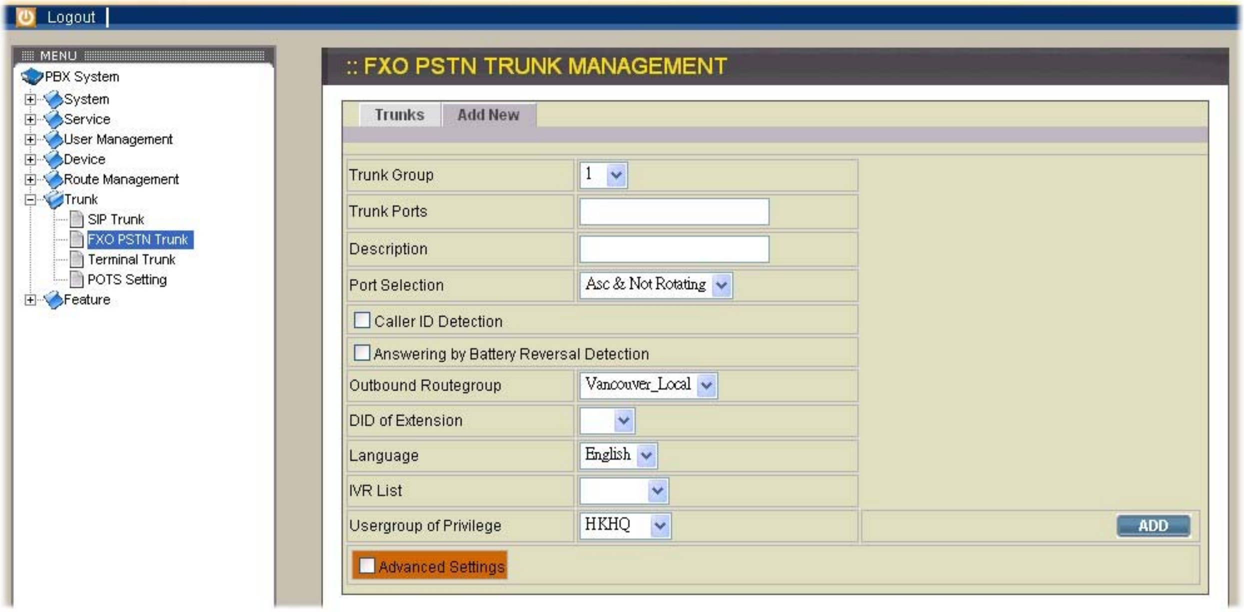
2.6.2. FXO PSTN Trunk (PC#1, PC#2 only)

A PSTN trunk group is a logical group of one or more PSTN subscriber lines connecting to FXO ports on VOI-9200 IP PBX.

The PSTN Trunk Management page allows the administrator to configure PSTN trunks. By selecting Trunk -> FXO PSTN Trunk, one can add, edit, and delete PSTN trunks. IP PBX service must be reloaded to activate changes.

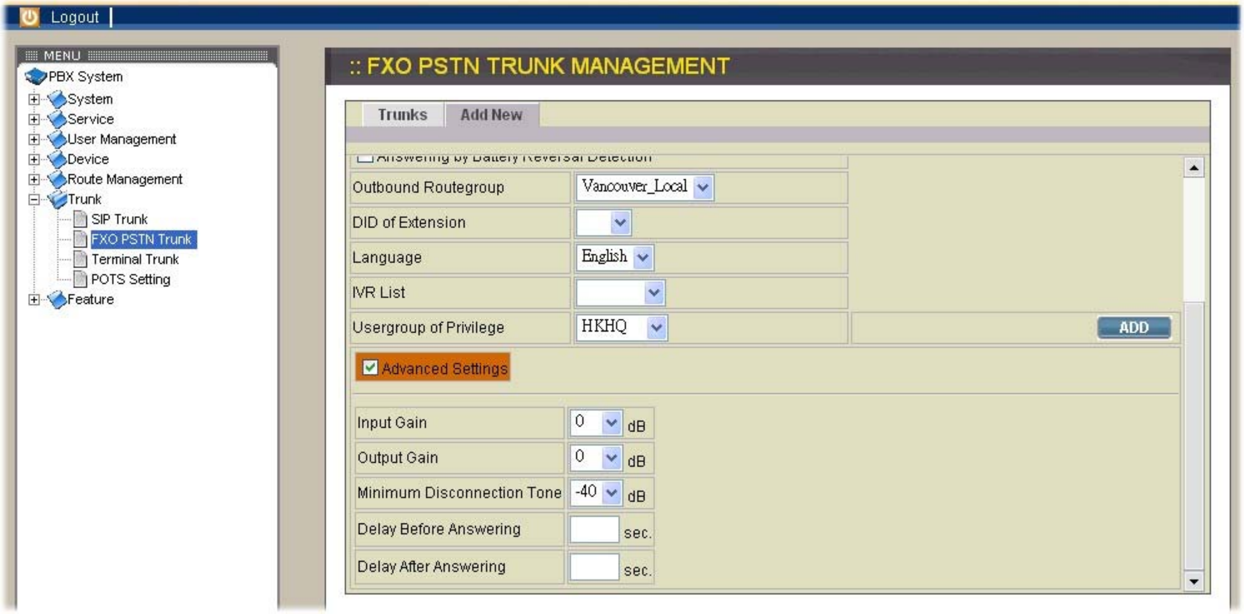
Add PSTN Trunk

Click <Add new> button from the left panel.
Enter settings shown in Table 2-6-2
Click APPLY button.
The newly added PSTN trunk shall be displayed in Trunks panel on the left side.



Edit PSTN Trunk

Click the link of trunk to edit from the Trunks panel.
Edit settings shown on the right side.
Click Save button.



Delete PSTN Trunk

Click the link of trunk to delete from the Trunks panel.
Click Delete button.
The deleted PSTN trunk shall disappear from the Trunks panel.

Table 2-6-2 PSTN Trunk Configuration Settings

Field	Description
Trunk group	ID number of this PSTN trunk group. A valid number ranges from 1 to 31.
Description	Arbitrary description info.
Trunk ports	FXO port indices grouped by this PSTN trunk, such as 1 or 1,2 or 1-3, etc. Maximum port index depends on the number of physical ports available.
Port selection	Order to hunt for an available port in the group. Besides Ascending and Descending options, one could check Rotating to force ports being selected by turns to even cost.

Field	Description
Outbound routegroup	Select a routegroup to associate routes with this trunk. Outbound calls match included route patterns could employ this trunk to access PSTN. ☞ There may not be appropriate routegroup to select initially. One can come back later to revise it once the expected routegroup is added.
DID of extension	When enabled DID, choose an extension to be an unconditional destination for incoming calls to this trunk. The PSTN numbers of the included ports are therefore regarded as the direct line numbers of the extension.
Language	Preferred language for system instructions heard from the trunk.
Usergroup of privilege	When disabled DID, select a usergroup whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this trunk. ☞ There may not be appropriate usergroups to select initially. One can come back later to revise it once the expected usergroups are added.
Input/Output gain	Voice amplification or attenuation in dB scale to adjust input/output volume of a PSTN line.
Minimum disconnection tone	Minimum volume level of the disconnection tone. If a PSTN trunk is found to have disconnection problem and voice sounds low, choose a lower dB.
Delay before/after answering	Delay in seconds before and after answering a call from PSTN trunk.

