Internet Telephony PBX System

IPX-2000

User's Manual

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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
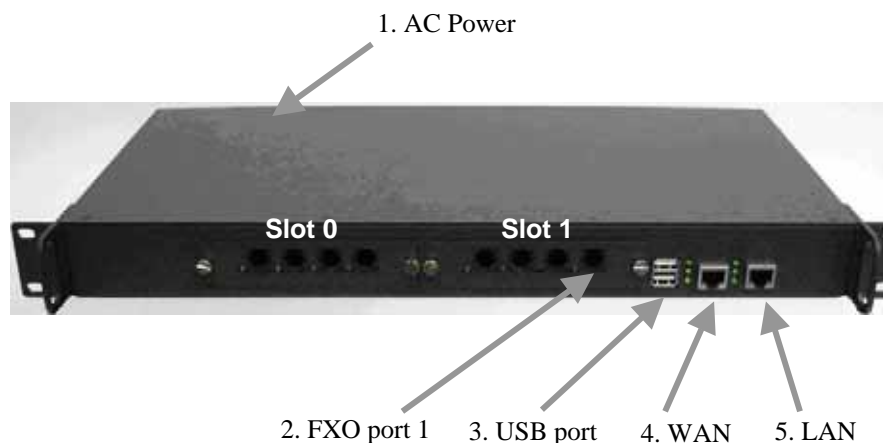
1. AC Power	110/220 Volt, 60 Hz
2. FXO ports	4 ports per daughter card, numbered from right to the left. The rightmost port at slot 0 is port 1 and the leftmost port at slot 1 (if installed) is port 8. FXO ports are to be connected to FXS jacks on wall or analog PBX using RJ-11 cables.
3. USB ports	2 external ports with compliance to USB 1.1/2.0. Plug in a USB hard drive for voicemail backup from the internal one.
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 as *admin* to log in for administration. The administrator password could be changed in the User Management page.

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.

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. You can choose one of these three options for your configuration, which are Static, DHCP, and PPPoE.

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.

2.3 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 accordingly. Besides, IP PBX service must be restarted.

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.

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.

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.

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   PBX status

This page includes two status viewers: **SIP peers status** lists registration status of each client or remote IP PBX and the locations from where they register. SIP trunk registrations, if any, are also shown near the end of the list. Column **Dynamic** stands for whether the listed IP address is dynamic or specified. **CDR status** 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.

2.8   PBX firmware

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.

2.9   Event viewer

Choose **System -> Event viewer** to list system log of NTP/DNS/DHCP/PPP services in a new window.

2.10 Reboot

By selecting **System -> Reboot**, you can reboot the machine after clicking the **Yes** button in this page. In case this would not reboot the system, press and release the hardware reset button quickly to 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.

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.

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@planet.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.

☞ 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 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 options[1] 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.

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.

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

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. **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

Select **Service -> Stackable management** to enlarge IP PBX capacity by stacking up to 4 IP PBX boxes to form a cluster. There are three operation modes for an IP PBX: Standalone if no cluster, Master to publish configuration, and Slave to subscribe configuration. IP PBX service must be reloaded to activate changes.

3.7.1 Standalone mode

This is the default operation mode. IP PBX runs independently to take client registrations without aid from another IP PBX. All extension accounts, SIP and PSTN trunks must be created on the box before they are available.

3.7.2 Master mode

In a stackable setup, there must be and only be one master. All extensions are managed on the master box only and it will publish them to slave boxes right after each extension creation or update. During run-time, client registrations to different stackable boxes will be advertised among them to synchronize the registered client list across cluster. PSTN and SIP trunks are not published, however, meshed SIP trunks are generated automatically to allow sharing trunks with other boxes in the cluster. To add a slave, give its IP address and click **Add slave**. To delete a slave, select it from the pull-down menu and click **Remove**.

3.7.3 Slave mode

When an IP PBX runs under slave mode, all extension accounts are updated from the master box so that no extension change should be done on a slave box. If a slave box is acquired later than the master box, one has to backup IP PBX configuration on the master, transfer the file onto the tftpboot directory on slave, and restore it there. Then, change mode to slave and remove all trunks since they are local to the master box only. One could add trunks to a slave or leverage meshed cluster trunks to access other's trunks. Note that once configured slave mode, a box cannot change back to the standalone or master mode. A work-around is to restore an earlier configuration when the box was not yet configured in the slave mode. Or, revert IP PBX configuration will reset to the standalone mode.

4   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 effective[2].

4.1   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 reach-ability 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.

4.1.1   Add Usergroup

- Click **<Add new>** button from the left panel

- Enter settings shown in **Table 4-1**.

- Click **Save** button.

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

4.1.2   Edit Usergroup

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

- Edit settings shown on the right side.

- Click **Save** button.

4.1.3   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.**

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[2] Please refer to 3.5.1 for details.

