



**Internet Telephony PBX System
IPX-2000
User's Manual**

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Revision

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1 Introduction

1.1 Overview

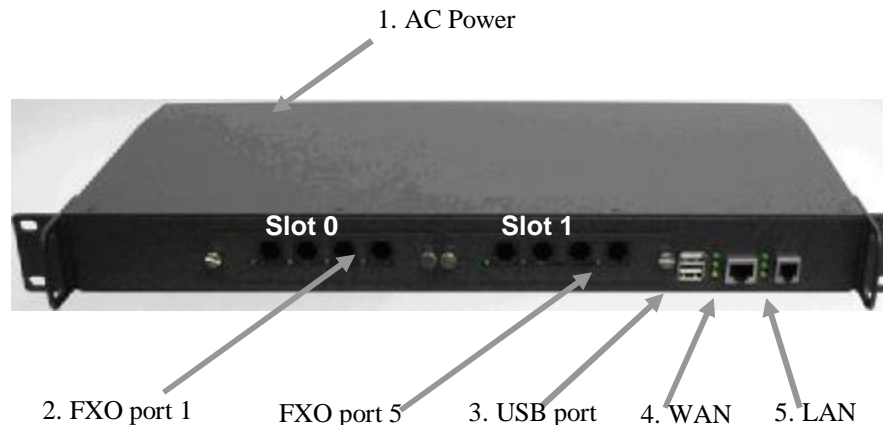
The *IP PBX Administration Guide* provides instructions for administering the IP PBX system. IP PBX is an embedded call-processing server communicating with client stations with Session Initiation Protocol (SIP). It migrates the telephony network and the data network of a small-to-medium business (SMB) company into a manageable converged network. IP PBX works with various IP phones (desktop, WiFi, Bluetooth, and DECT), voice-over-IP (VoIP) gateways, and analog telephone adapters to route calls among client phones, analog phones, and PSTN network. Additional voice features such as conferencing, auto attendant, and voicemail are seamlessly enabled to all phones. IP PBX also provides Internet access to all LAN devices through Network Address Translation (NAT).

IP PBX provides call control and media relay services to SIP clients and applications. It performs the following primary functions:

- SIP Registrar
- SIP Outbound Proxy with media relay
- SIP Gateway (FXO)
- SIP PBX for extension calls
- Auto attendant IVR
- Voice mail IVR
- Meet-me conference

IP PBX has a built-in suite of voice applications for supplementary services and therefore no special-purpose hardware is required. This means the total cost of ownership of a converged network enabled by IP PBX is lower than building separated infrastructures for legacy telephony network and data network. Besides, it comes with a web-browsable interface to the data network configuration and voice service provisioning, which brings the manageability of both networks together to facilitate administration locally or remotely.

1.2 Installation



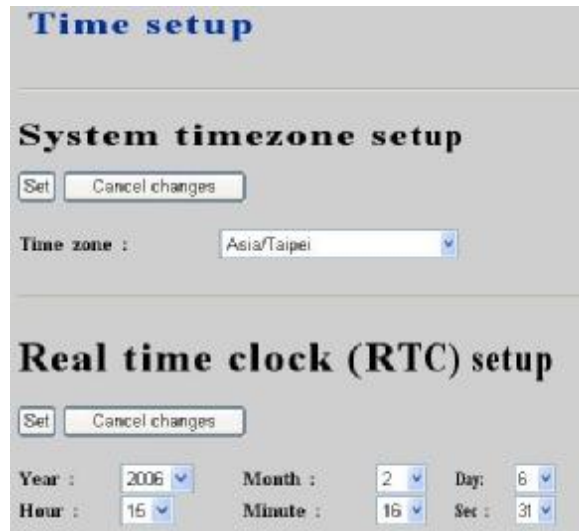
1. AC Power	110/220 Volt, 60 Hz
2. FXO ports	4 ports per daughter card, numbered from right to the left. The rightmost port at slot 0 is port 1 and the leftmost port at slot 1 (if installed) is port 8. FXO ports are to be connected to FXS jacks on wall or analog PBX using RJ-11 cables.
3. USB ports	2 external ports with compliance to USB 1.1/2.0. Plug in a USB hard drive for CDR/voicemail backup from the internal storage.
4. WAN	Connect to a broadband modem or a WAN router.
5. LAN	Connect to a LAN switch.

2 System Configuration

This section describes how to configure system parameters used by IP PBX. The factory default of LAN IP address is 192.168.1.1. Connect to LAN port and the configuration Web interface could be reached at <https://192.168.1.1/>. Once connected, the browser will warn about accepting a certificate. Please accept it. Then, give default administrator ID and password (both are *admin*) to log in for administration. The administrator password could be changed in the User Management page under user ID admin.

2.1 Time setup

The Time Setup page allows administrator to configure time zone and date for IP PBX. Select **System -> Time setup**, current setting of time zone and date is displayed.



Time setup

System timezone setup

Set Cancel changes

Time zone : Asia/Taipei

Real time clock (RTC) setup

Set Cancel changes

Year : 2006 Month : 2 Day : 6

Hour : 15 Minute : 16 Sec : 31

2.1.1 Time Zone Setup

Choose time zone in drop-down list, then click **Set** button.

2.1.2 Date Setup

Choose year, month, day, hour, minute, and sec in drop-down lists respectively, then click **Set** button.

2.2 WAN setup

The WAN Setup page allows administrator to configure WAN network interface for IP PBX. Select **System -> WAN setup**, current setting of WAN network interface is displayed, e.g. type, IP address etc. Unless **LAN only** checkbox is checked, you can choose one of the three options for your configuration, Static, DHCP, and PPPoE.

WAN setup

Save Cancel Changes

Type : STATIC IP

LAN only

Interface MAC : 00304F123456

IP address: 192.168.0.1

Netmask: 255.255.255.0

Gateway: 192.168.0.254

DNS 1: 168.95.1.1

DNS 2: 168.95.192.1

2.2.1 Static

You can choose Static IP in **Type** drop-down list, and manually configure the following information:

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

And, click **Save** button to submit.

2.2.2 DHCP

Simply choose DHCP in **Type** drop-down list, and then click **Save** button. The acquired IP address, network mask, and default gateway information will be displayed when revisit this page later.

2.2.3 PPPoE

Choose PPPoE in **Type** drop-down list, enter username and password, and then click **Save** button. The PPPoE dialing will start right away. When there is an active connection, 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.

2.2.4 Lan only

Check **Lan only** to disable WAN IP setting but allow the configuration of default gateway and primary/secondary DNS servers.

2.3 LAN setup

The LAN Setup page allows administrator to configure LAN network interface for IP PBX. Select **System** -> **LAN setup**, current settings of LAN network interface are displayed. You can modify them by entering new IP address and network mask followed by clicking **Save** button. Note that by default IP PBX grants IP addresses to LAN devices via DHCP 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.

LAN setup

Save Cancel changes

Interface MAC : 00304F654321

IP address : 192.168.1.1

Net mask : 255.255.255.0

2.4 LAN Routing

To enable static routing among LAN subnets, enter network info and the IP address of the corresponding gateway in IP PBX's LAN. It is important to assure that the given gateway IP address sits in the IP PBX's LAN. Each subnet requires an entry even multiple subnets share the same gateway, unless masking does the same. 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 **Save** button, however, IP PBX service needs to be restarted to regard calls from designated LAN subnets as LAN traffic.