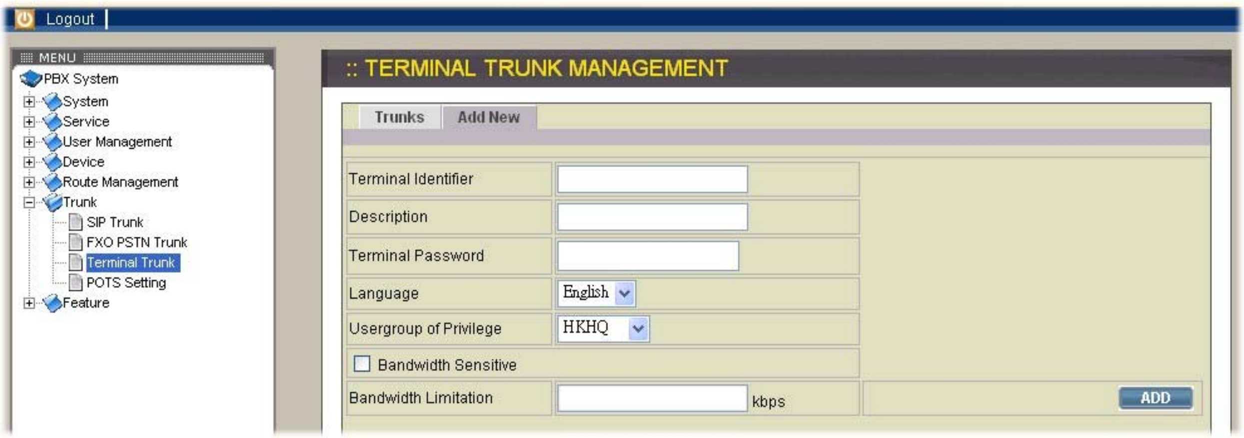
2.6.3. Terminal Trunk (PC#1, PC#2 only)

A SIP terminal trunk refers to a SIP account for a remote SIP trunk to register with. It terminates SIP registration and invitation from a remote IP PBX and relay calls to local clients, PSTN trunks, or further SIP trunks. In a site-to-site SIP trunking application, a SIP trunk on one side usually pairs with a terminal trunk on the other side to form a unidirectional call hand-off path. To allow trunking in the other direction, the two sides swap roles and form another pair. Since a terminal trunk is the account for a SIP trunk to authenticate with, exact the same identifier and password must be used for both.

The Terminal Trunk Management page allows to configure terminal trunks used by IP PBX. Select Trunk -> Terminal Trunk and one can add, edit, and delete terminals. IP PBX service must be reloaded to activate changes.

Add Terminal Trunk

Click <Add new> button from the panel.
Enter settings shown in Table 2-6-3
Click ADD button.
The newly added terminal shall be displayed in Trunk panel.



Edit Terminal Trunk

Click the link of the terminal to edit from the Trunk panel.
Edit settings shown on the right side.
Click ADD button.

Delete Terminal Trunk

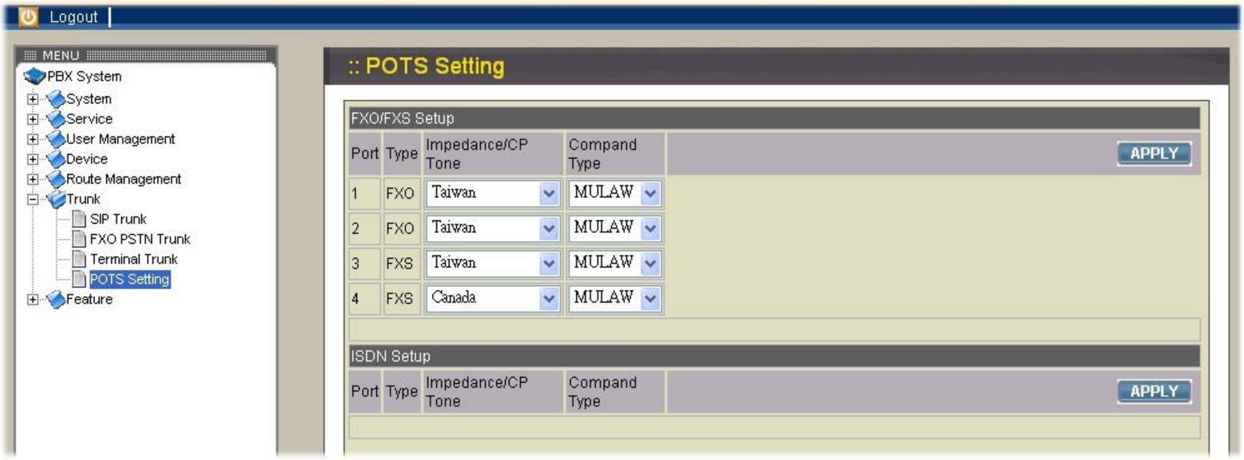
Click the link of the terminal to delete from the Trunk panel.
Click Delete button.
The deleted terminal shall disappear from the Trunk panel.

Table 2-6-3 Terminal Trunk Configuration Settings

Field	Description
Terminal identifier	A unique number consisting of digits only. This is the trunk identifier configured on the other IP PBX.
Description	Arbitrary description info.
Terminal password	Password of SIP trunk given on the other IP PBX for authentication.
DID of extension	When enabled DID, choose an extension to be an unconditional destination for incoming calls to this terminal. The number of the terminal is therefore regarded as the direct line number of the extension.
Language	Preferred language for system instructions heard from the terminal.
Usergroup of privilege	When disabled DID, select a usergroup whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this terminal. ☞ There may not be appropriate usergroups to select initially. One can come back later once the expected usergroup has been added.
Bandwidth sensitive	Indicate the trunk is over a bandwidth-sensitive link, e.g. across Internet. If checked, a limit must be specified for call admission.

2.6.4. POTS setup (PC#1, PC#2 only)

This page allows selection of country-based progress tones and/or impedance of POTS ports. IP PBX service needs to be restarted before new setting takes effect.



2.7. Feature

A feature is a logical entity presenting a function module of IP PBX, e.g. meet-me conference, Interactive Voice Response (IVR) subsystem, voice mail, etc. Note that IP PBX service must be reloaded to activate changes.

2.7.1. Call park

During a call, the callee may want to continue the conversation using another phone. The call park feature enables so by letting the callee transfer the call to the call park base extension and IP PBX will respond an available park line from the pool of call park numbers. After that the callee hangs up current phone, moves to another phone, and dials the park line number told by IP PBX to resume conversation with the caller. If the callee does not call the given park line number to resume conversation in 45 seconds, IP PBX will ring the original extension where the callee answered the call. To configure Call Park feature, select Feature -> Call park, enter settings shown in Table 2-7-1, and then click APPLY button.

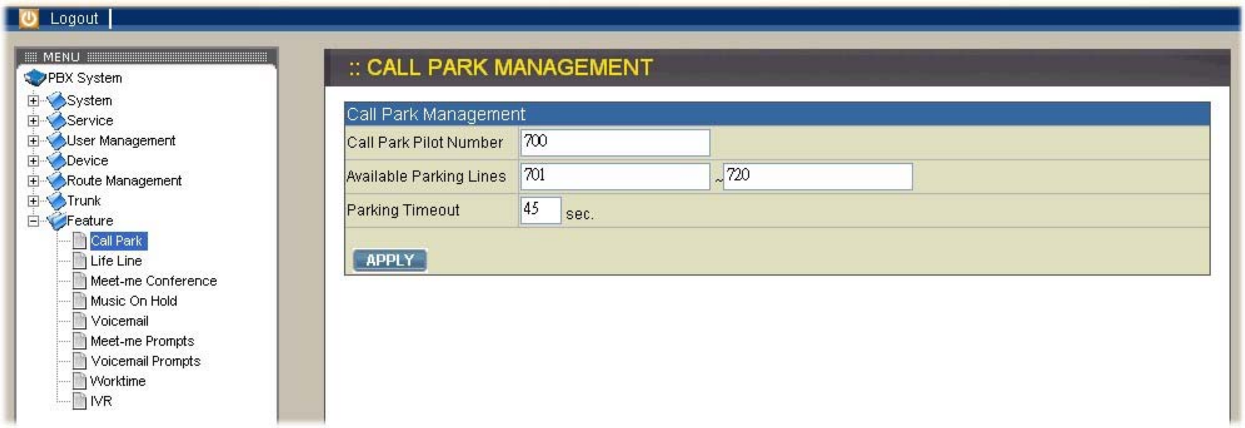
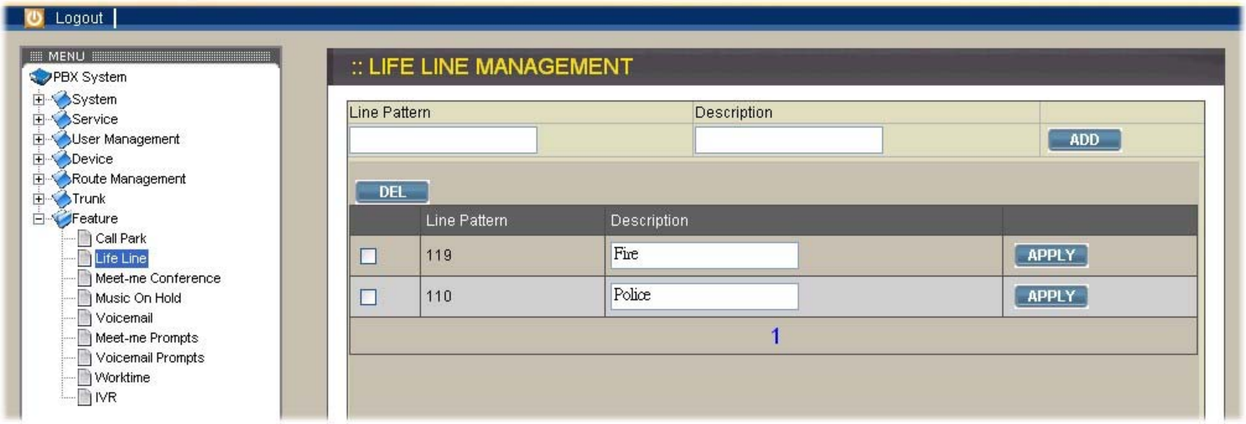


