ISDN Internet Telephony PBX System

IPX-1800N

User's Manual

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This equipment complies with the requirements relating to electromagnetic compatibility, EN 55022 class A for ITE and EN 50082-1. This meets the essential protection requirements of the European Council Directive 89/336/EEC on the approximation of the laws of the Member States relating to electromagnetic compatibility.

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

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This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed in accordance with the instruction manual, may cause harmful interference to radio communication. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at the user's own expense.

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To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

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Revision

PLAENT IPX-1800N User's Manual

Revision: 1.0 (Nov. 2006)

Part No. EM-IPX1800NV1.0

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1 Overview

PLANET IPX-1800N ISDN IP PBX system are designed and optimized for the SMB, and SOHO daily communications. The IPX-1800N is the next generation voice communication platform for the small to medium enterprise. Designed as an open, scalable, and highly reliable telephony solution, the IPX-1800N is able to accept 30 extension registrations, and effectively meeting scales from various enterprises. Designed to run on a variety of VoIP applications, the IPX-1800N provides centralized call control, auto-attendant, voice conferencing, PSTN, and IP-based communications. The **IPX-1800N** integrates up to 4 ISDN telephony interfaces (Euro-ISDN ST-interface) to become a feature-rich PBX system that supports seamless communications between existing PSTN calls, IP phones and SIP-based endpoints.

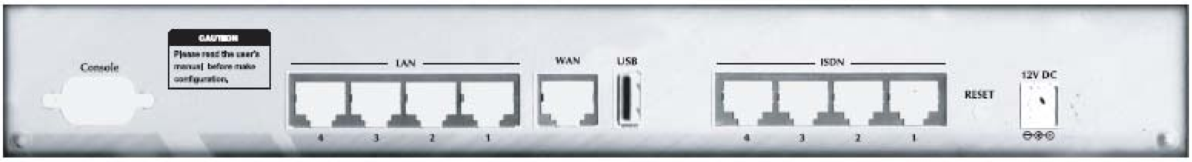
The IPX-1800N ISDN IP PBX system integrates telephony call processing, call control, voice mail, and a widely PBX application programming interface into a highly scalable architecture designed to support both traditional circuit-based and the Internet telephony service within a distributed enterprise communications network.

With IPX-1800N, standard SIP phones can be easily integrated in your office; plus the auto-config feature, you may integrate our IP Phone series - VIP-153T/VIP-154T, and the ATA (analog telephone adapter) series - VIP-156/VIP-157 to build up the VoIP network deployment in minutes.

Allowing distributed IP technology to meet traditional voice services, with proactive management interface, the IPX-1800N ISDN IP PBX system in the daily business processes, enterprises can make people more productive, more intelligent tasks, and more customer satisfaction.

1.1 Physical Interfaces

Front Panel of IPX-1800N



Rear Panel of IPX-1800N

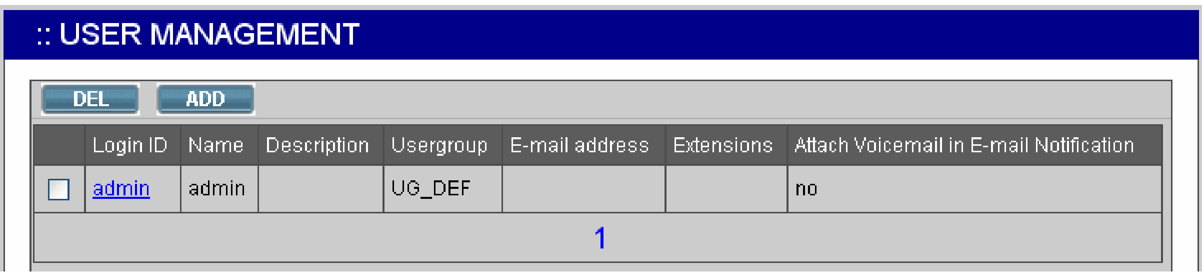
Power adapter	12V DC
Telephony interface ports	ISDN BRI TE ports are to be connected to NT points from PSTN or other ISDN network-side devices.
USB ports	1 external port with compliance to USB 1.1/2.0. Plug in a USB hard drive for voicemail backup from the internal one
WAN	Connect to a broadband modem or a WAN router
LAN	Connect to a LAN switch

2 System Configuration

This section describes how to configure system parameters used by PLANET ISDN IP PBX. The factory default of LAN IP address is 192.168.1.1. Connect to LAN port and the configuration Web interface is at https://192.168.1.1/. Once connected, the browser will ask for accepting a certificate. Click **Yes** to see the home page. Type in the default administrator ID and password (both are *admin*) to log in for administration.

The administrator password can be changed in the **User Management** -> **User**.

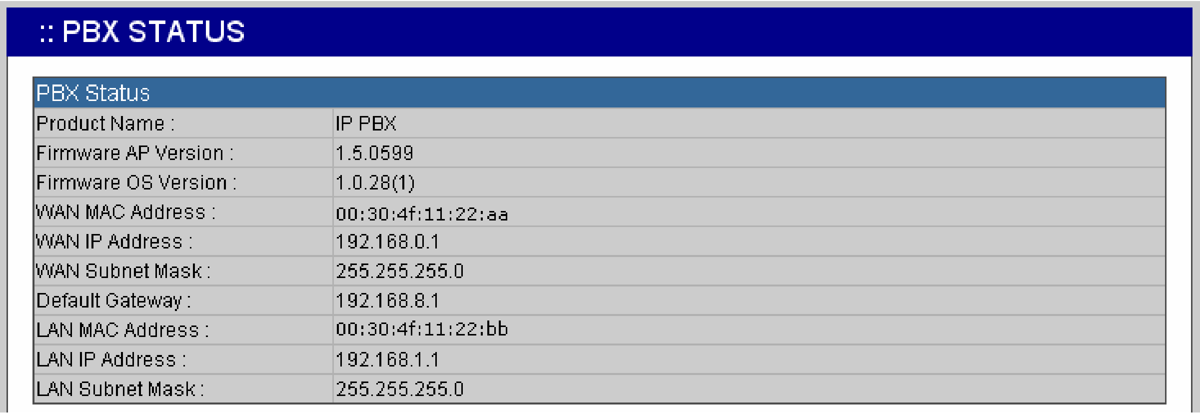
1. Click **admin** in the **Login ID**.
2. Change the password in **Password**.
3. Click **UPDATE** to change the password.



Note: For the system security, please change the password after the first log-in.

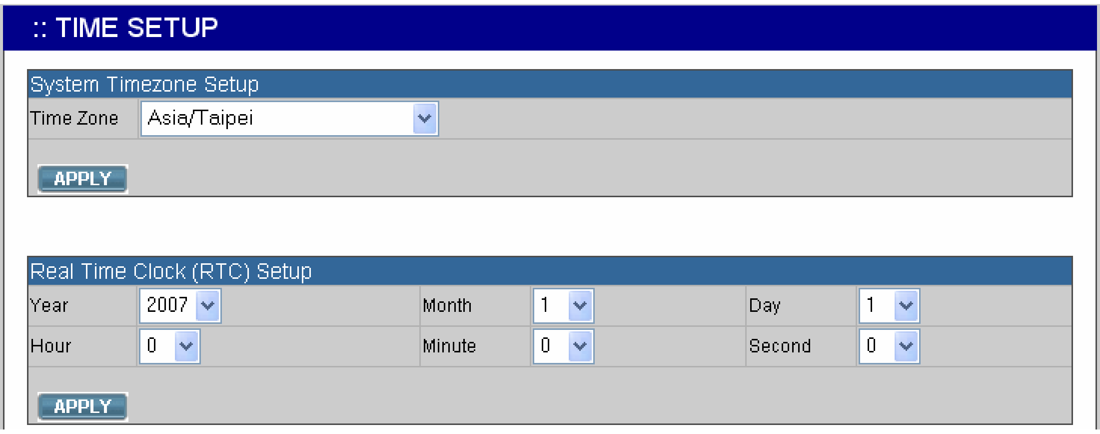
2.1 PBX System

The PBX System page briefs IP PBX status to the administrator. Firmware versions, IP addresses of WAN and LAN interfaces, and default gateway router are shown in this page. Click **PBX System** to see the basic information of IP PBX.



2.2 Time Setup

The Time Setup page allows administrator to configure time zone and date for PLANET IP PBX. With correct time setup, functions such as IVR, work time, and voicemail can present the actions at the right time. Select **System** -> **Time Setup** to see the current setting of time zone and date.