Static LAN routing management

Save Cancel changes Delete

IP Routes
<Add new >

IP Route ID :

Subnet : . . .

Netmask : . . .

Gateway : . . .

2.5 DynDNS setup

Dynamic WAN IP address causes difficulty for inbound connections from remote clients or IP PBX systems. A popular work-around is to adopt domain names provided by DynDNS and run a client on or behind the gateway router (or IP PBX). It is required to apply an account and create a hostname in the account before configuration. Select **Enable** and give account info 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 DynDNS in **Interval** seconds periodically.

Dynamic DNS client

Enable Disable

Username :

Password :

Hostname :

Interval : (sec)

2.5.1 Enable DynDNS

Select **Enable** radio button, enter the **Username**, **Password**, **Hostname**, and **Interval**, and then click **Save** button. Typical hostname has a form of `<hostname>.dyndns.org`. The refresh interval is usually between 60 – 600 seconds depending on the volatility of WAN IP assignment.

2.5.2 Disable DynDNS

Select **Disable** radio button, and then click **Save** button.

2.6 QoS setup

To assure the bandwidth reserved for the outgoing VoIP traffic over regular data traffic from LAN, QoS Setup page offers three parameters to characterize the WAN link. By default QoS is disabled since these parameters must be correctly given according to the actual WAN speed.

Network QoS setup

Enable Disable

WAN uplink speed : kbps

WAN downlink speed : kbps

Uplink VoIP reserved : kbps

2.6.1 Enable QoS

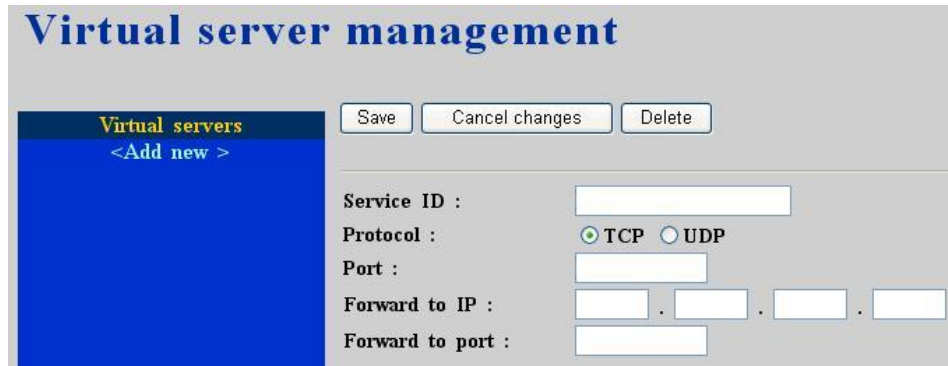
Select **Enable** radio button, enter the **WAN uplink speed**, **WAN downlink speed**, and **Uplink VoIP reserved** (bandwidth), and then click **Save** button. For a popular 2M/256K ADSL program, the WAN uplink speed would be 256 and the WAN downlink speed would be 2048 since the unit is in kbps. The Uplink VoIP reserved could be, say, 192 out of the total 256 kbps to allow 2 concurrent G.711 calls.

2.6.2 Disable QoS

Select **Disable** radio button, and then click **Save** button.

2.7 Virtual server

To enable access servers in LAN from a machine beyond WAN, select **System -> Virtual server** to configure port mappings. **Service ID** names the service. **Protocol** and **Port** specify the TCP/UDP port number on WAN IP to be forwarded to the **Forward to port** of **Forward to IP** in LAN. Say 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.



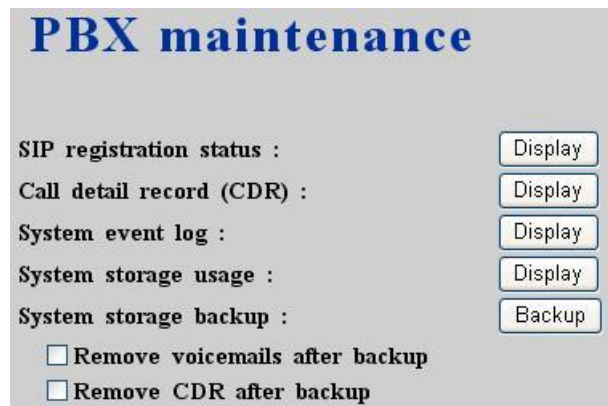
The screenshot shows the 'Virtual server management' web interface. At the top, there are three buttons: 'Save', 'Cancel changes', and 'Delete'. Below these is a section titled 'Virtual servers' with a '<Add new >' link. The main area contains several form fields: 'Service ID' (text input), 'Protocol' (radio buttons for TCP and UDP, with TCP selected), 'Port' (text input), 'Forward to IP' (four text inputs separated by dots), and 'Forward to port' (text input).

2.8 Maintenance

This page includes maintenance functions of IP PBX, from status display to storage backup, etc.

2.8.1 SIP registration status

This Display button lists registration status of each client and remote IP PBX and the IP/Port from where they register. SIP trunk registrations, if any, are also shown in the end of the list. Column **Dynamic** stands for whether the listed IP address is dynamic or specified. **Reg. Progress** is the response code and message if registration has been attempted but not successful so far. **Slave Registrar** column is used only under the stackable mode. It indicates with which slave box a SIP client is registered. Blank means a client is registered with the master box.



The screenshot shows the 'PBX maintenance' web interface. It features a list of maintenance functions, each with a corresponding button: 'SIP registration status' (Display), 'Call detail record (CDR)' (Display), 'System event log' (Display), 'System storage usage' (Display), and 'System storage backup' (Backup). Below these are two checkboxes: 'Remove voicemails after backup' and 'Remove CDR after backup', both of which are currently unchecked.

2.8.2 Call Detail Record

CDR details each call record including calling and called numbers, channel (trunk if outbound) in use, epochs when the call was made, answered, and ended, which yield the total and billable durations. The last column denotes the disposition of a call like answered or not.

2.8.3 System event log

Event log includes reported events from following system services: NTP, DNS, DHCP, and PPPoE.

2.8.4 System storage usage

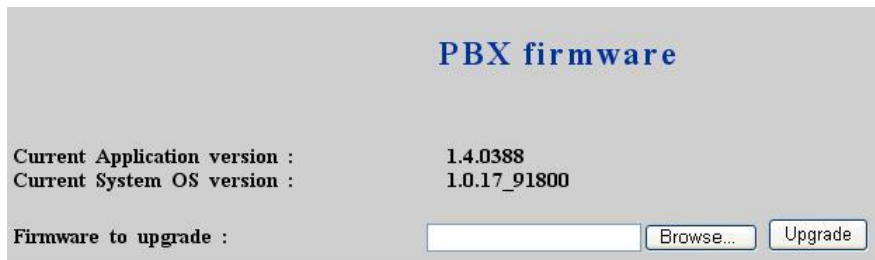
This displays the size and utilization percentage of the internal main system storage.