Table 2-7-1 Call Park Configuration Settings

Field	Description
Call park pilot number	A unique extension number for call parking, e.g. 700.
Available parking lines	Extension pool available for call parking, e.g. 701-720 forms a pool available for system to park calls.
Parking timeout	Timeout waiting for picking up the parked call

2.7.2. Life line (PC#1, PC#2 only)

Life line feature allows specification of emergency number patterns to seize a PSTN line with absolute priority. For example, someone dials an emergency call while all PSTN lines are in use. In such case, if the called number matches any specified pattern, the PSTN line with longest talk time so far will be disconnected right away to allow the connection of the emergency call.
Select Feature -> Life Line to configure life-line feature.



Add Life-Line pattern

Click <ADD> button from the left panel to add a new pattern.
Enter settings shown in Table 2-7-2
Click APPLY button.
The newly added pattern should be displayed in Life Line panel on the left side.

Delete Life-Line pattern

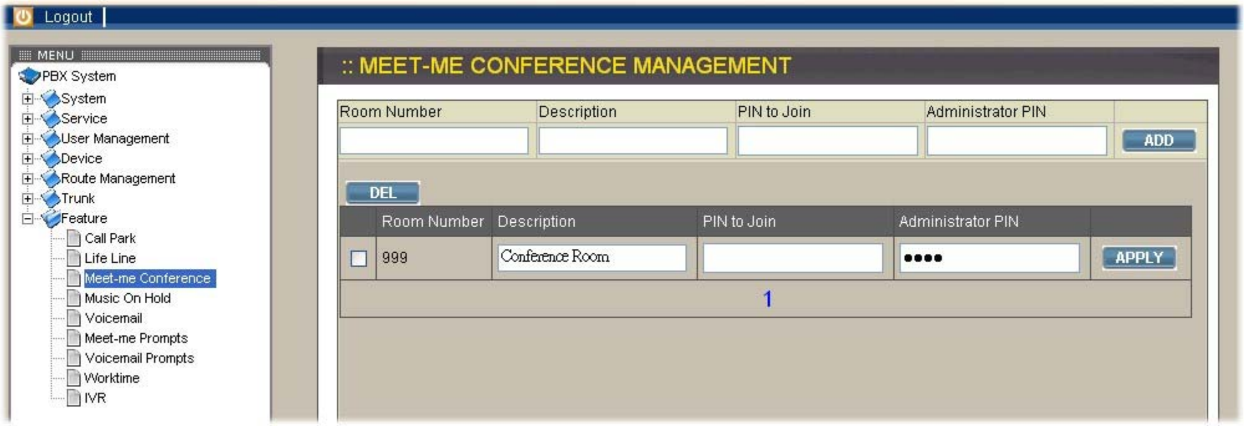
Click link to the line pattern to delete from the Life Line panel.
Click Delete button.
The deleted pattern shall disappear from the Life Line panel.

Table 2-7-2 Life line Configuration Settings

Field	Description
Number pattern	Pattern for emergency numbers. Note: this is the pattern after digit stripping. For example, configure 911 here even if users dial 9911 to reach the 911 service over PSTN when the PSTN trunk has an outbound dialplan of “9.”.
Description	Arbitrary description info.

2.7.3. Meet-Me Conference (PC#1, PC#2, PC#3 only)

Meet-me conference enables conferencing of multiple parties from various sources. A party could dial in a conference from an internal IP phone, an external IP phone on Internet, an analog phone via PSTN, or an IP phone in a remote site. IP PBX allows multiple conference rooms going concurrently using different room numbers. Before entering a meeting room, the caller must enter the correct PIN of the room number. Select Feature -> Meet-me Conference to configure meet-me conference feature.



Add Meet-me Conference

Click <ADD> button from the left panel to add a new conference room.
Enter settings shown in Table 2-7-3
Click APPLY button.
The newly added room should be displayed in Meet-me rooms panel

Edit Meet-me Conference

Click the link of the room to edit from the Meet-me rooms panel.
Edit settings shown on the right side.
Click APPLY button.

Delete Meet-me Conference

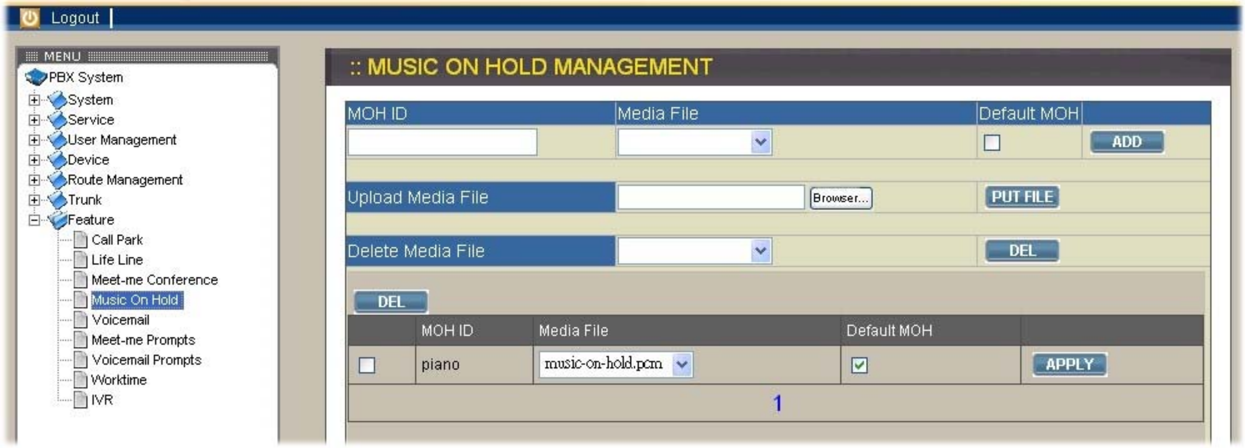
Click link to the room to delete from the Meet-me rooms panel.
Click Delete button.
The deleted conference room shall disappear from the Meet-me rooms panel.