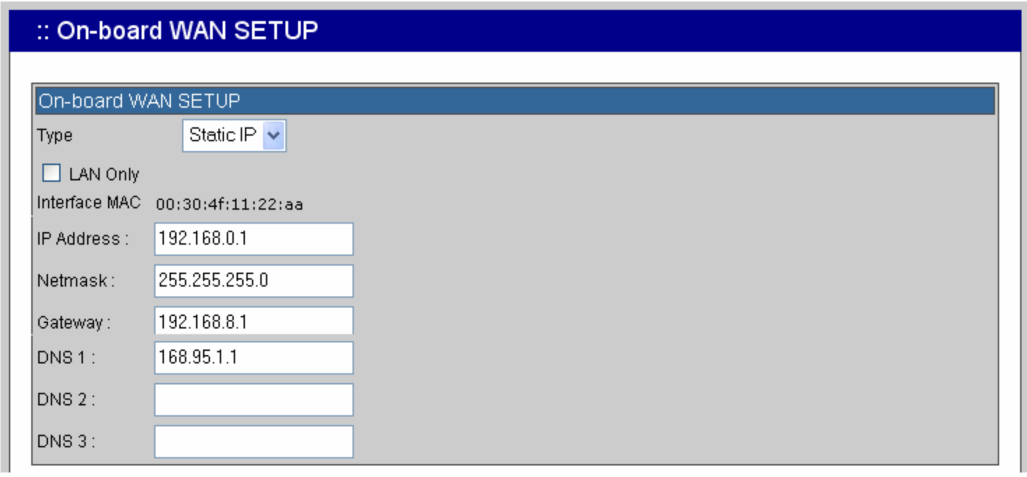


System Time Zone	Click a region/country in the **Time Zone** list, and click **APPLY** in **System Timezone Setup**.
Real Time Clock (RTC) Setup	Click year, month, day, hour, minute, and second in the correspondent list, and click **APPLY** in **Real Time Clock Setup**.

Note: When reset the time 15 minutes later than the time showed in RTC Setup, the system will ask for re-login.

2.3 On-board WAN Setup

The On-board WAN Setup page allows administrator to configure WAN network interface for PLANET IP PBX. Select **System** -> **On-board WAN Setup**, and the current setting of WAN network interface is displayed, e.g. type, IP address etc. Unless the "**LAN Only"** is selected, you can choose one of the three options, **Static IP**, **DHCP**, and **PPPoE** from the **Type** list for your configuration. Select **LAN Only** check box to disable WAN and only default router and DNS settings are applicable.

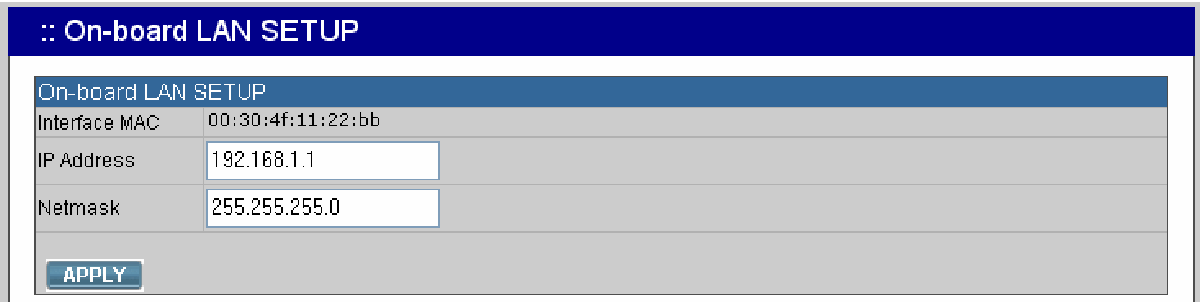


Static IP	You can click **Static IP** in the **Type** list, and manually configure the following information: • IP Address • Netmask • Default gateway IP address • Primary, secondary or third DNS servers Click "**APPLY"** to submit.
DHCP	Simply click **DHCP** in the **Type** list, and click **APPLY**. The acquired IP address, netmask, and default gateway information will show when revisit this page later.
PPPoE	1. Click **PPPoE** in the **Type** list. 2. Enter a user name and its password in **User Name** and **Password** boxes. 3. Click "**APPLY"** to submit. 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.
LAN Only	Select **LAN Only** to disable WAN IP settings but allow the configuration of default gateway and primary/secondary/third DNS servers.

2.4 On-board LAN Setup

The On-board LAN Setup page allows administrator to configure LAN network interface for PLANET IP PBX.

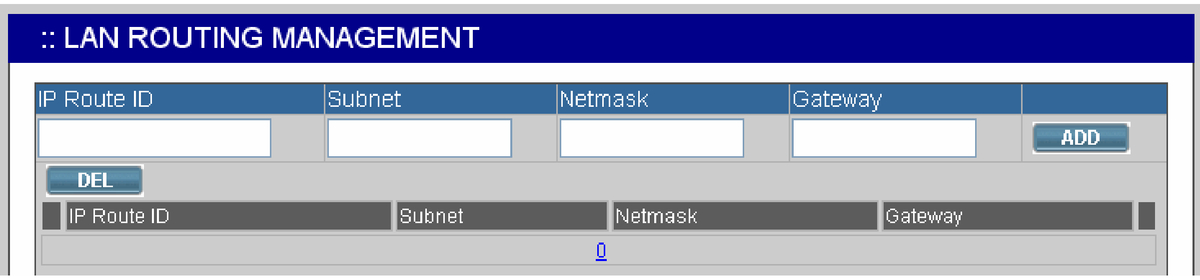
1. Select **System** -> **On-board LAN Setup** to see the current settings of LAN network interface.
2. Enter a new IP address and network mask.
3. Click "**APPLY"** to change the settings.



Note: By default PLANET 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. After configuration, go to **Service** -> **IP PBX Service**, and click **Restart** to active the changes.

2.5 LAN Routing

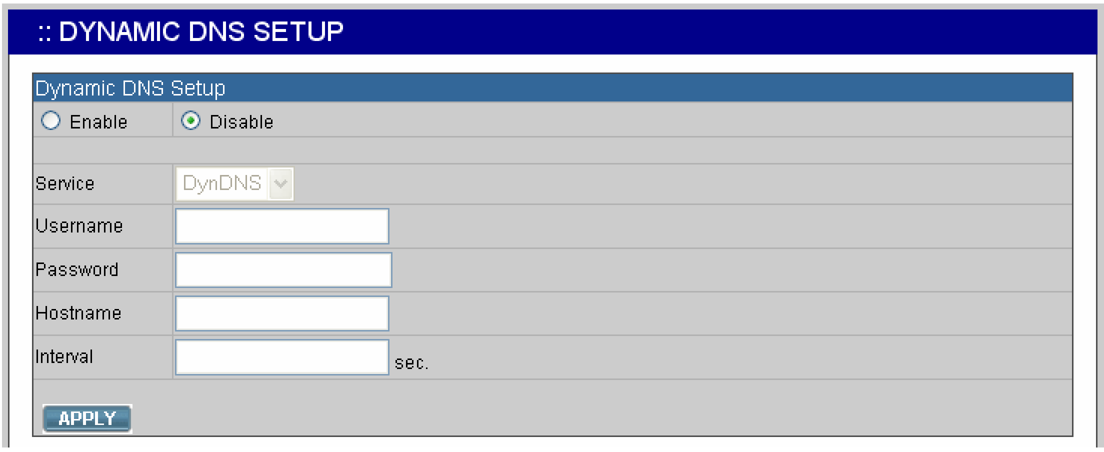
To enable static routing among LAN subnets, enter network information and the IP address of the corresponding gateway in the IP PBX's LAN. It is important to assure that the given gateway IP address sits in the IP PBX's LAN. Each subnet requires an entry even multiple subnets share the same gateway, 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 **ADD**. However, the IP PBX Service needs to be restarted to regard calls from designated LAN subnets as LAN traffic. Go to **Service** -> **IP PBX Service**, and click **Restart** to regard calls as LAN traffic.



Add a Route	1. Enter the **IP Route ID**, **Subnet**, **Netmask**, and **Gateway**. 2. Click **ADD** to have the newly added route in **IP Rout ID**.
Edit a Route	1. Edit the information in a row. 2. Click "**APPLY"** in the row to update the information.
Only Delete a Route	1. Select a route ID. 2. Click **DEL** to remove the route ID from the **IP Route ID** column.