2.8.5 System storage backup

To back up internal main storage, click the Backup button and follow instructions to insert the USB connector of an external USB drive. Options include whether to keep or remove CDR and/or voicemails after backup. After a confirmation of the insertion, back starts a few seconds later if the external USB drive is accessible and found enough available space. On a successful backup, the name of the new folder created on the external drive will be displayed. Either the backup is successful or failed, a user **must remove** the USB connector of the external drive.

2.9 Firmware upgrade

The version of the running PBX firmware could be found in **System -> PBX firmware**. To upgrade current firmware, locate a release file obtained from the vendor, and click **Upgrade**. Note that the filename of firmware should not be changed, otherwise system will refuse to upgrade with it.



2.10 Logout

Log out administrator and close browser windows.

2.11 Shutdown

By selecting **System -> Shutdown**, you can shutdown or reboot the machine after clicking the **Yes** button in this page. In case such software reboot fails, press and release the hardware reset button quickly will do the same.



3 Service Configuration

This section describes details to configure various services built in the IP PBX.

3.1 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.



NTP

Enable Disable

NTP server FQDN or IP address:

3.1.1 Enable NTP service

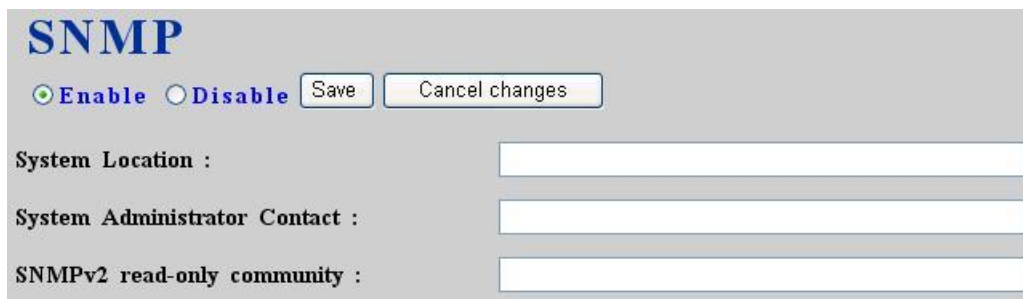
Select **Enable** radio button, enter the fully qualified domain name or the IP address of a NTP server, and then click **Save** button.

3.1.2 Disable NTP service

Select **Disable** radio button, and then click **Save** button.

3.2 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.



SNMP

Enable Disable

System Location :

System Administrator Contact :

SNMPv2 read-only community :

3.2.1 Enable SNMP service


Select **Enable** radio button, enter **System Location**, **Administrator Contact**, **read-only community**, and then click **Save** button. Example settings are *Taipei Office*, *John Doe admin@foobar.com*, and *public* for the three fields.

3.2.2 Disable SNMP service

Select **Disable** radio button, and then click **Save** button.

3.3 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.



3.3.1 Enable STUN service

Select **Enable** radio button, enter the fully qualified domain name or the IP address of a STUN server, and then click **Save** button. IP PBX service needs to be reloaded to reflect the change.

3.3.2 Disable STUN service

Select **Disable** radio button, enter the fully qualified domain name or the static IP address of the external WAN interface and then click **Save** button. Usually this address refers to the static WAN IP address if there is a NAT device between the IP PBX and the Internet. If the WAN port of IP PBX directly connects to Internet or it is unused, leave the address blank.

3.4 TFTP service

Select **Service** -> **TFTP service**, this page indicates current status of TFTP service. You can enable or disable TFTP service at any time.



3.4.1 Enable TFTP service

Select **Enable** radio button, then click **Save** button. Afterwards, you are able to do file management, e.g. upload and download files to and from IP PBX through TFTP service.

3.4.1.1 Change Directory

- Choose directory in **Directory** drop-down list.

By default, the root directory is /tftpboot. Initially, you might not be able to change directory, since there is no folder created under /tftpboot yet.

F Current directory is shown in the gray field on the right side, for instance, it is /tftpboot at the beginning.

3.4.1.2 Add Folder

- Choose a directory under which you want to add a new folder.
- Click **Add folder** button.
- Enter name of the new folder in the pop-up window, e.g. myfolder.
- Click **OK** button.

The new folder is then created accordingly, e.g. /tftpboot/myfolder.

3.4.1.3 Delete Folder

- Choose a directory you would like to delete.
- Click **Delete folder** button.

The folder you just deleted shall disappear from the **Directory** drop-down list.

3.4.1.4 Download File

- Choose a directory where you would like to download a file.
- Choose a file from the drop-down list.
- Click **Get** button.

3.4.1.5 Delete File

- Choose a directory where you would like to delete a file.
- Choose a file from the drop-down list.
- Click **Delete** button.

3.4.1.6 Upload File

- Choose a directory where you would like to upload a file.
- Click **Browse** button to locate a file in the local storage.
- Choose a file in the **File Upload** menu.
- Click **Open** button to confirm selection.
- Click **Put** button.
- Enter the destination filename in the pop-up window.
- Click **OK** button.

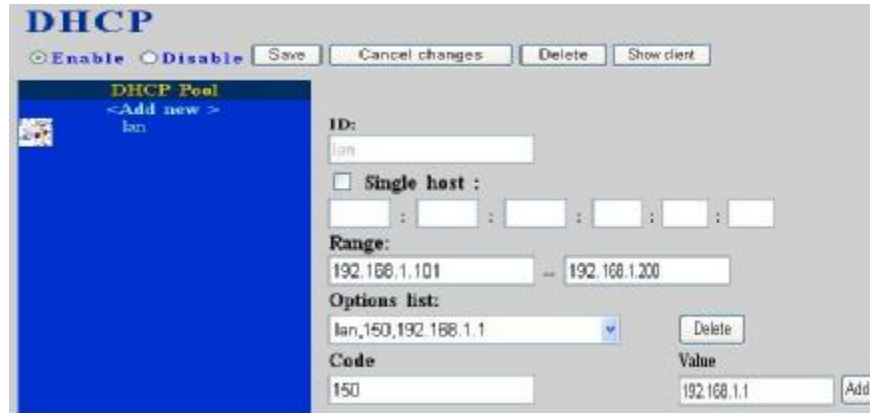
Now, the file you just uploaded should appear in current directory and is displayed in the **Download** drop-down list.

3.4.2 Disable TFTP service

Select **Disable** radio button, and then click **Save** button.

3.5 DHCP service

Select **Service** -> **DHCP service**, this page indicates current status of DHCP service. You can enable or disable DHCP service at any time.



3.5.1 Enable DHCP service

Select **Enable** radio button, and then click **Save** button. Afterwards, you can configure DHCP settings in details in this page.

3.5.1.1 Add DHCP Range

- Click **<Add new>** button from the left panel.
- Specify a pool ID (must have an alphabet initial) in the **Name** field.
- Check **Single host** if the binding is intended for a specific host only. Then give the MAC address of the host right below the checkbox.
- Enter a DHCP range of addresses available for lease in **Range**. If checked **Single host** the end address will be grayed out.
- Optionally, DHCP options¹ could be configured by entering an option code in **Code** and the option value in **Value** field and click **Add** button. The DHCP option just added shall be displayed in the **Options list** drop-down list. Follow the same steps to add more DHCP options. To delete an option, choose it from the **Options list** drop-down list and click **Delete** button.
- Click **Save** button to commit changes.