Table 2-7-3 Meet-me Conference Configuration Settings

Field	Description
Room number	Meeting room number, e.g. 8000.
Description	Arbitrary description info.
PIN to join	PIN for normal users to join the conference. During a conference, a normal user has following options: # to quit conference *1 to mute/unmute *9 to log in as the administrator if there is no administrator in yet.
Administrator PIN	PIN for the administrator of the conference. During a conference, the administrator has following options: # to quit conference *1 to mute/unmute *2 to lock/unlock the conference *3 to invite a user into the conference *4 to drop a user from the conference *5 to drop all users in the conference *6 to drop the last invited user by *3 ** to send DTMF digits to the last invited user by *3

2.7.4. Music On Hold

Music-On-Hold (MOH) is used in several occasions for a single purpose—to comfort the waiting party with music. One could upload some candidate music files and pick one as the default one. Select Feature -> Music On Hold to manage MOH files.



Add MOH file
Click <ADD> button from the left panel to add a new MOH file.
Enter settings shown in Table 2-7-4
Click APPLY button.
The newly added file should be displayed in MOH files panel on the left side.

Edit MOH file
Click the link of the file to edit from the MOH files panel.
Edit settings shown on the right side.
Click APPLY button.

Delete MOH file
Click link to the file to delete from the MOH files panel.
Click Delete button.
The deleted MOH file shall disappear from the MOH files panel.

Table 2-7-4 MOH file Configuration Settings

Field	Description
MOH ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Media file	Candidate music files in the repository. To upload a new music file, click Browse to locate a Windows PCM (8000 Hz, 16-bit) file from local host and press Upload. On successful uploading, the filename will appear in the pull-down menu of media files. Similarly, click Remove to remove selected MOH file from the candidate list.
Default MOH	Check to use this music file for system default MOH globally.

2.7.5. Voicemail

IP PBX has a built-in voice mail subsystem with a sophisticated IVR. A call to an extension in use or no answer could be configured to enter voice mail recording procedure. After leaving a message, a notification e-mail will be sent to the user owns the extension with or without the message in the form of an attached .wav file. The Message Waiting Indicator (MWI) on phone (if any) will be lit. The user could then dial the voicemail pilot number to enter voice mail system to manage messages such as playback, delete, or move them from inbox to different folders.

To configure Voicemail feature, select Feature -> Voicemail from the menu, enter settings shown in Table 2-7-5, and then click APPLY button.

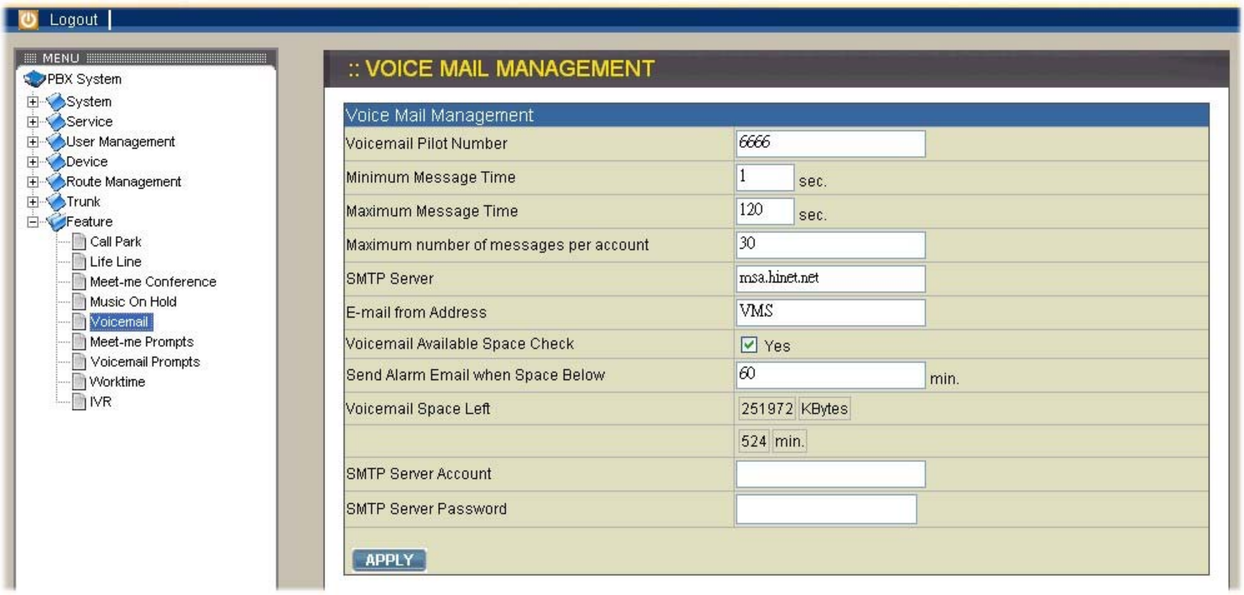


Table 2-7-5 Voice Mail Configuration Settings

Field	Description
Voicemail pilot number	Number to access voice mail system IVR.
Minimum message time	Message less than this duration will be discarded. E.g., 3 (sec).
Maximum message time	Maximum duration allowed for a single message. E.g., 60 (sec).
Maximum messages per account	Maximum number of messages allowed per extension.
SMTP server	Hostname or IP address of the SMTP server for voicemail notification.
E-mail from address	Most SMTP servers require a valid from address to accept a mailing request.
SMTP server account	Specify account ID if the SMTP server requires authentication for outgoing mails.
SMTP server password	Specify account password if the SMTP server requires authentication for outgoing mails.

2.7.6. Meet-Me Prompts (PC#1, PC#2, PC#3 only)

This page allows replacing built-in meet-me conference prompts with user recordings. Choose a language and browse a corresponding recording from local storage. Then, click PUT FILE to complete the replacement. To reset a prompt back to default, leave browsed file blank and directly click the PUT FILE button. Note that the replacement is done for the selected language only. Currently only following prompts could be replaced.

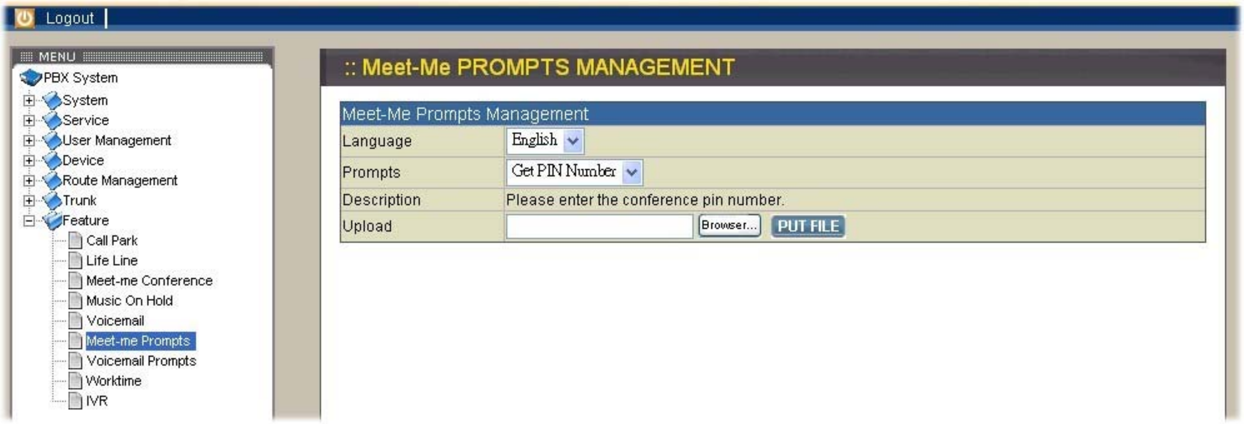
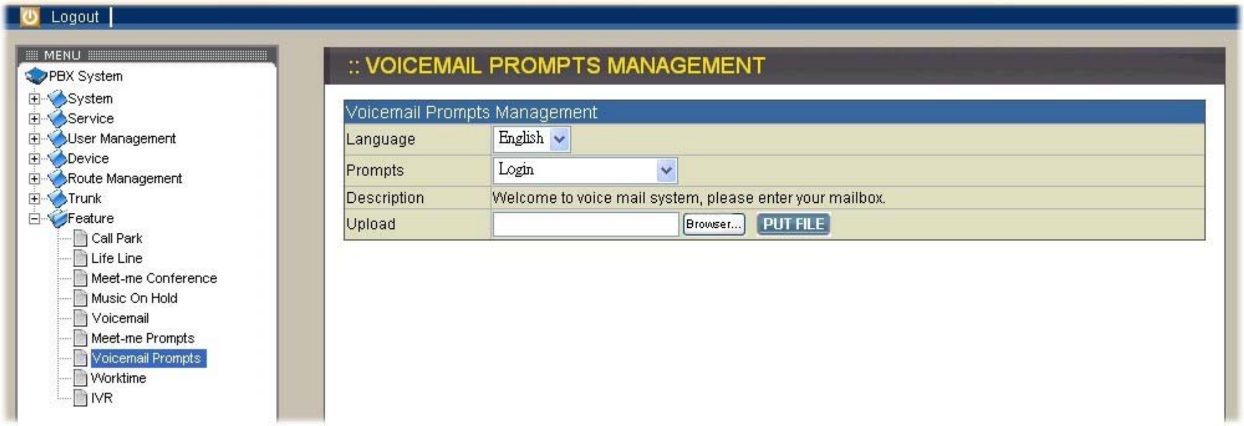