2.6 Dynamic DNS Setup

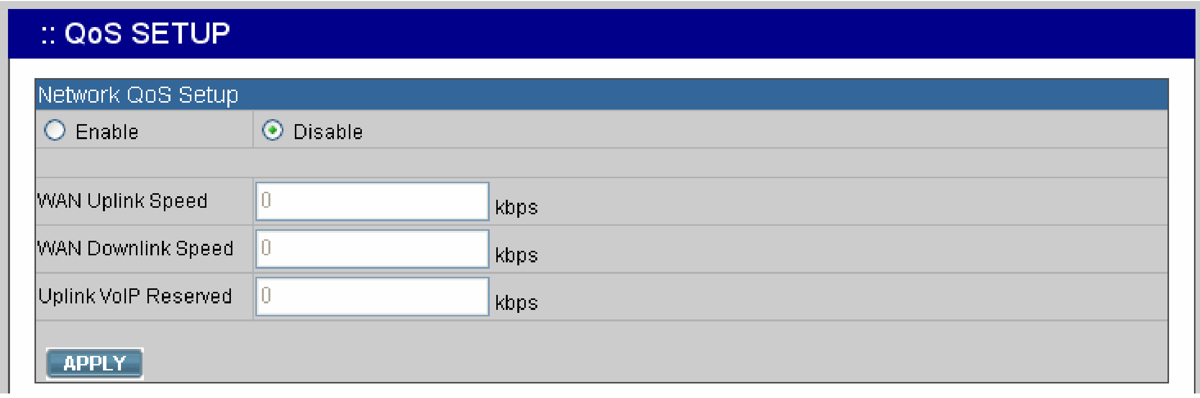
Dynamic WAN IP address causes difficulty for inbound connections from remote clients or IP PBX systems. A popular work-around is to adopt domain names provided by Dynamic DNS service providers and run a client on or behind the gateway router (or IP PBX). It is required to apply an account and create a hostname in the account before configuration. Click **Enable**, give account information and refresh interval to activate a Dynamic DNS client. The client then uses **Username** and **Password** to access its account and update the **Hostname** with the latest WAN IP address at **DynDNS** or **3322.net Service** in **Interval** seconds periodically.



Enable Dynamic DNS	Typical hostname has a form of *<hostname>.dyndns.org* or *<hostname>.3322.net*. The refresh interval is usually between 60 – 600 seconds depending on the volatility of WAN IP assignment. 1. Click **Enable**. 2. Click **DynDNS** or **3322.net** in the **Service** list. 3. Enter the **Username**, **Password**, **Hostname**, and **Interval**. 4. Click **APPLY**.
Disable Dynamic DNS	Click **Disable**, and then click **APPLY**.

2.7 QoS Setup

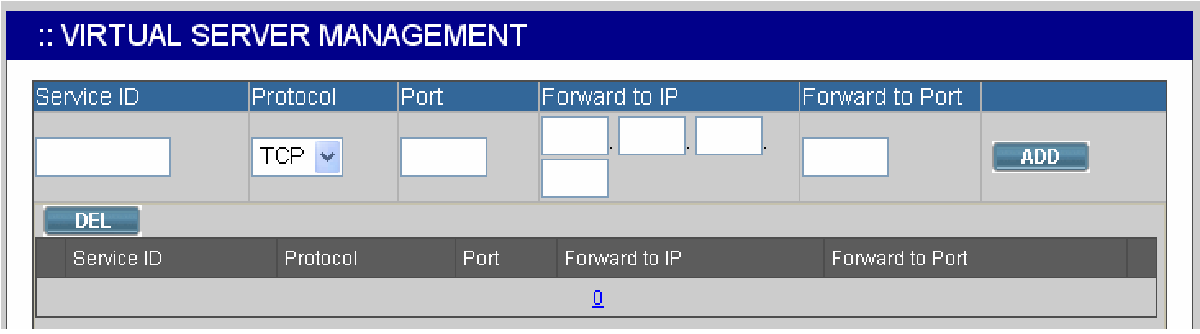
To assure the bandwidth reserved for the outgoing VoIP traffic over regular data traffic from LAN, the QoS Setup page offers three parameters to characterize the WAN link. The default QoS setting is disabled because these parameters must be correctly given according to the actual WAN speed.



Enable QoS	1. Click **Enable** 2. Enter the **WAN Uplink Speed**, **WAN Downlink Speed**, and **Uplink VoIP Reserved** (bandwidth). 3. Click **APPLY**. For a popular 2M/256K ADSL program, the WAN uplink speed would be 256 and the WAN downlink speed would be 2048. The Uplink VoIP reserved could be, say, 192 out of the total 256 kbps to allow 2 concurrent G.711 calls.
Disable QoS	Click **Disable**, and then click **APPLY**.

2.8 Virtual Server

You can configure PLANET IP PBX as a virtual server for remote users to access services such as the Web or FTP at your local site via Public IP Addresses. With proper settings, PLANET IP PBX can automatically redirect inbound traffic from WAN to local servers configured with private IP addresses. In other words, depending on the requested service (TCP/UDP) port number, the IP PBX redirects the external service request to the appropriate internal server (located at one of your LAN's Private IP Address). 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.



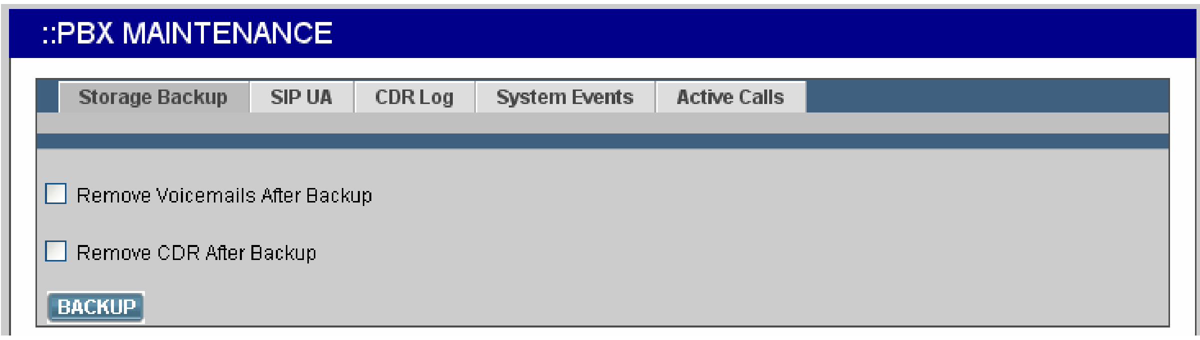
Add a Service	1. Enter the **Service ID**, **Protocol**, **Port**, **Forward to IP**, and **Forward to Port**. 2. Click **ADD** to add the newly service in the **Service ID**.
Edit a Service	1. Change any information in a row. 2. Click **APPLY** in the row to update the information.
Delete a Service	1. Select a service ID. 2. Click **DEL** to remove the service from the **Service ID**.

2.9 Maintenance

This page includes maintenance functions of IP PBX, including **Storage Backup**, **SIP UA**, **CDR Log**, **System Event**, and **Active Calls**.

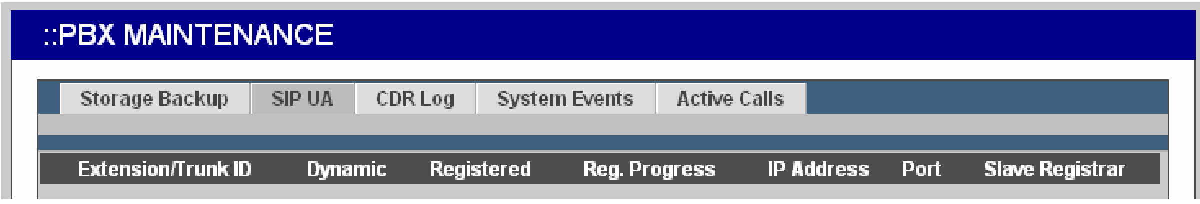
2.9.1 Storage Backup

To back up internal main storage, click **BACKUP**, and follow the 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, backup starts a few seconds later if the external USB drive is accessible and has enough free space. If the backup is successful, a new folder will be created on the external drive. After the backup, remove the USB connector of the external drive.



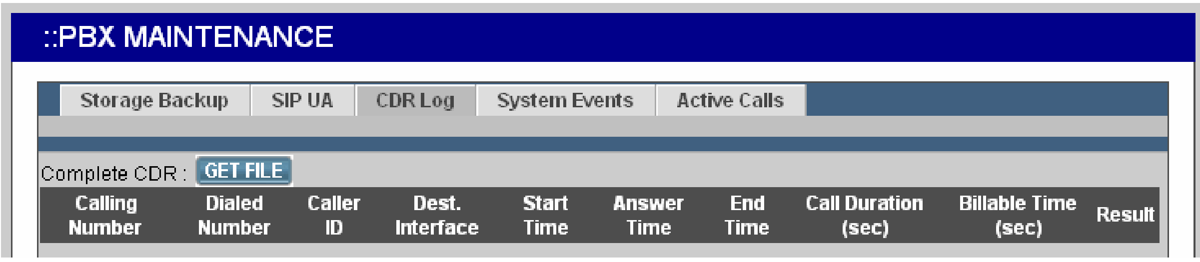
2.9.2 SIP UA

SIP UA lists the registration status of each client and remote IP PBX, and the **IP Address/Port** from where they register. SIP trunk registrations, if any, also show at the end of the list. The **Dynamic** column shows the listed IP address is dynamic or static. **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 locally.