Associated SIP trunks[3]	**Select outbound SIP trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time.** ☞ **There is no SIP trunk initially. Come back later to revise selection once SIP trunks have been created.**
Associated PSTN trunks[4]	**Select outbound PSTN trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time.** ☞ **There is no PSTN trunk initially. Come back later to revise selection once PSTN 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[5]	**Select PBX features enabled to this usergroup. Here** vm **stands for Voice Mail,** mm **for Meet-me Conference,** parkedcalls **for Call Parking, and** lifeline **for Emergency trunk seizure.** ☞ **Most features have to be configured to function correctly. Remember to examine the settings of selected features before activating current configuration.**

4.2 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.

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[3] Please refer to 4.6 for details.
[4] Please refer to 4.7 for details.
[5] Please refer to 4.8 for details.

4.2.1 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.

4.2.2 Edit User

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

4.2.3 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.

☞ 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.**
Name	**Name of the user, either a real or a virtual one, e.g. Alice Lee or Conference Room.**
Password	**Password for the user to access IP PBX Web management.**
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[6]	**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 for now.**

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[6] Please refer to 4.1 for details.

4.3   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 -> Phone**, one can add, edit, and delete devices. IP PBX service must be reloaded to activate changes.

4.3.1   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.

4.3.2   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.

4.3.3   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	**Select the usergroup the extension belongs to.**
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[7]	**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.**
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.**
Selective call blocking	**(Optional) Enter calling numbers rejected by this extension, such as 102 or 102,103,104, etc. Numbers must be separated by commas.**
Unconditional call forward	**(Optional) Enter one 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 one number to which incoming calls are forwarded when not answered. The number could be an extension or a PSTN number with appropriate outbound prefix.**
Busy call forward	**(Optional) Enter one number to which incoming calls are forwarded when the**

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[7] Please refer to 4.2 for details.

	extension is in use. The number could be an extension or a PSTN number with appropriate outbound prefix.
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 accordingly when a forwarding is no longer required.**

4.4   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.

4.4.1   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.

4.4.2   Edit Route
- Click the link of the route to edit from the **Routes** panel.

- Edit settings shown on the right side.

- Click **Save** button.

4.4.3 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.**
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.**

4.5 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.

4.5.1 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.

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

4.5.3 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.**
Associated routes[8]	**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.**

4.6 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

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[8] Please refer to 4.4 for details.

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.

4.6.1 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.

4.6.2 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.

4.6.3 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 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.**
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.**
Trunk peer IP	**Specify IP address and UDP port of the remote SIP peer, which usually refer to the SIP proxy or registrar on the ITSP side.**
Trunk peer 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[9]	**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.**
Usergroups of privilege[10]	**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.**

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[9] Please refer to 4.5 for details.

[10] Please refer to 4.1 for details.

	☞ **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.**

4.7 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.

4.7.1 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.

4.7.2 Edit PSTN Trunk

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

4.7.3 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.**
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[11]	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.
Usergroups of privilege[12]	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.

4.8 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 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

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[11] Please refer to 4.5 for details.
[12] Please refer to 4.1 for details.

service must be reloaded to activate changes.

4.8.1 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.

4.8.2 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.

4.8.3 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.**
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.**
Usergroups of privilege[13]	**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**

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[13] Please refer to 4.1 for details.

	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.**

4.9 FXO setup

This page allows selection of country-based impedance of FXO ports. IP PBX service needs to be restarted before new setting takes effect.

4.10 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.

4.10.1 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 4.10-1
- Click **Save** button.

Table 4.10-1 *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.**

4.10.2 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 4.10-2, and then click **Save** button.

Table 4.10-2 *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**

4.10.3 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.

4.10.3.1 Add Life-line pattern
- Click **<Add new>** button from the left panel to add a new pattern.

- Enter settings shown in Table 4.10-3.

- Click **Save** button.

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

4.10.3.2 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 4.10-3 *Life line Configuration Settings*

Field	Description
Number pattern	**Pattern for emergency numbers**

4.10.4 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.

4.10.4.1 Add Meet-me Conference
- Click **<Add new>** button from the left panel to add a new conference room.

- Enter settings shown in Table 4.10-4.

- Click **Save** button.

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

4.10.4.2 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.

4.10.4.3 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 4.10-4 *Meet-me Conference Configuration Settings*

Field	Description
Room number	**Meeting room number, e.g. 8000.**
PIN to join	**PIN for admission to join a conference.**

4.10.5 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.

4.10.5.1 Add MOH file
- Click **<Add new>** button from the left panel to add a new MOH file.

- Enter settings shown in Table 4.10-5.

- Click **Save** button.

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

4.10.5.2 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.