Table 2-7-6 Replaceable Meet-me Prompts

Prompt	Description
Get PIN number	Please enter the conference pin number.
Invalid PIN	That pin is invalid for this conference.
Only person	You are currently the only person in this conference.

2.7.7. Voicemail Prompts

This page allows replacing built-in voicemail system prompts with user recordings. Choose a language and browse a corresponding recording from local storage. Then, click PUT FILE to complete the replacement. To reset a prompt back to default, leave browsed file blank and directly click the PUT FILE button. Note that the replacement is done for the selected language only. Currently only following prompts could be replaced.



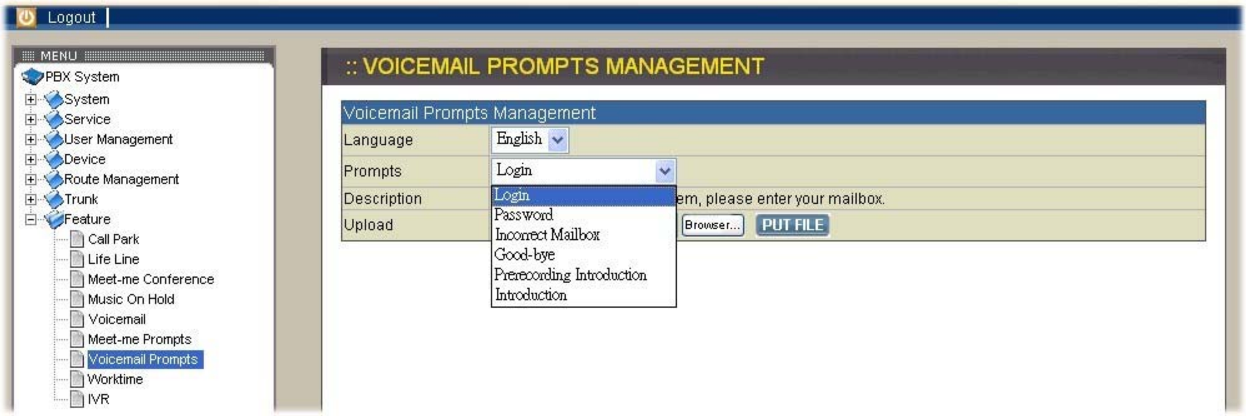


Table 2-7-7 Replaceable Voicemail System Prompts

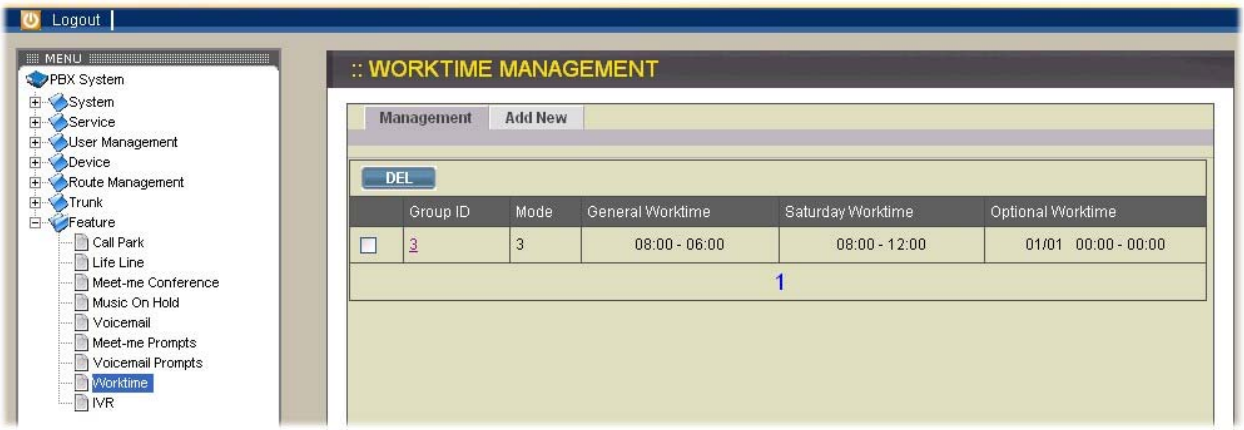
Prompt	Description
Login	Welcome to voice mail system, please enter your mailbox.
Password	Password?
Incorrect mailbox	Login incorrect, mailbox?
Good-bye	Good-bye.
Prerec-intro	Press star(*) to cancel recording and return to the main menu. Or, press pound(#) to start recording right away.
Intro	Please leave your message after the tone. When done, hang up or press the pound(#) key.

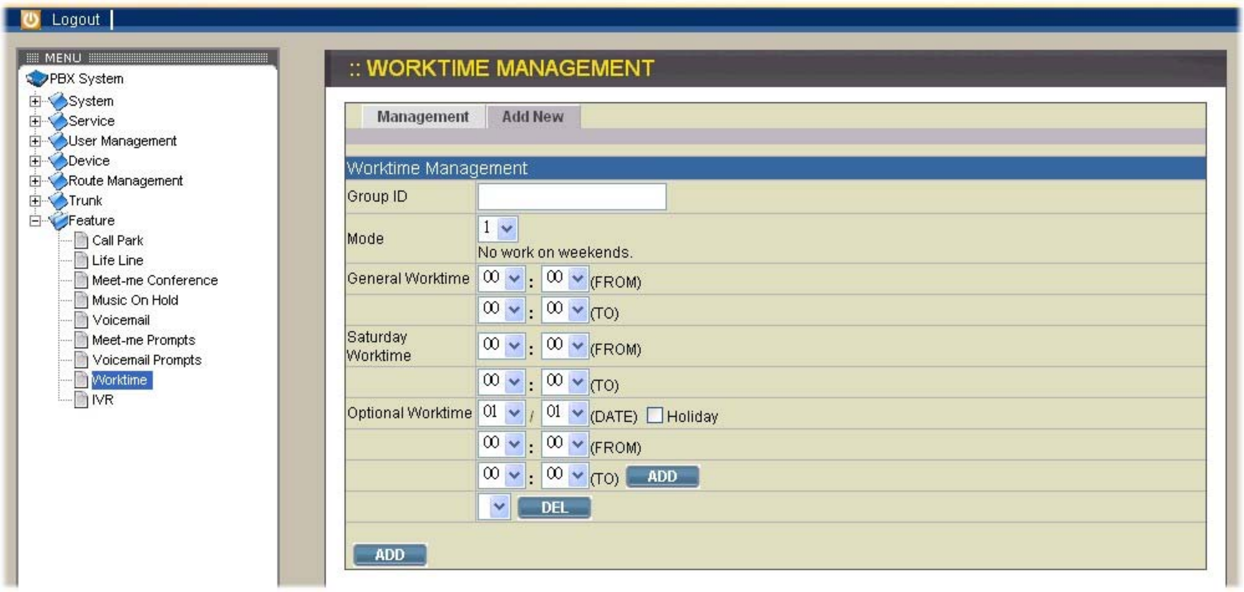
2.7.8. Worktime

IP PBX has a built-in worktime system for users in different places. To configure Worktime feature, select Feature -> Worktime from the menu, enter settings shown in the window, and then click APPLY button.