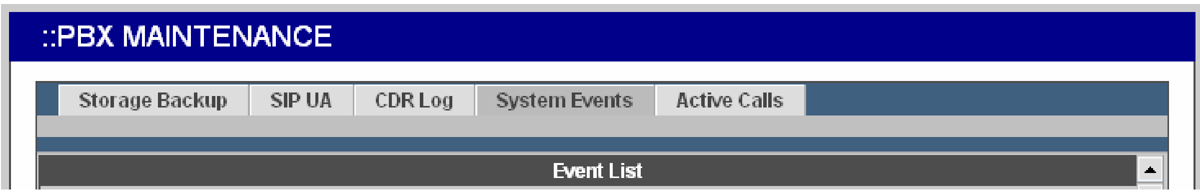
2.9.3 CDR Log

The CDR(Call Detail Record) Log shows each call record including Calling and Dialed Numbers, Caller ID, Destination Interface(trunk if outbound) in use, epochs when the call was made, answered and ended, and which yield the total and billable durations. The last column denotes the disposition of a call like answered or not.



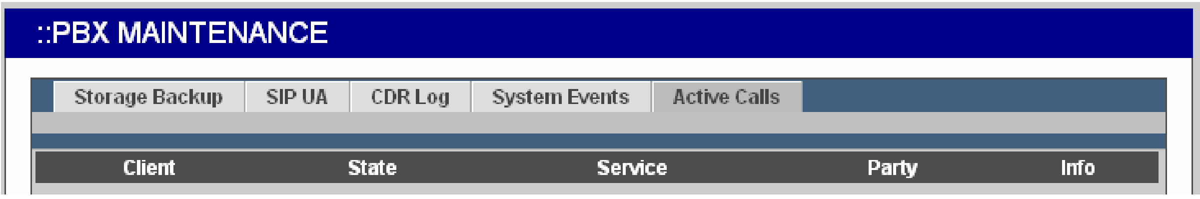
2.9.4 System Events

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



2.9.5 Active Calls

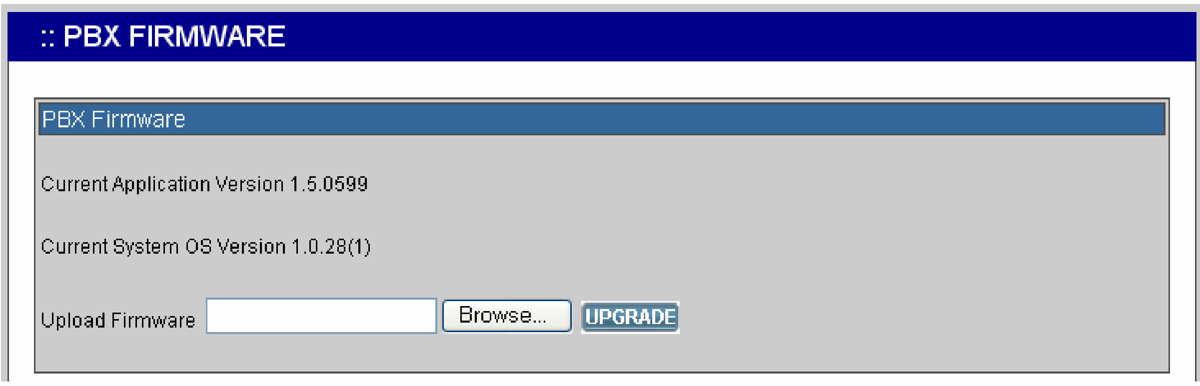
The Active Calls page shows current active calls. Columns Client and Party indicate the involved extensions or trunks of a call. State shows the state of a call, while Service gives the current action of the listed Client.



Field	Description	
Client	Show the caller or callee's extension number, port number, or SIP trunk ID.	
State	Connected	In the conversation.
	Ring	The client is a caller and is ringing a callee.
	Ringing	The client is a callee and is ringed by a caller.
	Reserved	FXS detects off-hook.
Service	Dial	The client is a caller.
	Answer	The client is a callee.
	IVR	Calls from FXO are picked up by Auto-Attendant.
	Meet-me	The client enters meet-me.
	Voicemail	The client enters voicemail.
Party	Shows extension number, POTS number or SIP trunk ID that is talking to this client.	

2.10 Firmware Upgrade

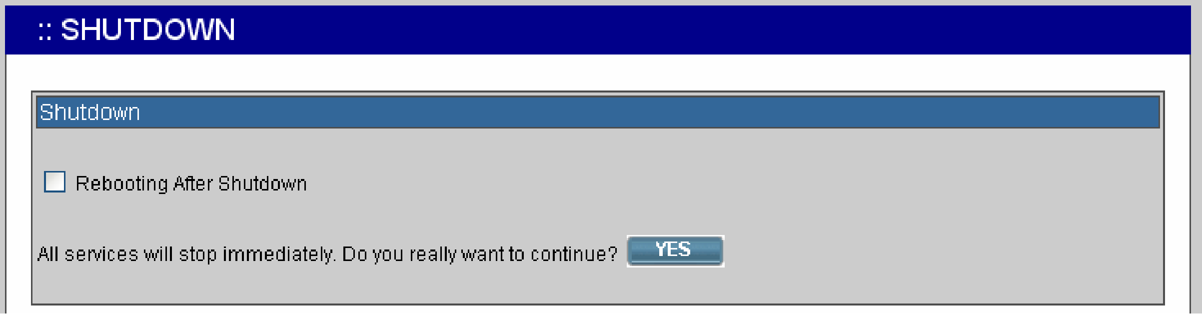
The version of the running PBX firmware could be found in **System** -> **Firmware Upgrade**. To upgrade current firmware, click **Browse** to locate a release file obtained from the vendor, and click **UPGRADE** to have the latest version of PBX firmware.



Note: Do not change the firmware file name, otherwise the system will reject it.

2.11 Shutdown

In **System** -> **Shutdown**, you can shutdown the machine by clicking **YES**, or reboot the machine by selecting the **Rebooting After Shutdown** check box and clicking **YES**. In case the software reboot fails, you can also press the hardware **Reset** button. It is advised to shut down IP PBX system before a power-off.



2.12 Logout

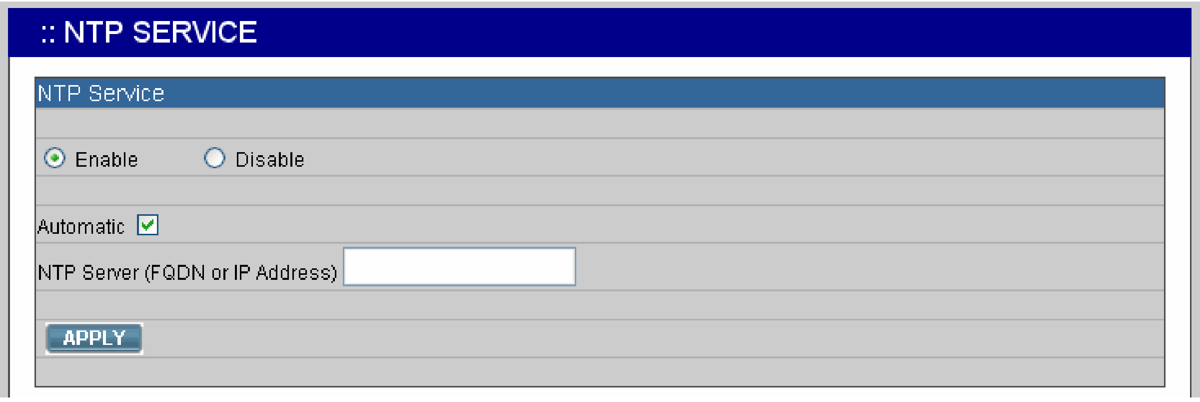
Logout button locates at the top-left of the webpage. Administrator can logout, and go back to the login page by clicking it.