You should be able to see the newly added DHCP pool displayed in the **DHCP Pool** panel on the left side.

3.5.1.2 Edit DHCP Range

- Click the link of the pool ID to edit from the **DHCP Pool** panel.
- Edit settings shown on the right side.
- Click **Save** button.

3.5.1.3 Delete DHCP Range

- Click the link of the pool ID to delete from the **DHCP Pool** panel.
- Click **Delete** button.

¹ Refer to RFC 2132 for the details of available DHCP options.

The DHCP pool you just deleted shall disappear from the **DHCP Pool** panel.

3.5.1.4 Show Clients

Click the **Show client** button to list all leased LAN IP addresses and client details in a separated window.

3.5.2 Disable DHCP service

Select **Disable** radio button, and then click **Save** button.

3.6 IP PBX service

In **Service -> IP PBX service**, you can specify IP PBX global parameters and reload, backup, restore, or revert IP PBX configuration. You can also restart IP PBX service in this page.

The screenshot shows the 'IP PBX service' configuration interface. At the top, there are 'Save' and 'Cancel changes' buttons. The configuration fields include: 'PBX SIP port' (5060), 'RTP port range' (10000-20000), 'Max expiration time' (1800), 'Default expiration time' (600), 'PBX Caller ID' (PBX), and two unchecked checkboxes: 'Enable video coder' and 'Support devices multiplex Call-ID'. Below these are three empty input fields for 'Max active users', 'Max active calls', and 'Max wireless calls'. A red warning message states: 'IP PBX will reload configuration as soon as possible. Currently active calls will be disconnected in 3 minutes. Do you really want to Continue?'. Below this are buttons for 'Reload', 'Backup', and 'Restore'. Another red warning message states: 'IP PBX service will be restarted. Currently active calls will be disconnected instantly. Do you really want to Continue?'. At the bottom, there are buttons for 'Restart' and 'Revert'.

3.6.1 IP PBX global parameters

Global SIP settings, call records, and status of clients could be found in the first half of the page. **PBX SIP port** specifies the UDP port where the SIP service listens on. **RTP port range** limits the UDP ports used by the IP PBX for media transport. **Max** and **default expiration time** guard and advertise SIP registration respectively. **PBX Caller ID** is the default Caller ID if that of an incoming call is unknown. Check the **Enable video codec** checkbox if there will be video clients registering to the system. Checking **Support devices multiplex Call-ID** forces discrimination of SIP tags. Do this only when there is such a client device in the system and other devices

supporting the same. Otherwise, one may find the special device only got registered with this option but other clients or even SIP trunks fail due to such change. **Max active users** is for registration admission control to limit the maximum number of actively registered clients. **Max active calls** is for call admission control to limit the maximum number of concurrent calls, while **Max wireless calls** limits the calls made by explicitly specified wireless extensions. IP PBX service must be reloaded to activate changes.

3.6.2 Reload IP PBX configuration

By clicking **Reload** button, IP PBX will reload configuration as soon as possible until there is no active call. Current active calls will be retained up to 3 minutes. This is the most frequently used function in this page since any IP PBX configuration change has to be reloaded to take effect.

3.6.3 Backup IP PBX configuration

By clicking **Backup** button, IP PBX archives and encrypts current configuration into a time-stamped backup file under /tftpboot. To secure configuration files, it is suggested to download them to a local host through the **Get** function in **Service -> TFTP service** once a while. Note that the filename of a configuration file should not be changed; otherwise, it will be rejected by the **Restore** function.

3.6.4 Restore IP PBX configuration

Select a configuration backup file and click **Restore** button, and IP PBX will restore the configuration as current setup. One has to choose the **Reload** function to activate settings.

3.6.5 Restart IP PBX service

Click **Restart** button, and the IP PBX service will be restarted completely. Currently active calls will be disconnected immediately. This function should be rarely required unless the LAN setting has been changed or the service is found operating abnormally without problematic configuration could be identified.

3.6.6 Revert IP PBX configuration

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

To completely revert the whole system back to the factory default, press the hardware rest 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.

3.7 Stackable management

IP PBX is scalable and allows you to enlarge capacity by stacking up to 4 IP PBX boxes to form a cluster.

Stackable management

Current Operation Mode : **Standalone**

Save Cancel changes

Enable Master mode

Slaves : [dropdown] Remove Remove all

[input] . [input] . [input] . [input] : [input] Add slave

Registration Load Balance :

Disable By 305 Use Proxy By 302 Moved Temporarily

Master IP address: [input: 192] . [input: 168] . [input: 0] . [input: 1]

3.7.1 Operation modes

There are three different operation modes provided by IP PBX. They are standalone mode, master mode, and slave mode. The default mode is standalone mode while there is no cluster and IP PBX box works independently. Once a cluster is formed, there must be only one box running in master mode, rest of boxes run in slave mode in the cluster. For instance, let us consider full-capacity case, there would be one master box and 3 slave boxes in a cluster. Select **Service -> Stackable management**, you can see current operation mode of each IP PBX box.

3.7.2 Consolidated management

Beside, **Stackable management** page of master box not only indicates IP addresses of current slave boxes in a cluster but also allows you to add new slave boxes to cluster and remove slave boxes from cluster. You would find you can manage a cluster simply by **Stackable management** page of master box.

Within a cluster, any update of configuration information on master box would affect all of slave boxes automatically. As a result, you do not need to configure each box in a cluster. Instead, you shall only configure the master box to take control over the whole cluster in most cases².

Meanwhile, all of boxes in the same cluster share the same registration information and thus they work as one single box.

3.7.3 Configuration

The following examples illustrate how to configure a stack.

3.7.3.1 Case I: Stack multiple boxes at a time

- Ensure all of boxes have factory default setting before being stacked.
- Ensure all of boxes have TCP routing to each other via LAN interfaces if they are in different local networks.
- Select one of them to be the master box. Others are slave boxes consequently.

² However, for SIP and PSTN trunks and Meet-me conferences, you need to configure them respectively on each box in stack. It is similar to how you configure them in standalone mode.

- Go to **Service -> Stackable management** page on master box. Select **Enable Master mode** checkbox, enter IP address and port of each slave box then click **Add slave** button, one by one for each of them.
- Enable Registration Load Balance if clients support 305 or 302 response.
- After adding all of slave boxes, click **Save** button.
- On slave boxes, specify master's IP address.
- Finally, go to **IP PBX Service** page, click **Restart** button.

Now, all of boxes are stacked and a cluster is formed. You can start to configure extensions etc. on master box to make internal calls possible. You can also configure SIP and PSTN trunks on each box to enable inbound and outbound calls². Please note that after you make any changes of configuration in stack, remember to click **Reload** button on **IP PBX service** page of master box.

Once a Meet-me conference is configured on any one box in a cluster, it can be shared with all of other boxes. However, in order to share SIP and PSTN trunks with other boxes, you need to do following settings. For instance, there is a PSTN trunk (with a route pattern) configured on box A and assigned to some usergroups already. You would like to share this PSTN trunk with box B.

- On box B, configure a route with pattern equal to the one associated with this PSTN trunk, and then assign this route to the routegroup associated with the intra trunk from box B toward box A.
- Click **Reload** button on **IP PBX service** page of master box.