Add Worktime

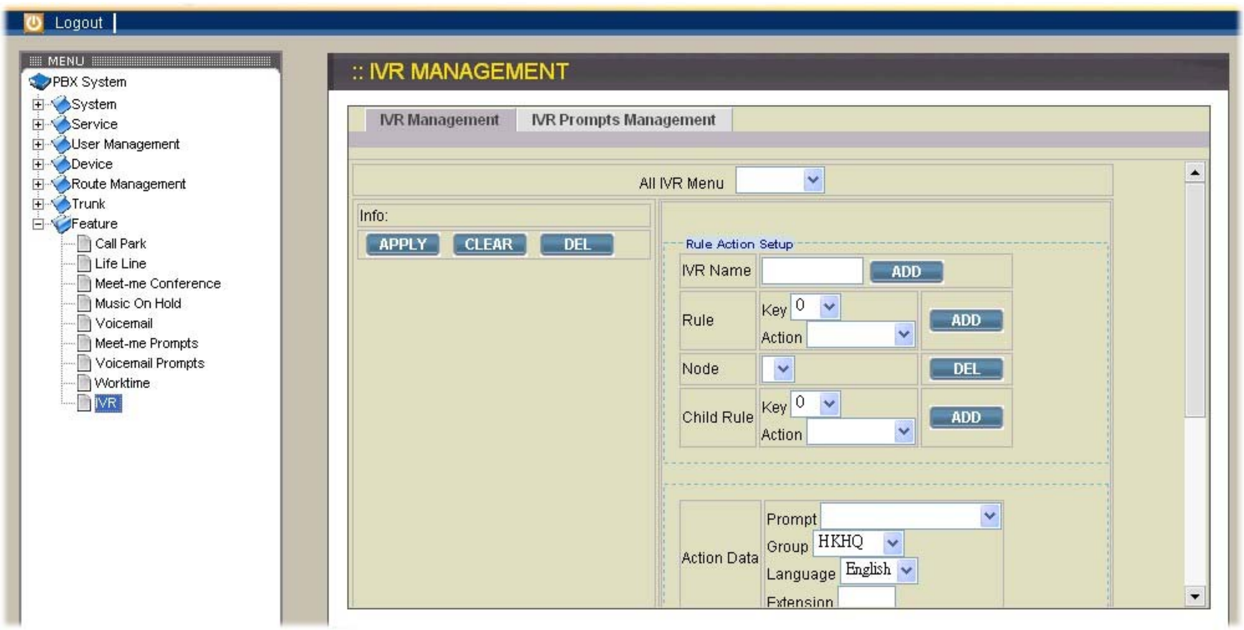
Click <Add New> button from the left panel to add a new worktime.
Enter settings shown in the window.
Click ADD button.
The newly added worktime shall be displayed in Worktime panel.





2.7.9. IVR

Interactive Voice Response (IVR) answers incoming calls to a trunk and prompt voice menu to guide the caller to reach his expected extension. Usually the caller dial the desired extension by pressing digit keys from the touch-tone phone, and the auto attendant to make the transfer. In some case, the caller may not know the extension of the party so that there should be a key, say 0, to call the operator for help. Sometimes, the incoming call may stay on the line for some time without giving a valid extension, IP PBX will transfer the call to the operator as well. Select Feature -> IVR to configure IVR feature. Enter settings shown in Table 2-7-9, and click APPLY button.



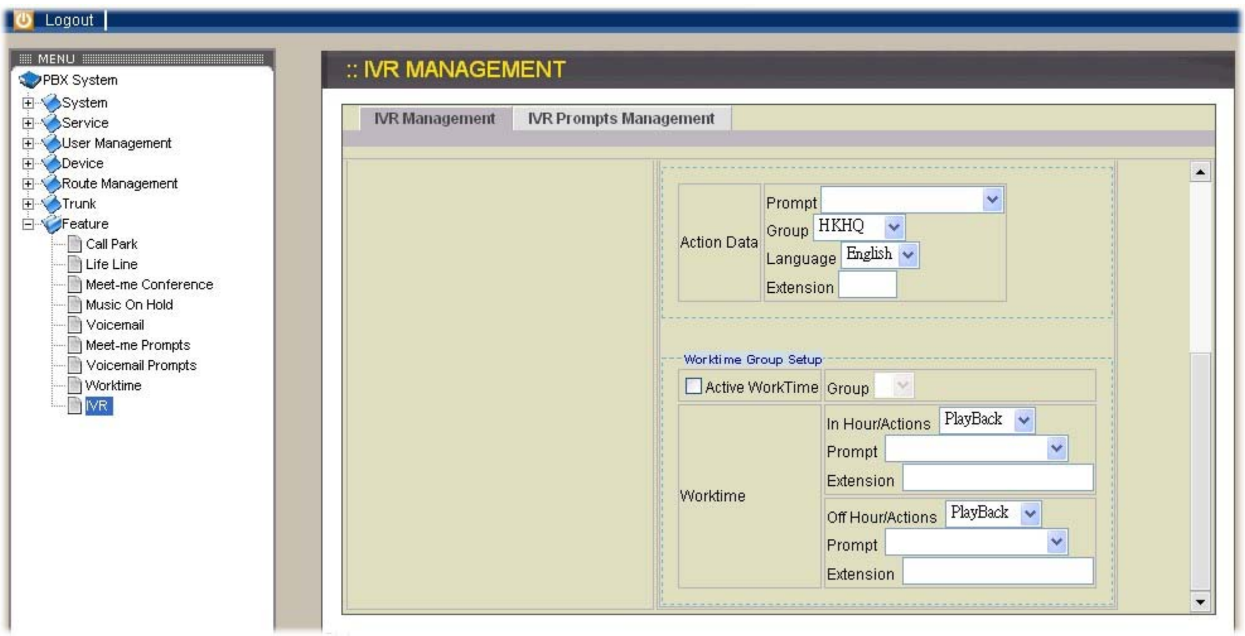


Table 2-7-9 IVR Settings

Field	Description
Key for reaching operator	Select key for reaching operator if this option is checked, e.g. 0.
Operator extension	The designated extension number of the operator.
Action	Select an action to be taken if the caller does not respond to the IVR for a certain period of time. Option "disconnect" means to hang up calls when timeout, while option "operator" transfers calls to operator when timeout.
Digit input timeout	Enter timeout for digit collection, e.g. 5 sec.
User response timeout	Enter timeout for caller response, e.g. 15 sec.

IVR Prompts Management

This page allows replacing built-in IVR prompts with user recordings. Choose a language and browse a corresponding recording from local storage. Then, click PUT FILE to complete the replacement. To reset a prompt back to default, leave browsed file blank and directly click the PUT FILE button. Note that the replacement is done for the selected language only. Currently only following prompts could be replaced. The recording format must be 8000 Hz, 16 bit, Windows PCM .wav file.

Table 2-7-10 Replaceable IVR Prompts

Prompt	Description
Greeting	Welcome to ${company}, please dial an extension or press ${key} for the operator.
Invalid	I am sorry, that is not a valid extension. Please try again.
Extension	Extension
Unavailable	is not available
Busy	is on the phone

3. Application Examples

This chapter provides configuration examples for IP PBX deployment. The internal extension can serve as a short demonstration for IP PBX. In addition, two more cases for IP PBX applications will be shown. The first case is for Single-Site configurations, and the second case for Two-Site configurations. The settings are configured in sequence for usergroups, routegroups, and trunks. These examples are very much flexible and scalable enough to support various network architectures for IP PBX. Users could refer to these examples and build a larger network involving multiple sites and advanced services.

Internal Extension Configuration
The procedure below is recommended as the first step of a configuration sequence. The configuration only enables internal extension calls, but it serves as a good practice for IP PBX administration.