3 Service Configuration

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

3.1 NTP Service

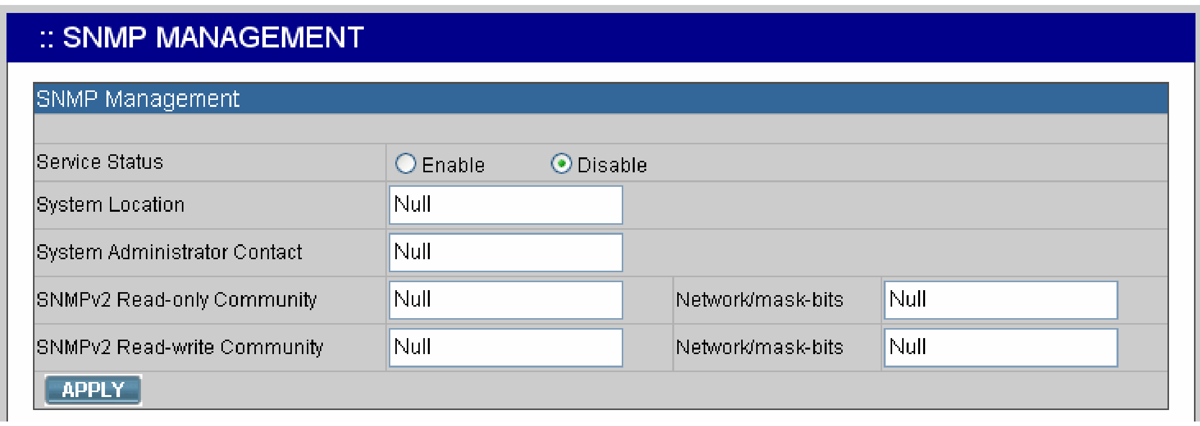
Select **Service** -> **NTP Service** to specify a NTP server for network time synchronization. You can enable or disable NTP service at any time.



Enable NTP Service	1. Click **Enable**. 2. Select **Automatic** check box to use server pool at pool.ntp.org; or, enter a fully qualified domain name or the IP address of a NTP server. 3. Click **APPLY**.
Disable NTP Service	Click **Disable**, and click **APPLY**.

3.2 SNMP Service

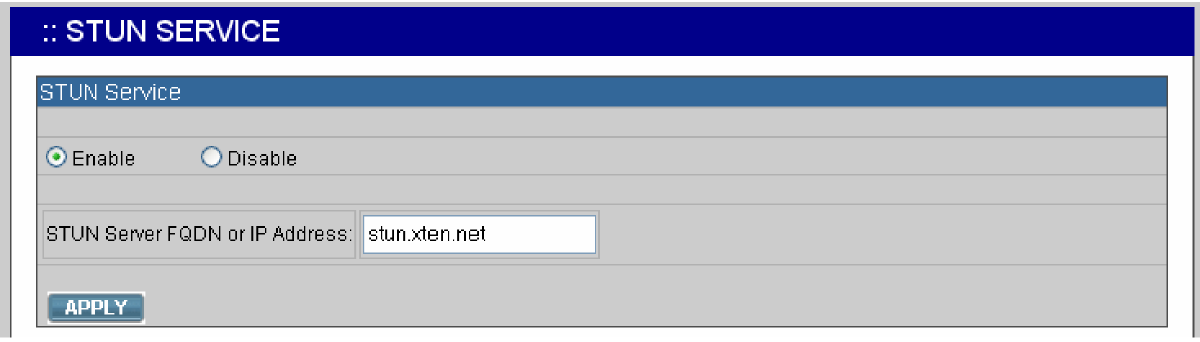
Select **Service** -> **SNMP Service** to specify Simple Network Management Protocol (SNMP) parameters for network status retrieval. You can enable or disable SNMP service at any time.



Enable SNMP Service	1. Click **Enable**. 2. Enter **System Location**, **System Administrator Contact**, **SNMPv2 Read-only Community** with allowed network specifications, and also those of the **Read-write Community**. 3. Click **APPLY**.
Disable SNMP Service	Click **Disable**, and click **APPLY**.

3.3 STUN Service

PLANET IP PBX has a built-in STUN client to solve NAT problems. 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.



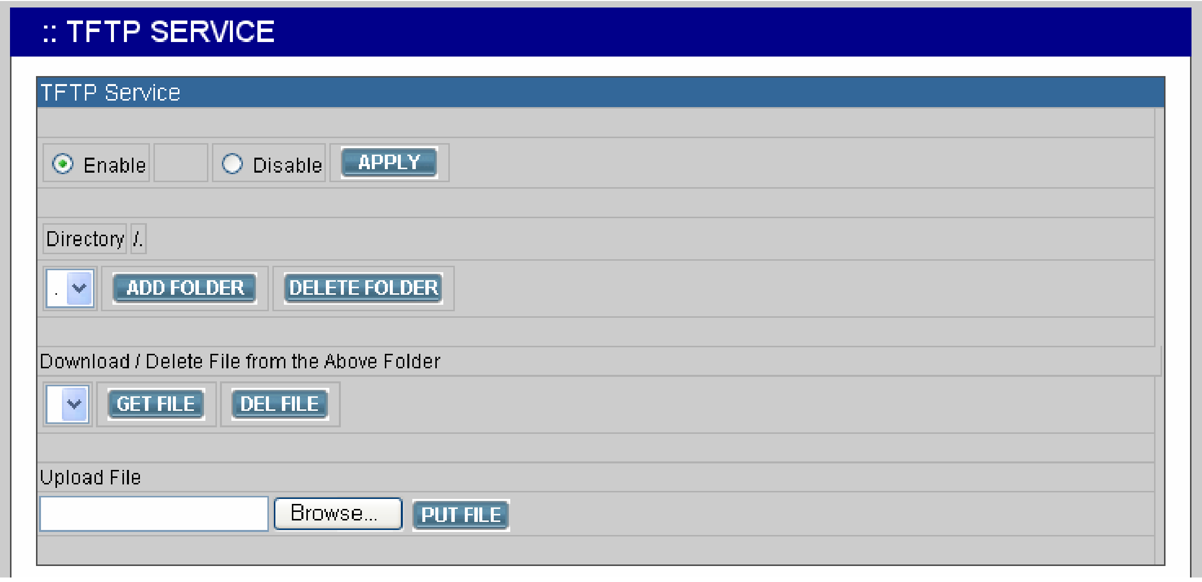
Note: You have to restart the IP PBX Service, after changing the STUN setting.

Enable STUN Service	1. Click **Enable**. 2. Enter a fully qualified domain name or the IP address of a STUN server. 3. Click **APPLY**. 4. Go to **Service** -> **IP PBX Service**, and click **RESTART** to reflect the changes.
Disable STUN Service	Click **Disable**, enter the fully qualified domain name or the static IP address of the external WAN interface and then click **APPLY**. 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. Go to **Service** -> **IP PBX Service**, and click **RESTART** to reflect the changes.

3.4 TFTP Service

Select **Service** -> **TFTP Service** to view the current status of TFTP Service. You can enable or disable TFTP Service at any time.

Enable TFTP Service: To click Enable, and then click **APPLY** to manage files, e.g. upload and download files to and from the IP PBX. Uploaded files can then be retrieved through TFTP Service.



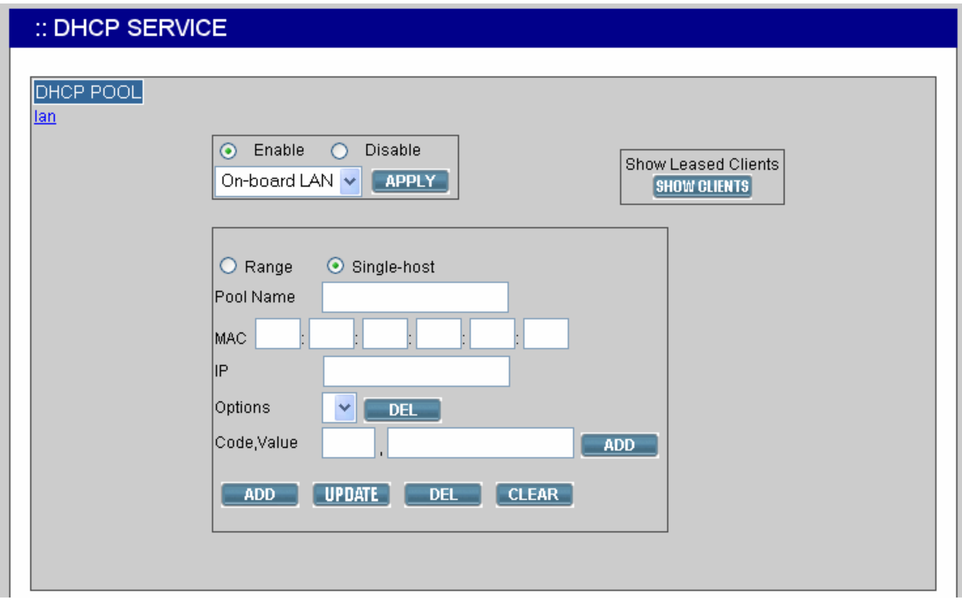
Change Directory	Current directory is shown in the field on the right side of **Directory**, for instance, it is /.at the beginning. Click a directory in the **Directory** list to change to a different folder. **Note:** The default directory is /. Initially, you may not be able to change the directory, since no folder is created under /. yet.
Add a Folder	1. Click a directory under which you want to add a new folder in the **Directory** list. 2. Click **ADD FOLDER**. 3. Enter a folder name in the pop-up dialog box, e.g. myfolder. 4. Click **OK** to see the newly added folder in the **Directory** list, e.g. /myfolder/.
Delete a Folder	1. Click a directory of a folder in the **Directory** list. 2. Click **DELETE FOLDER** to remove the folder from the **Directory** list. **Note:** A folder cannot be deleted if there is still file inside.