To explain step one more clearly, for example, box A is 192.168.1.100 and box B is 192.168.1.101. There is a PSTN trunk "pstn1" with a route pattern "9./1"³ configured on box A. In order to share this PSTN trunk among box A and B, on box B we can configure a route "BoxA-pstn1" with pattern "9./0"⁴ and assign it to the routegroup "_192168001101_192168001100" which associates with the intra trunk "_192168001101_192168001100"⁵.

3.7.3.2 Case II: Stack a new box with a running old box

This case means that there is a standalone box up running for a while. Now you would like to add a new box to form a cluster. Therefore, the old box will become master box later.

- Ensure the new box has factory default setting before being stacked.
- Ensure the new box has TCP routing to the old box via LAN interface if they are in different local networks.
- Now, backup IP PBX configuration on the old box. Then restore this configuration to the new box⁶.

³ 9./1 means pattern is 9. and number of stripped digits is 1.

⁴ Here, please note that we always use 0 for number of stripped digits.

⁵ You probably notice IP addresses of two boxes meshed by an intra trunk determine name of the intra trunk. In this case, the name of intra trunk from box B (192.168.1.101) toward box A (192.168.1.100) is "_192168001101_192168001100". Meanwhile, the routegroup associated with this intra trunk has the same name.

⁶ Please always ensure the initial operation mode of new box is standalone after it being restored with IP PBX

- On new box, remove all of trunk configuration including routes and routegroups.
- Go to **Service -> Stackable management** page on old box. Select **Enable Master mode** checkbox, enter IP address of new box then click **Add slave** button.
- Click **Save** button.
- Finally, go to **IP PBX Service** page on old box, click **Restart** button.

Now, the old box turns to be master box and the new box is slave box in cluster.

3.7.3.3 Case III: Add new slave boxes to an existing stack

Similarly to previous case, you can also add more slave boxes to an existing cluster.

3.7.3.4 Remove slave boxes from stack

- Go to **Service -> Stackable management** page on master box. Select IP address of slave box you would like to remove in **Slaves** drop-down menu, then click **Remove** button. You can also click **Remove all** button to remove all of slave boxes at a time.
- Click **Save** button⁷.
- Finally, go to **IP PBX Service** page on master box, click **Restart** button.

From now on, the cluster no longer owns the removed box. The removed box gets off the cluster and would turn back to be standalone once you click **Restart** button on **IP PBX service** page on it. Moreover, the master box would turn back to be standalone as well if all of slave boxes are removed and the cluster does not exist any more.

configuration.

⁷ Please deselect **Enable Master mode** checkbox before clicking **Save** button if there is no more slave box.

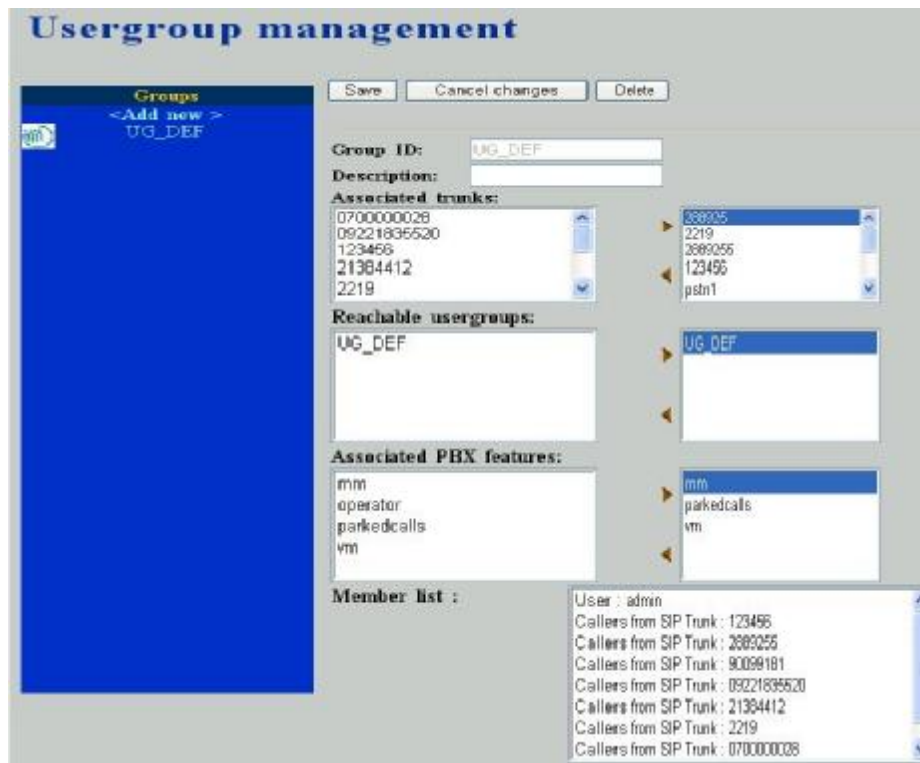
IP PBX Configuration

This section introduces steps to provision the IP telephony part of the IP PBX. Please note that reloading configuration is required in order to make new configuration effective⁸.

3.8 Usergroup Configuration

A usergroup is a logically grouping of users and their privileges. For instance, one could have couple of usergroups in an IP telephony network, e.g. Sales, Marketing, Administration, Accounting, and Engineering, etc. Each usergroup associates with a set of PBX features and call routing scopes. In other words, all users in the same usergroup share the same reachability of PBX features and final destinations.

The User Group Management page allows the administrator to manage usergroups. By selecting **User -> Usergroup**, one can add, edit, or delete usergroups. IP PBX service must be reloaded to activate changes.



3.8.1 Add Usergroup

- Click **<Add new>** button from the left panel
- Enter settings shown in **Table 4-1**.
- Click **Save** button.

⁸ Please refer to 3.6.2 for details.

Now, you should be able to see the newly added usergroup displayed in the **Groups** panel on the left side.

3.8.2 Edit Usergroup

- Click the usergroup link in **Groups** panel.
- Edit settings shown on the right side.
- Click **Save** button.

3.8.3 Delete Usergroup

- Click the usergroup link in **Groups** panel.
- Click **Delete** button.

The usergroup just deleted shall disappear from the **Groups** panel.

Table 4-1 Usergroup Configuration Settings

Field	Description
Group ID	A unique group name containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description info.
Associated trunks ⁹	Select outbound SIP trunks and PSTN trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time. F There is no SIP trunk and PSTN trunks 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. F There is no usergroup initially. Come back later to revise selection once more usergroups have been created.
Associated PBX features ¹⁰	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. F Most features have to be configured to function correctly. Remember to examine the settings of selected features before activating current configuration.