Configuration steps:

1. Create and configure a usergroup named ALL (refer to Usergroup).
2. Create and configure users and assign to usergroup ALL (refer to User).
3. Configure devices and an extension for each device (refer to Device).
4. Assign each extension to a corresponding user (refer to Device)
5. Configure each client phone with respect to the extension number and password in its IP PBX extension configuration accordingly.
6. Reload IP PBX service (refer to IP PBX Service).

Up to this point all configured phones should register with the IP PBX with a usable extension. Since these phones are all belonged to the same usergroup ALL, they can call one another without limitation.

3.1. Case I: Single-Site IP PBX

This case shows typical settings of a single-site configuration for SMB Companys. The IP PBX combines the telephony network and the data network with ADSL/Cable modem connection to Internet access and 2 PSTN subscriber lines as shown in Figure 5-1. The provisioning tasks include:

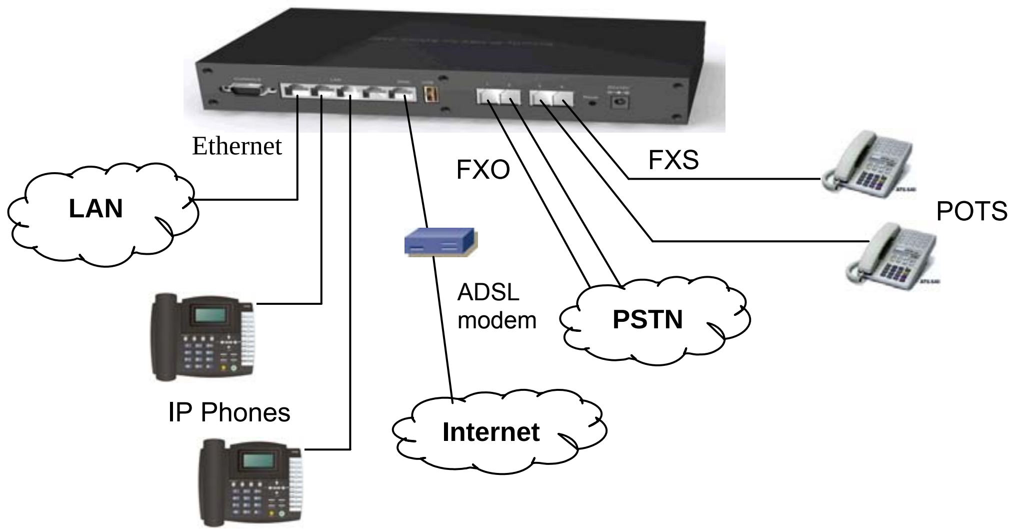


Figure 5-1 Single-Site IP PBX Network Connections

- There are staff phones in cubes and offices and utility phones in public areas.
- Staff and utility phones can be either IP phones or analog phones (POTS).
- Each phone has one extension and it can call any extension without limitation.
- Only staff phones can call out to PSTN with a prefix 9.
- Incoming PSTN calls are answered by auto attendant and could be transferred to any extension.

Configuration steps:

1. Create usergroups staff, utility, and ext-all.
2. Include staff and utility in the Reachable usergroup of ext-all.
3. Create a user account for each staff and assign it to usergroup staff.
4. Create an additional user account public and assign it to usergroup utility.
5. Create a device for each physical phone and designate an extension.
6. Assign extensions of staff phones to corresponding users.
7. Assign all extensions of utility phones to share the same user public.
8. Create a route pstn with pattern 9. with number of digits stripped 1, no prefix.
9. Create a routegroup pstn-out containing route pstn only.
10. Create a PSTN trunk with ID 1, port 1-2, choose pstn-out as Outbound routegroup, leave DID unchecked, and select ext-all as the Usergroup of privilege.
11. Return to usergroup configuration. For usergroup staff, include PSTN trunk pstn1 and Reachable usergroup utility; while for usergroup utility, include Reachable usergroup staff only.
12. Reload IP PBX Service.

3.2. Case II: Two-Site IP PBX

This case describes the typical settings of a two-site network for company B headquarters B-HQ and its branch B-BR in another country. Assuming each site has a DSL connection for Internet access. B-HQ has 4 PSTN subscriber lines and B-BR has 2 lines as shown in Figure 5-2. The provisioning tasks include:

- Both sites have staff phones in cubes and offices and utility phones in public areas.
- Each phone has one extension. A utility phones can call extensions within the site it is in only, while the staff phones can call any extension in both sites without limitation. B-HQ has extensions 1XX and B-BR has extensions 2XX.
- Calls between B-HQ and B-BR use private SIP trunks across Internet. IP PBX at B-HQ has a static IP address 64.1.0.1 and IP PBX at B-BR has a static IP address 222.44.0.1.
- Only staff phones can call out to PSTN with a prefix 9.
- B-HQ staff phones call out starting with 90118621 will relay to B-BR through the SIP trunk and then hop off to PSTN in B-BR. Similarly, B-BR staff phones call out starting with 9001408 will relay to B-HQ for PSTN hop-off.
- Incoming PSTN calls are answered by auto attendant and could be transferred to any extension.

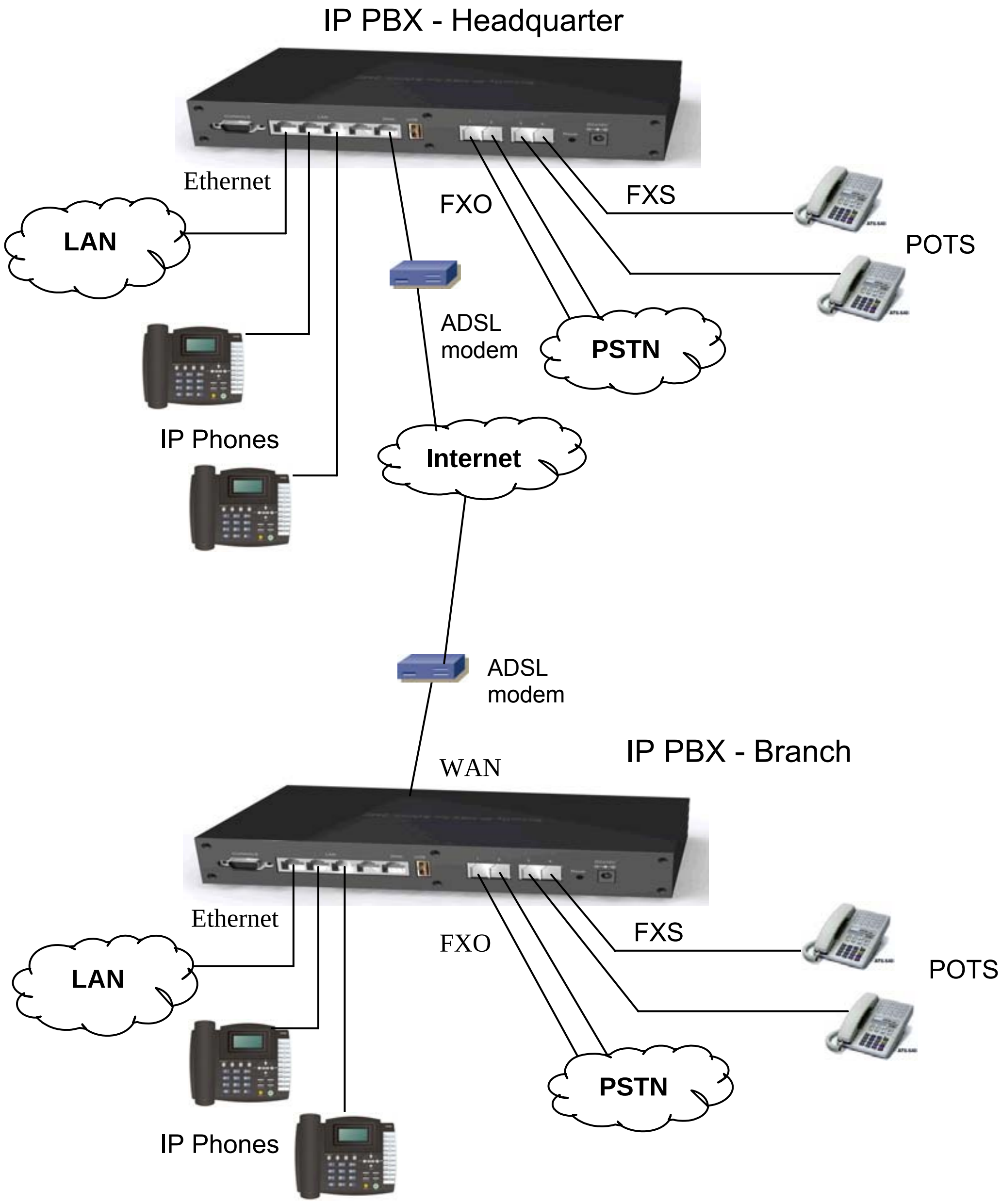


Figure 5-2 Two-Site IP PBX Network Connections

Configuration steps in IP PBX-Headquarter:

1. Create usergroups staff, utility, and ext-all.
2. Include staff and utility in the Reachable usergroup of ext-all.
3. Create a user account for each staff and assign it to usergroup staff.
4. Create an additional user account public and assign it to usergroup utility.
5. Create a device for each physical phone and designate an extension.
6. Assign extensions of staff phones to corresponding users.
7. Assign all extensions of utility phones to share the same user public.
8. Create a route pstn with pattern 9Z. with number of digits stripped 1, no prefix.
9. Create a route pstn-br with pattern 90118621. with number of digits stripped 8, prefix 9.
10. Create a route ext-br with pattern 2XX and number of digits stripped 0, no prefix.
11. Create a routegroup pstn-out containing route pstn only.
12. Create a routegroup to-br containing routes pstn-br and ext-br.
13. Create a PSTN trunk with ID 1, port 1-4, choose pstn-out as Outbound routegroup, leave DID unchecked, and select ext-all as the Usergroup of privilege.
14. Create a dynamic peer SIP trunk with ID 100; password hq-secret; choose to-br as Outbound routegroup, leave DID unchecked, and select staff as the Usergroup of privilege.
15. Return to usergroup configuration. For usergroup staff, include SIP trunk 100, PSTN trunk pstn1 and Reachable usergroup utility; while for usergroup utility, include Reachable usergroup staff only.
16. Reload IP PBX Service.

Configuration steps in IP PBX-Branch:

1. Create usergroups staff, utility, and ext-all.
2. Include staff and utility in the Reachable usergroup of ext-all.
3. Create a user account for each staff and assign it to usergroup staff.
4. Create an additional user account public and assign it to usergroup utility.
5. Create a device for each physical phone and designate an extension.
6. Assign extensions of staff phones to corresponding users.
7. Assign all extensions of utility phones to share the same user public.
8. Create a route pstn with pattern 9Z. with number of digits stripped 1, no prefix.
9. Create a route pstn-hq with pattern 9001408. with number of digits stripped 7, prefix 9.
10. Create a route ext-hq with pattern 1XX and number of digits stripped 0, no prefix.
11. Create a routegroup pstn-out containing route pstn only.
12. Create a routegroup to-hq containing routes pstn-hq and ext-hq.
13. Create a PSTN trunk with ID 1, port 1-2, choose pstn-out as Outbound routegroup, leave DID unchecked, and select ext-all as the Usergroup of privilege.
14. Create a SIP trunk with ID 100 pointing to 64.1.0.1 port 5060; password hq-secret; choose to-hq as Outbound routegroup, leave DID unchecked, and select staff as the Usergroup of privilege.
15. Return to usergroup configuration. For usergroup staff, include SIP trunk 100, PSTN trunk pstn1 and Reachable usergroup utility; while for usergroup utility, include Reachable usergroup staff only.
16. Reload IP PBX Service.

4. Appendix - Specification

Supported Standards

RFC 3261, RFC 3311, RFC 3515
RFC 3265, RFC 3892, RFC 3361
RFC 3842, RFC 3389, RFC 3489
RFC 3428, RFC 2327, RFC 2833
RFC 2976, RFC 3263, RFC 3264

SIP Registrar

- Static/Dynamic registration
- Configurable expiry time
- MD5 authentication
- Registration proxy to external registrars
- Configurable PBX Caller ID
- User profile
- Handle loose RFC-compliant phones
- Resilient message retry mechanism
- Seeding historical registrations

SIP Proxy

- Proxy server
- Call-based MD5 authentication
- NAT traversal for clients
- Outbound proxy with or without WAN
- Inter-proxy call hand-off

PBX Features

- Support call hold, call waiting, 3-way call conference with feature phones
- Built-in in-line call transfer
- Unconditional, unavailable, busy call forward
- Per-calling-number forward and rejection
- Group-based call pick-up
- Call-parking
- Multi-room meet-me conference
- Auto-attendant
- Voice mail system
- Call privilege grouping
- FXO interface for PSTN Inbound/outbound
- FXO disconnection tone detection
- FXO hunt group
- Caller ID detection
- Echo cancellation
- In-band/RFC2833/SIP-INFO DTMF translation
- Support 5 SIP trunk
- Intra-PBX stackable trunking over Ethernet
- FWD/Vonage account sharing for extensions
- Interoperable with Cisco CallManager, CCME, IOS SIP gateway, Unity CUE, 79XX, ATA
- Call admission control for wired/wireless phones
- Music on hold
- Direct line
- Outbound 900/0204 blocking

Auto Attendant

- Configurable Greeting
- Key to reach operator
- Timeout interval and timeout action
- Music on ringing extensions
- Forward to voice mail on no-answer

Voice Mail

- User PIN
- Multi-folder archive
- Fast-forward/Rewind/Undelete
- MWI notification
- E-mail notification and attachment(Unified messaging)
- Personal reception on unavailability and busy
- Voicemail forwarding
- Reply call or new call in voicemail menu

Storage

- Support USB 2.0 storage media

Meet-me Conference

- Multiple rooms with configurable number and PIN
- Music on first dial-in party
- Hot key to leave conference

NAT

- Auto NAT discovery and traversal
- Built-in STUN client
- RTP proxy
- RTP port range designation

Relational Provision

- Logical partition/relation on users and trunks
- Logical provision on outgoing and incoming calling search scopes
- Rich dial-plan expressiveness through route patterns
- Object-oriented provisioning paradigm

Administration

- Web-based configuration
- Flat system event syslog
- Flat Call Detail Record (CDR)
- Extension status display
- TFTP server and TFTP repository maintenance
- Network Time Protocol time synchronization
- Real Time Clock setting
- DHCP server with multiple partitions, per-MAC IP binding, list of options
- Configurable time zone
- Firmware upgrade through Web interface

Network Management

- DHCP/PPPoE/Static IP on WAN
- LAN IP and netmask specification
- Firewalling on predefined services
- NAT for outbound traffic from LAN
- DNS and dynamic DNS
- QoS queuing mechanism for VoIP and data traffic

Maximum Capacity

- 30 extension registrations
- 30 voicemail accounts
- 10 concurrent calls with RTP
- 4 analog calls

Hardware

- Intel IXP425BD at 533 MHz
- 16MB Flash, 64MB SDRAM
- On board DSP chip AC49008
- Two USB2.0 ports for a storage space.
- 4-Port LAN, 1-Port WAN
- 2-FXO ports, 2-FXS ports
- G.723.1, G.729, G.711 (μ/A-Law), G.726 ADPCM
- Caller ID : Bellcore and ESTI
- Fax support: T.38 compliant
- Echo canceller: G.168-2002 compliant

5. GENERAL PUBLIC LICENSE

This product incorporates open source code into the software and therefore falls under the guidelines governed by the General Public License (GPL) agreement. Adhering to the GPL requirements, the open source code and open source license for the source code are available for free download at http://global.level1.com.
If you would like a copy of the GPL or other open source code in this software on a physical CD medium, LevelOne (Digital Data Communications) offers to mail this CD to you upon request, for a price of US$9.99 plus the cost of shipping.