Download a File	1. Click a directory in the **Directory** list. 2. Click a file in the **Download / Delete File from the Above Folder** list. 3. Click **GET FILE** to download the file.
Delete a File	1. Click a directory in the **Directory** list. 2. Select a file in the **Download / Delete File from the Above Folder** list. 3. Click **DEL FILE** to remove the file.
Upload a File	1. Click a directory in the **Directory** list. 2. Click **Browse**. 3. Select a directory in the **Find** list, and then a file. 4. Click **Open**. 5. Click **PUT FILE** to upload the file. Now, the uploaded file should appear in current directory and is displayed in the **Download / Delete File from the Above Folder** list.
Disable TFTP Service	Click **Disable**, and then **APPLY**.

3.5 DHCP Service

Select **Service** -> **DHCP Service** to view the current status of the DHCP Service. You can enable or disable the DHCP Service at any time.

Enable DHCP Service: To click **Enable**, choose the main interface offering addresses, and then **APPLY** to configure DHCP settings.



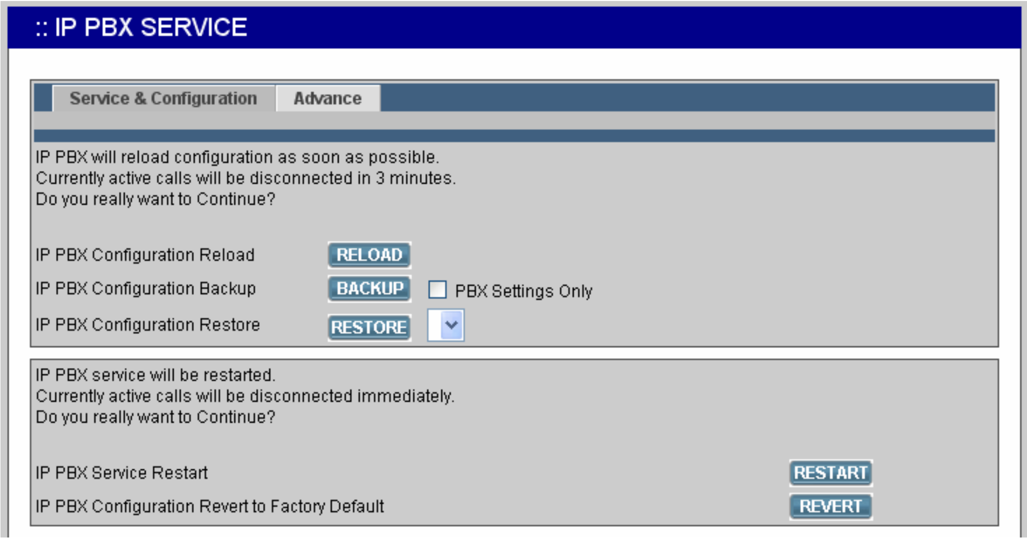
Note: If the IP PBX was shut down abnormally, Select **Service** -> **DHCP Service** and click **APPLY**, or Go to **Service** -> **IP PBX Service**, and click **RESTART** to active the DHCP service.

Add DHCP Range	1. Click **CLEAR**. 2. Enter a pool name (must have an alphabet initial) in **Pool Name**. 3. Select **Single-host** to enter an IP address of the host with **MAC**, if the binding is intended for a specific host only. 4. Enter a DHCP range of addresses available for lease in **IP**. The right address box will not show if **Single-host** is selected. 5. Optionally, DHCP options[1] could be configured by entering an option code and value in **Code,Value** and click **ADD**. The new DHCP option will show in the **OPTIONS** list. To delete an option, choose it from the **OPTIONS** list and click **DEL** after the box. 6. Click **ADD** at the bottom of the page to commit changes. You can see the newly added DHCP POOL displayed in the **DHCP POOL** list.
Edit DHCP Range	1. Click any pool name in the **DHCP POOL** list to see the settings on the right. 2. Edit the settings. 3. Click **UPDATE** to change the settings.
Delete DHCP Range	1. Click any pool name in the **DHCP POOL** list. 2. Click **DEL** to remove the pool name from the **DHCP POOL** list.
Show Clients	Click **SHOW CLIENTS** to see all leased LAN IP addresses and client details.
Disable DHCP Service	Click **Disable**, and click **APPLY**.

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

3.6 IP PBX Service

In **Service** -> **IP PBX Service**, you can click the **Service & Configuration** tab to reload, backup, restore, restart or revert the IP PBX configuration, or click the **Advance** tab for the IP PBX parameters settings.



3.6.6 Service & Configuration

Select **Service** -> **IP PBX Service**, and then click the **Service & Configuration** tab.

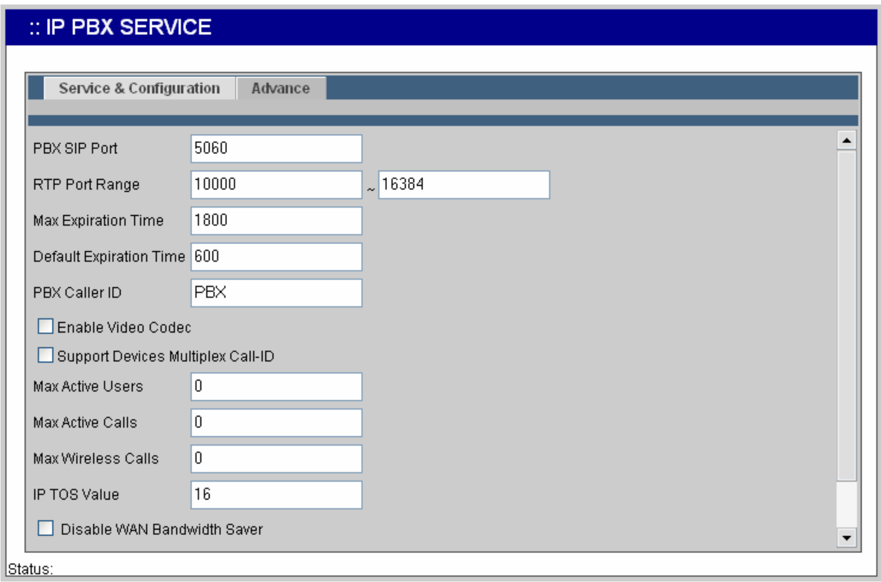
Reload IP PBX Configuration	Click **RELOAD**, and IP PBX will reload the configuration once there is no active call. If there is any active call, it will retain up to 3 minutes, and then IP PBX will reload. This is the most frequently used function in this page since any IP PBX configuration change has to be reloaded to take effect.
Backup IP PBX Configuration	Click **BACKUP**, and IP PBX archives and encrypts current configuration into a time-stamped backup file under tftpboot root directory. To secure configuration files, download them to a local host through the **GET FILE** function in **Service** -> **TFTP Service** once a while. Clear **PBX Settings Only** check box, both PBX and system (interfaces and services) settings will be archived in the backup file. **Note:** Do not change the configuration file name, or the **RESTORE** function will reject the configuration file.
Restore IP PBX Configuration	Click a configuration backup file in the list, click **RESTORE**, and IP PBX will restore the configuration as current setup. Go to **Service** -> **IP PBX Service**, and click **RESTART** to activate the

	settings.
Restart IP PBX Configuration	Click **RESTART**, and the IP PBX Service will restart completely. Currently active calls will be disconnected immediately. This function is rarely required unless the network setting has been changed, or the service operates abnormally without problematic configuration could be identified.
Revert IP PBX Configuration	Click **REVERT**, 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 other system services such as DHCP, TFTP, and NTP. 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 revert the whole system back to the factory default as much as possible, hold the hardware **Reset** button for 10 seconds. Since this will wipe out almost everything generated by the user, all system interfaces and services must be configured from scratch again if no appropriate backup configuration could be restored. Note that such reversion will not erase backup configurations and existing voicemails. Backup configuration files could be deleted in the TFTP Service page and voicemails could be deleted in the Maintenance page.