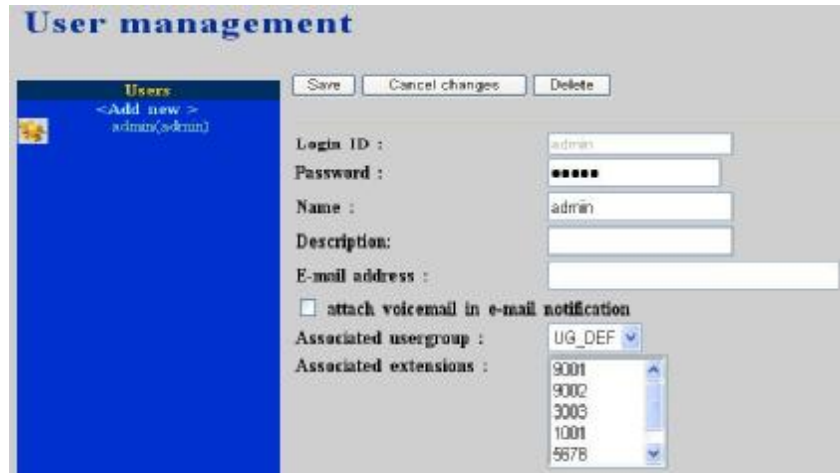
⁹ Please refer to 3.13 and 3.14 for details.

¹⁰ Please refer to 3.17 for details.

3.9 User Configuration

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 reception, break room, and lab areas.

The User Management page allows the administrator to manage users in the IP telephony network. Select **User -> User**, one can add, edit, and delete users. IP PBX service must be reloaded to activate changes.



3.9.1 Add User

- Click **<Add new>** button from the left panel.
- Enter settings shown in **Table 4-2**.
- Click **Save** button.

The newly added user should be displayed in the **Users** panel on the left side.

3.9.2 Edit User

- Click the link of the user to edit from the **Users** panel.
- Edit settings shown on the right side.
- Click **Save** button.

3.9.3 Delete User

- Click the link of the user to delete from the **Users** panel.
- Click **Delete** button to confirm deletion.

The deleted user shall disappear from the **Users** panel.

F A user can be deleted only when no extension is associated with it.

Table 4-2 User Configuration Settings

Field	Description
Login ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum. This is the ID for personal configuration through IP PBX Web management.
Password	Password for the user to access IP PBX Web management.
Name	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	E-mail address of the user for voicemail notification.
attach voicemail in e-mail notification	Check to enclose the message received in the notification e-mail as an attachment.
Associated usergroup ¹¹	Select the usergroup this user belongs to. F There is no usergroup initially. Come back later to revise selection if no appropriate usergroup could be chosen for now.

3.10 Device Configuration

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 Management page lets the administrator to create accounts for device extensions. Before a device can be reached from the IP PBX, the same account information has to be programmed into it through the configuration interface enabled by the device. Select **Device -> IP Phone**, one can add, edit, and delete devices. IP PBX service must be reloaded to activate changes.



¹¹ Please refer to 3.8 for details.

3.10.1 IP Phone

4.3.1.1 Add Device

- Click <**Add new**> button from the left panel.
- Enter settings shown in **Table 4-3-1**.
- Click **Save** button.

The newly added device should be displayed in **Devices** panel on the left side. Following shows steps to add extensions for the new device.

- Click link of the device shown in the **Devices** panel.
- Click **Show extensions** button to switch to the extension management page.
- Click <**Add new**> button from the left panel.
- Enter settings shown in **Table 4-3-2**.
- Click **Save** button.

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.

4.3.1.2 Edit Device

- Click device link in **Devices** panel.
- Edit settings shown on the right side.
- Click **Save** button.

Extensions associated with this device could be modified by following steps.

- Click **Show extensions** button.
- Click extension link in **Extensions** panel.
- Edit settings shown on the right side.
- Click **Save** button.

4.3.1.3 Delete Device

To delete one or more extensions associated with a certain device, follow steps below.

- Click device link in **Devices** panel.
- Click **Show 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 4-3-1 Device Configuration Settings

Field	Description
Device ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Device administration URL	(Optional) Administration URL of the device.
Enable Automatic Client Configuration	(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.
Vendor prefix	Specify if provided by phone
MAC address	MAC address of the device
Supplementary configuration	Specify if provided by phone
Codec preference	Preference order of supported codecs and packet times by phone
Enable VAD	VAD on phone
DTMF mode	DTMF mode used by phone

Table 4-3-2 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 ¹²	Select the user this extension associates with. F 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.

¹² Please refer to 3.9 for details.

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.
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 and Remove all button accordingly when some forwardings are no longer required.
--	--

3.10.2 Analog Phone

Connect an analog phone to a FXS port and configure the properties of the port as detailed in **Table 4-3-3**.

Table 4-3-3 FXS Extension Configuration Settings

Field	Description
POTS port	FXS port index
Extension number	A unique line number composed of digits only, e.g. 101; 32 digits maximum.
Pickup group	The pickup group the extension belongs to.
Unavailable timeout	Timeout for ringing before a call is answered.
User ¹³	Select the user this extension associates with. F 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.

¹³ Please refer to 3.9 for details.

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.

3.11 Route Configuration

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 4-4-1 explains digit set and wildcard characters.

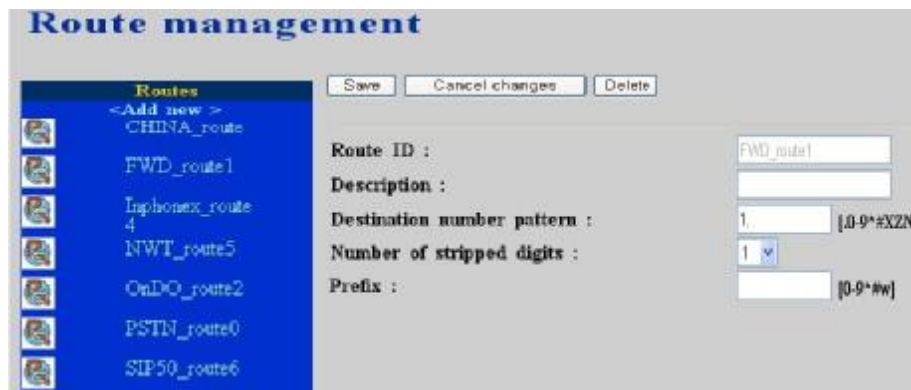


Table 4-4-1 Digit Set and Wildcard Characters for Route Patterns

Expression	Description
[<digits>]	Match any single digit listed explicitly. E.g., 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.

By selecting **Route -> Route**, the administrator can add, edit, and delete routes in the Route

Management page. IP PBX service must be reloaded to activate changes.

3.11.1 Add Route

- Click <**Add new**> button from the left panel.
- Enter settings shown in **Table 4-4-2**.
- Click **Save** button.

The newly added route should be displayed in **Routes** panel on the left side.

3.11.2 Edit Route

- Click the link of the route to edit from the **Routes** panel.
- Edit settings shown on the right side.
- Click **Save** button.

3.11.3 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 4-4-2 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.
--	---

3.12 Routegroup Configuration

A routegroup groups 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 -> Routegroup**, the administrator can add, edit, and delete routegroups in the Routegroup Management page. IP PBX service must be reloaded to activate changes.



3.12.1 Add Routegroup

- Click **<Add new>** button from the left panel.
- Enter settings shown in **Table 4-5**.
- Click **Save** button.

The newly added routegroup should be displayed in **Groups** panel on the left side.

3.12.2 Edit Routegroup

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

3.12.3 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 4-5 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 ¹⁴	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. F There may not be appropriate routes to select initially. One can come back later to revise it once the expected routes are added.

3.13 SIP Trunk Configuration

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.

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

3.13.1 Add SIP Trunk

- Click **<Add new>** button from the left panel.
- Enter settings shown in **Table 4-6**.
- Click **Save** button.