4.10.5.3 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 4.10-5 *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 raw PCM (16 bits, 8000 Hz) file from the local host and press Upload. On successful uploading, the filename will appear in the pull-down menu of media files.**
Default MOH	**Check to use this music file for system default MOH globally.**

4.10.6 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 4.10-6, and then click **Save** button**.**

Table 4.10-6 *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 *from* address to accept a mailing request.**

4.10.7 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. 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 4.10-7 *Replaceable Auto-attendant Prompts*

Prompt	Description
Greeting	**Welcome to ${company}, please dial extension or dial ${key} for 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**

5 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.

5.1 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.

5.2 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:
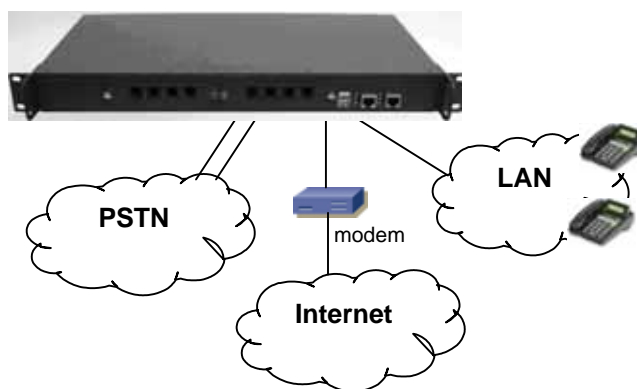
Figure 5-1

- 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.

5.3 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.
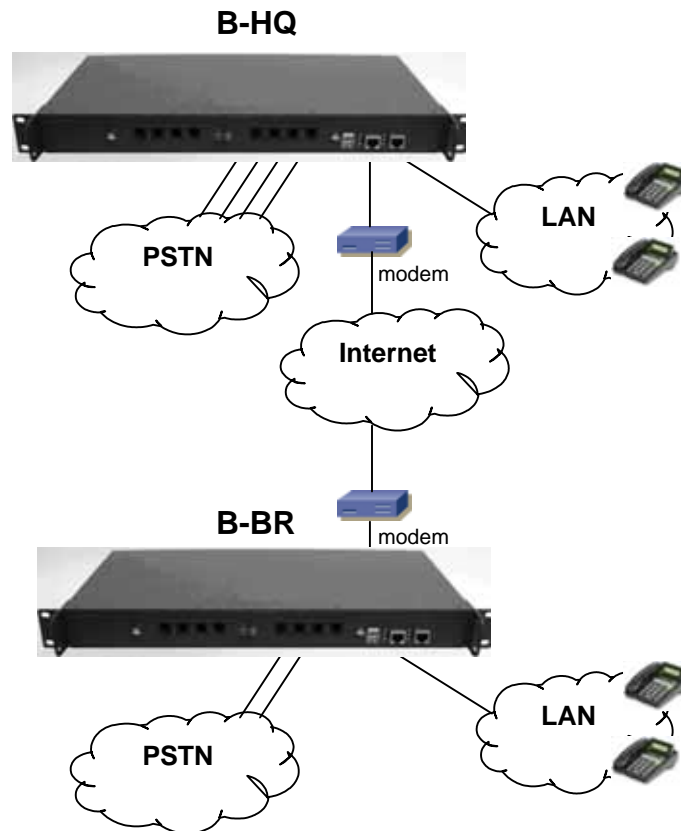

Figure 5-2

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.

5.4   Case III: Stackable Configuration

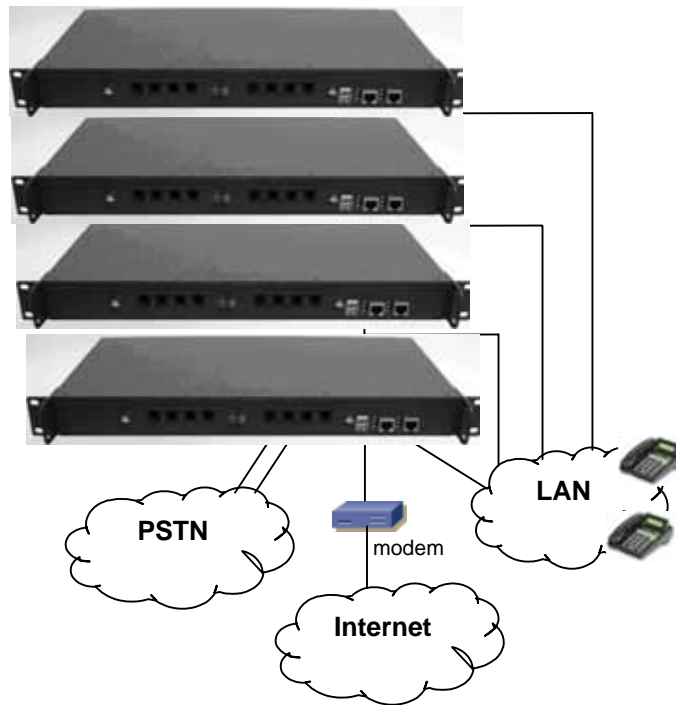This case shows a typical topology of a stackable configuration:

Figure 5-3