3.6.7 Advance

Select **Service** -> **IP PBX Service**, and then click the **Advance** tab to configure IP PBX parameters. After the configuration, go to **Service** -> **IP PBX Service**, and click **RESTART** to activate changes.



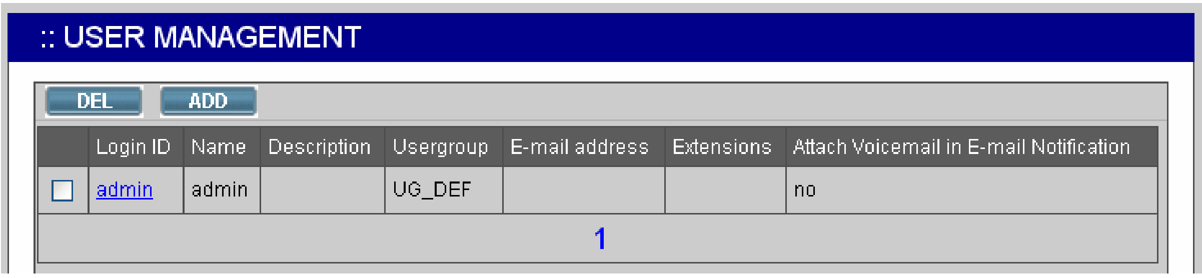
Field	**Description**
PBX SIP Port	Specify the UDP port where the SIP service listens on.
RTP Port Range	Limit the UDP ports used by the IP PBX for media transport. ☞ The port range needs to have at least equals to the (number of extensions (also count shared-lines) + number of SIP trunks (also count trunk terminals)) * 2. If selecting **Enable Video Codec**, the total amount needs to multiply by 2 to have the least requirements for RTP port range.
Max/Default Expiration Time	Guard and advertise SIP registration respectively.
PBX Caller ID	The default Caller ID for an unknown incoming call.
Enable Video Codec	Select if there will be video clients registering to the system
Support Devices Multiplex Call-ID	Select to force 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. Clear the box if you are not sure.
Max Active Users	Enter a number for registration admission control to limit the maximum number of active registered clients.
Max Active Calls	Enter a number for call admission control to limit the maximum number of concurrent calls.
Max Wireless Calls	Enter a number to limit the calls made by explicitly specified wireless extensions.
IP TOS Value	Set the TOS value in the IP header of RTP packets originated from IP PBX.
Disable WAN Bandwidth Saver	Select to disable attempts to use low-bit-rate codec (G.729A or G.723.1) for remote parties.
Enable DNS SRV Resolution	Select to enable looking up IP of dynamic clients or trunks by DNS Service records before their successful registrations.

4 IP PBX Configuration

This section introduces steps to provision the IP telephony part of the IP PBX. Note that reloading configuration is required in order to make new configuration effective[2].

4.1 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 Management** -> **User**, and one can add, edit, and delete users. Go to **Service** -> **IP PBX Service**, and click **RELOAD** to activate changes.

Add a User	1. Click **ADD**. 2. Enter settings shown in **Table 4.1**. 3. Click **ADD**. 4. Click **BACK** to see the newly added user in the **Login ID**.
Edit a User	1. Click a user in the **Login ID**. 2. Edit settings shown in **Table 4.1**. 3. Click **UPDATE**.
Delete a User	1. Select a **Login ID**. 2. Click **DEL** to remove the user from the **Login ID**.

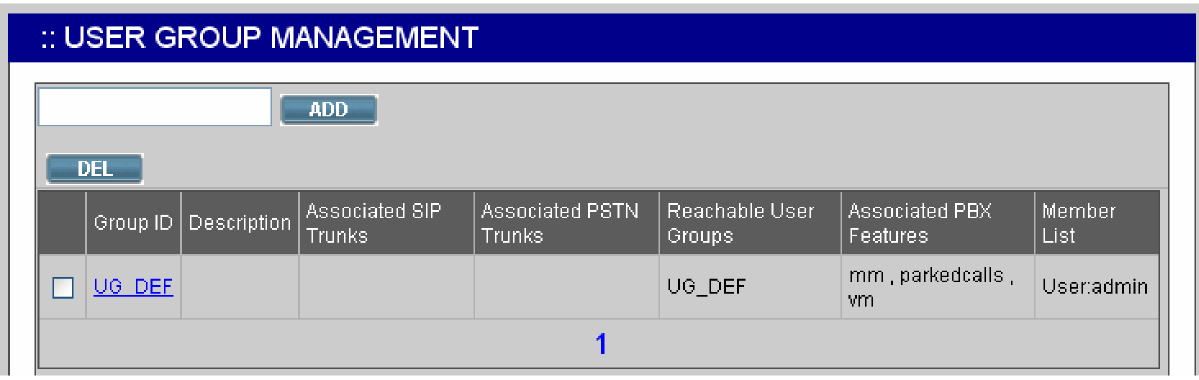
[2] Please refer to 0 for details.

Table 4.1 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.
Description	Arbitrary description information.
E-mail Address	E-mail address of the user for voicemail notification.
Attach Voicemail in E-mail Notification	Select to enclose the message received in the notification e-mail as an attachment.
Usergroup	Select the usergroup this user belongs to. ☞ If there is not any appropriate usergroup to select, come back later to revise this selection if no appropriate usergroup could be chosen for now.
Extensions	Show the extensions associated with this user.

4.2 User Group 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. Select **User Management** -> **User Group**, and one can add, edit, or delete usergroups. Go to **Service** -> **IP PBX Service**, and click **RELOAD** to activate changes.

Add a User Group	1. Enter a usergroup name beside the **ADD** button, and then click **ADD**. 2. The name will show in **Group ID**. 3. Click the name in **Group ID** to view the edit page. 4. Enter settings shown in **Table 4.2**. 5. Click **SET** to save the settings, and click **BACK** to return to the USERGROUP MANAGEMENT page. Now, you can see the newly added usergroup displayed in the **Group ID**.
Edit a User Group	1. Click a usergroup name in the **Group ID**. 2. Edit settings shown in **Table 4.2**. 3. Click **SET**.
Delete a User Group	1. Select a **Group ID**. 2. Click **DEL** to remove the usergroup from the **Group ID**.

Table 4.2 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 information.
Associated Trunks	Select outbound SIP trunks and PSTN trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time. **Group ID**: The default number is "0". A trunk with Group ID "0" does not form a balance group with any other trunks in Group 0. If Group ID is 1~9, trunks with the same Group ID form a usage balance group. **Weight**: the weight of a trunk to be selected in a trunk balance group for an outgoing call. ☞ Trunks with the same group ID must be put together, or the function will not work. ☞ If there is not any appropriate SIP trunk and PSTN trunks to select, come back later to revise selection once trunks have been created.

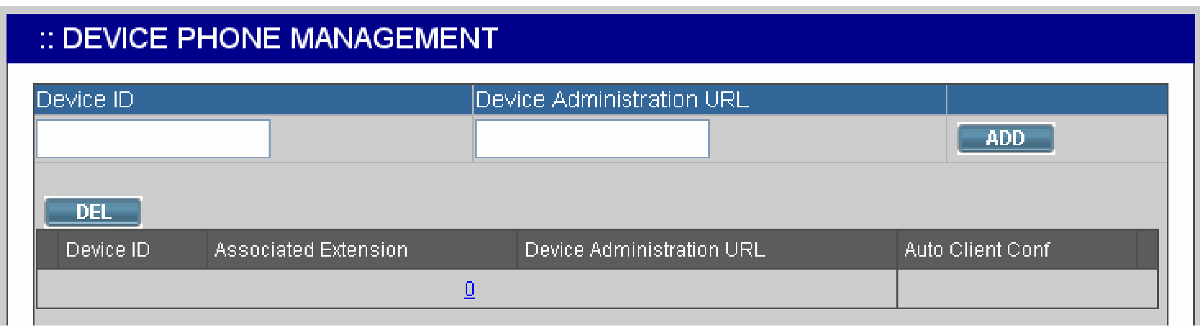
Reachable User Groups	Select other usergroups reachable from this usergroup. By default, only users in the same usergroup can reach one another. ☞ If there is not any appropriate usergroup to select, come back later to revise this selection, once more usergroups have been created.
Associated PBX Features[3]	Select PBX features enabled to this usergroup. Here vm stands for Voice Mail, mm for Meet-me Conference, parkedcalls for Call Parking, and operator for operator service. ☞ Most features have to be configured to function correctly. Remember to examine the settings of selected features before activating current configuration.
Member List	Show the users associated with this usergroup. ☞ If there is not any appropriate user to select, come back later to select, once one or more users have been created and associated with this usergroup.