The newly added SIP trunk shall be displayed in **Trunks** panel on the left side.

3.13.2 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.

3.13.3 Delete SIP Trunk

- Click the link of the SIP trunk to delete from the **Trunks** panel.

¹⁴ Please refer to 3.11 for details.

- Click **Delete** button.

The deleted SIP trunk shall disappear from the **Trunks** panel.

Table 4-6 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.
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.
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 the SIP server on the ITSP side.
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 the SIP server on the ITSP side (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.
Outbound routegroup ¹⁵	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. F 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

¹⁵ Please refer to 3.12 for details.

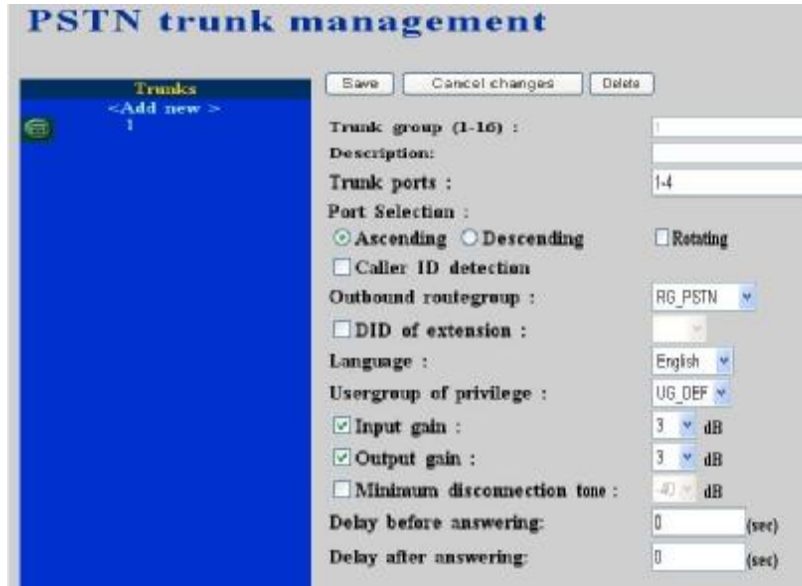
	heard from the trunk.
Usergroup ¹⁶ of privilege	When disabled DID, select a usergroup whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this trunk. F There may not be appropriate usergroups to select initially. One can come back later once the expected usergroup has been added.
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.

3.14 PSTN Trunk Configuration

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

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

¹⁶ Please refer to 3.8 for details.



3.14.1 Add PSTN Trunk

- Click **<Add new>** button from the left panel.
- Enter settings shown in **Table 4-7**.
- Click **Save** button.

The newly added PSTN trunk shall be displayed in **Trunks** panel on the left side.

3.14.2 Edit PSTN Trunk

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

3.14.3 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 4-7 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.
Outbound routegroup ¹⁷	Select a routegroup to associate routes with this trunk. Outbound calls match included route patterns could employ this trunk to access PSTN. F 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 ¹⁸ 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. F 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.

3.15 Trunk Terminal

A SIP trunk terminal 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 trunk terminal 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 trunk terminal is the account for a SIP trunk to authenticate with, exact the same identifier and

¹⁷ Please refer to 3.12 for details.

¹⁸ Please refer to 3.8 for details.

password must be used for both.

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

3.15.1 Add Trunk Terminal

- Click **<Add new>** button from the left panel.
- Enter settings shown in **Table 4-8**.
- Click **Save** button.

The newly added terminal shall be displayed in **Terminals** panel on the left side.

3.15.2 Edit Trunk Terminal

- Click the link of the terminal to edit from the **Terminals** panel.
- Edit settings shown on the right side.
- Click **Save** button.

3.15.3 Delete Trunk Terminal

- Click the link of the terminal to delete from the **Terminals** panel.
- Click **Delete** button.

The deleted terminal shall disappear from the **Terminals** panel.

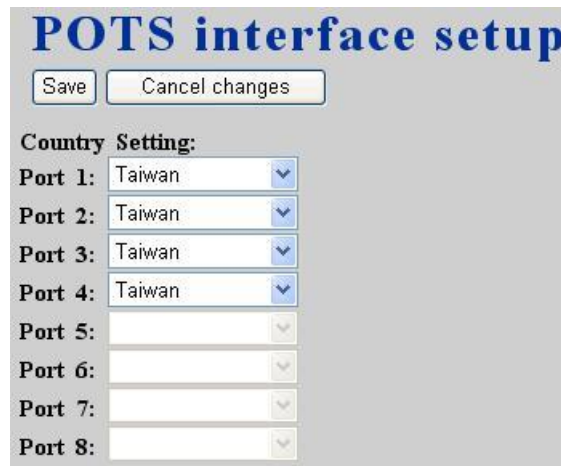
Table 4-8 Trunk Terminal Configuration Settings

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 ¹⁹ 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. F 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.

3.16 POTS setup

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.



3.17 Feature Configuration

A feature is a logical entity presenting a function module of IP PBX, e.g. meet-me conference, auto attendant, voice mail, etc. IP PBX service must be reloaded to activate changes.

¹⁹ Please refer to 3.8 for details.

3.17.1 Auto attendant

Auto attendant is an Interactive Voice Response (IVR) subsystem which answers incoming calls to a trunk and prompt voice menu to guide the caller to reach his expected extension. Usually the caller input the expected extension by pressing digits from the touch-tone pad on his/her phone, and the auto attendant to make the transfer. In some case, the caller may not know the extension of the party he/she is looking for so that there should be a key, say 0, to call the operator for help. Or, if the caller stays on the line for some time without giving a valid extension, IP PBX will transfer the call to the operator as well. Select **Feature -> Auto attendant** to configure auto attendant feature.

- Enter settings shown in Table 3.17-1
- Click **Save** button.

Table 3.17-1 *Auto Attendant Configuration 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.
Response timeout action	Select an action to be taken if the caller does not respond to the Auto Attendant 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.

3.17.2 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 3.17-2, and then click **Save** button.

Table 3.17-2 *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

3.17.3 Life line

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.



3.17.3.1 Add Life-line pattern

- Click **<Add new>** button from the left panel to add a new pattern.
- Enter settings shown in Table 3.17-3.
- Click **Save** button.

The newly added pattern should be displayed in **Life line** panel on the left side.

3.17.3.2 Delete Life-line pattern

- Click link to the pattern to delete from the **Life line** panel.
- Click **Delete** button.

The deleted pattern shall disappear from the **Life line** panel.

Table 3.17-3 *Life line Configuration Settings*

Field	Description
Number pattern	Pattern for emergency numbers. Note: this is the pattern <i>after digit stripping</i> . 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.

3.17.4 Meet-me conference

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 has to give the correct PIN of the room number.



Select **Feature** -> **Meet-me conference** to configure meet-me conference feature.

3.17.4.1 Add Meet-me Conference

- Click **<Add new>** button from the left panel to add a new conference room.
- Enter settings shown in Table 3.17-4.
- Click **Save** button.

The newly added room should be displayed in **Meet-me rooms** panel on the left side.

3.17.4.2 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 **Save** button.

3.17.4.3 Delete Meet-me Conference

- Click link to the room to delete from the **Meet-me rooms** panel.
- Click **Delete** button.

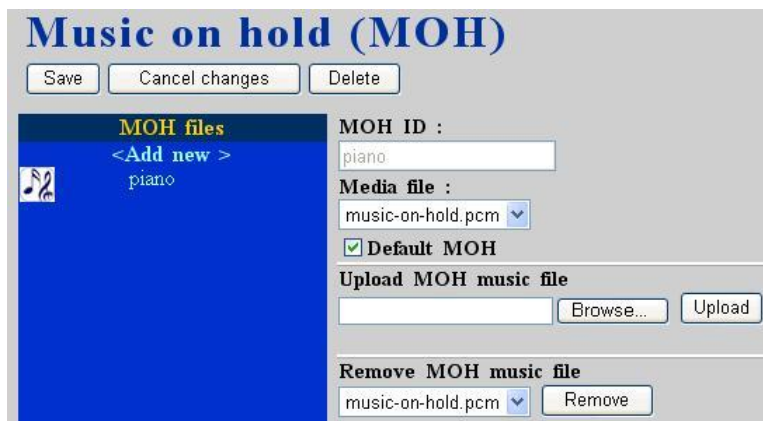
The deleted conference room shall disappear from the **Meet-me rooms** panel.

Table 3.17-4 *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: <ul style="list-style-type: none"> - # 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: <ul style="list-style-type: none"> - # 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

3.17.5 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.



3.17.5.1 Add MOH file

- Click **<Add new>** button from the left panel to add a new MOH file.
- Enter settings shown in Table 3.17-5.
- Click **Save** button.

The newly added file should be displayed in **MOH files** panel on the left side.

3.17.5.2 Edit MOH file

- Click the link of the file to edit from the **MOH files** panel.

- Edit settings shown on the right side.
- Click **Save** button.

3.17.5.3 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 3.17-5 *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.

3.17.6 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 top menu, enter settings shown in Table 3.17-6, and then click **Save** button.



Table 3.17-6 *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 <i>from</i> 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.

3.17.7 Auto-attendant prompts

This page allows replacing built-in AA prompts with user recordings. Choose a language and browse a corresponding recording from local storage. Then, click **Upload** to complete the replacement. To reset a prompt back to default, leave browsed file blank and directly click the **Upload** 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.

Auto-attendant prompts

Language : English

Prompts	Description
<input type="text"/>	Welcome to \${company}, please dial extension or dial \${key} for operator.
<input type="text"/>	I am sorry, that is not a valid extension Please try again.
<input type="text"/>	extension
<input type="text"/>	is not available
<input type="text"/>	is on the phone

Table 3.17-7 *Replaceable Auto-attendant 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

3.17.8 Meet-me prompts

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 **Upload** to complete the replacement. To reset a prompt back to default, leave browsed file blank and directly click the **Upload** button. Note that the replacement is done for the selected language only. Currently only following prompts could be replaced.

Table 3.17-8 *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.

3.17.9 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 **Upload** to complete the replacement. To reset a prompt back to default, leave browsed file blank and directly click the **Upload** button. Note that the replacement is done for the selected language only. Currently only following prompts could be replaced.

Table 3.17-9 *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.

4 Example Provisioning

This chapter introduces several practical configuration examples of IP PBX deployment. The configuration of IP PBX is very flexible and the expressiveness of usergroups, routegroups, and trunks are scalable enough to support various network architectures. Users could reference these examples and build a larger network involving multiple sites and advanced services.

4.1 Internal Extension Configuration

The procedure introduced below is recommended as the first step of a configuration task. The configuration only enables internal extension calls, but it serves as a good practice for administration.

- Configure a usergroup named **ALL** (refer to 4.1).
- Configure users and assign every user to usergroup **ALL** (refer to 4.2).
- Configure devices and an extension for each device (refer to 4.3).
- Assign each extension to a corresponding user (refer to 4.3).
- Configure each client phone with respect to the extension number and password in its IP PBX extension configuration accordingly.
- Reload IP PBX service (refer to 3.4.1).

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.

4.2 Case I: Single-Site Configuration

This case describes the typical settings of a single-site configuration; say Company A. Assume Company A has a DSL connection for Internet access and 2 PSTN subscriber lines as shown in **Figure 5-1**. The provisioning tasks include:

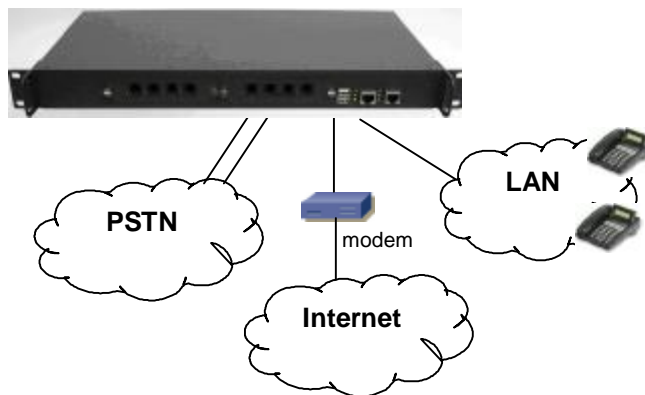


Figure 5-1

- There are staff phones in cubes and offices and utility phones in public areas.

- 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.

4.3 Case II: Two-Site Configuration

This case describes the typical settings of a two-site configuration; say Company B headquarters B-HQ and its branch B-BR in another country. Assume 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.

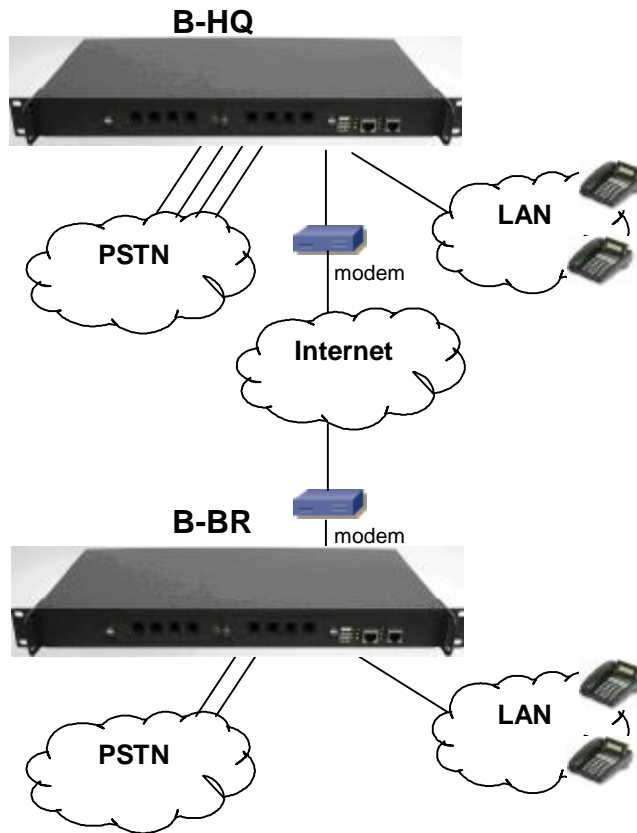


Figure 5-2

Configuration steps in B-HQ:

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 B-BR:

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.