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.

4.3.8 IP Phone

The DEVICE PHONE MANAGEMENT page lets the administrator to create IP Phone devices. Before a device can be reached from the IP PBX, the same account information has to be programmed into the device through the configuration interface enabled by the device. Select **Device** -> **IP Phone** to add, edit, and delete devices. Go to **Service** -> **IP PBX Service**, and click **RELOAD** to activate changes.



3 Please refer to 5 for details.

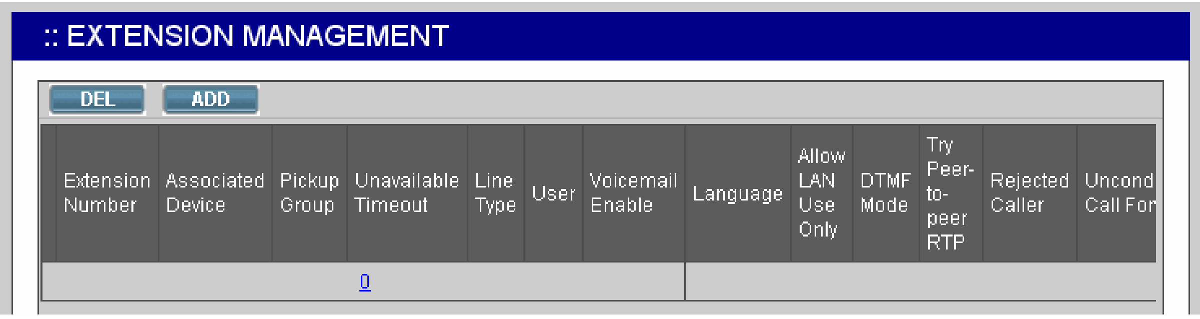
Add a Device	1. Enter a device name in the **Device ID** box, and a URL in the **Device Administration URL** box. 2. Click **ADD** to see the newly added device in the **Device ID**.
Edit a Device	Once create the device, you can modify its information through the following steps. 1. Modify the **Device Administration URL** and click **LINK** as a shortcut to the device administration URL. 2. Click **EDIT** to see the Enable Automatic Client Configuration (ACC) page. **Table 4.3.1** is a reference for detailed ACC settings which is used for auto-configuring IP phones. One can specify the MAC address and audio preferences of the phone. Note that for phones using HTTP for auto-configuring, DHCP setting needs a new option 151 with a value of *http://<IP PBX LAN IP>/tftpboot/* in the **Code,Value** box in **Service** -> **DHCP Service**. No extra settings needed if the phone uses TFTP for auto-configuring. 3. Click **ENABLE** to see **Enable** shows in the **Auto Client Conf** column. Click **EDIT** and then **DISABLE** to disable the function.
Delete a Device	1. Select a **Device ID**. 2. Click **DEL** to remove the device from the **Device ID**.

Table 4.3.1 ACC (Automatic Client Configuration) Settings

Field	Description
Device	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Vendor Prefix	Ask your IP Phone vendor for the Prefix.
MAC Address	MAC address of the device.
Supplementary Configuration	Specify if provided by the phone.
Codec Preference	Preference order of supported codec and packet times of the phone.
Enable Voice Activity Detection (VAD)	VAD is a technique that detects absence of audio and conserves bandwidth by preventing the transmission of "silent packets" over the network. ☞ Select if your IP Phone supports VAD.
DTMF mode	Choose a DTMF mode used by the phone

4.3.9 Extension of IP Phone

The EXTENSION MANAGEMENT page lets the administrator to create extensions. Select **Device -> Extension of IP Phone**, and one can add, edit, and delete extensions. Go to **Service -> IP PBX Service**, and click **RELOAD** to activate changes.



Add an Extension	1. Click **ADD** to set an extension. 2. Enter settings shown in **Table 4.3.2**. 3. Click **ADD**. 4. Click **BACK** to see the newly added extension.
Edit an Extension	1. Click an extension in the **Extension Number**. 2. Edit settings shown in **Table 4.3.2**. 3. Click **UPDATE**. 4. Click **BACK** to see the updated information.
Delete an Extension	1. Select an extension numbers. 2. Click **DEL** to remove the extension from the **Extension Number**.

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.
Associated Device	Select the Device this extension associates with.
Password	Password of this extension. Same password must be configured on the device side as well.
User[4]	Select the user this extension associates with. ☞ If there is not any appropriate users to select, one can come back later once the expected user has been added.

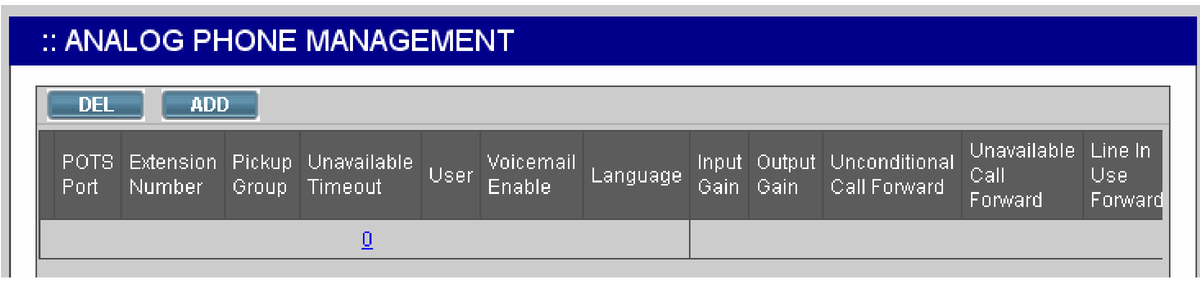
[4] Please refer to 4.1 for details.

Pickup Group	The usergroup that the extension can pick up. The extension can set a usergroup that when any extension in the usergroup rings, the extension can press *8 to pick up the call in ringing state.
Line Type	Specify the type of connection, wired or wireless, of the client with the extension.
Language	Preferred language for system instructions heard from the extension.
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.
Unavailable Timeout	Timeout for ringing before a call is answered.
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.
Try Peer-to-peer RTP	If click **YES**, 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.
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.
Advanced Settings	Select to see more optional settings shown below.
Selective Call Blocking	(Optional) Select **Block Anonymous Calls** to block all calls without a Caller ID.
	(Optional) Block one or more calling numbers by entering the calling numbers and clicking . Removing the blocked numbers by clicking the number from the list, and then click .
Forward Options	(Optional) Select **Unconditional Call Forward** and clicks a default destination in the list, e.g. Voicemail or Phone Number. ☞ If selecting Phone Number, 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.
Timeout To Next Forward	(Optional) Enter a period of time in seconds for rings the extension in Unavailable Call Forward. Click to add the extension in Unavailable Call Forward and the time here into the list. Remove the extension of Unavailable Call Forward from the list by clicking .
Play Unavailable Forward Prompt	(Optional) Notify the caller that callee is not available and the call is being forwarded to another extension.
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. ☞ If the function is enabled, the Line-in-use Call Back function will be disabled.
Selective Call Forward	(Optional) Unconditional Call Forwarding according to the calling number. Enters one or more calling numbers and a forwarding number, and clicks . E.g., forward only calls from 101 to a cellular number, while let the rest enter the voice mail by default. Selects a forwarding and click when the forwarding is no longer required.

4.3.10 Analog Phone

The ANALOG PHONE MANAGEMENT page lets the administrator to create analog phones. Select **Device** -> **Analog Phone**, and one can add, edit, and delete analog phones. Go to **Service** -> **IP PBX Service**, and click **RELOAD** to activate changes. Connect an analog phone to a FXS port and configure the properties of the port as detailed in **Table 4.3.3**.



Add an Analog Phone	1. Click **ADD** to see the detailed ANALOG PHONE MANAGEMENT page. 2. Enter settings shown in **Table 4.3.3**. 3. Click **ADD**.

	4. Click **BACK** to see the newly added analog phone in the **Extension Number**.
Edit an Analog Phone	1. Click a port in **POTS Port**. 2. Edit settings shown in **Table 4.3.3**. 3. Click **UPDATE**. 4. Click **BACK** to see the edit information.
Delete an Analog Phone	1. Select a **POTS Port**. 2. Click **DEL** to remove the extension from the **POTS Port**.

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 that the extension belongs to.
Unavailable Timeout	Timeout for ringing before a call is answered.
User[5]	Select a user that this extension associates with. ☞ If there is not any appropriate users to select, 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.
T.38 Enabled	Enable T.38 Fax-relay on this port when detecting fax tones in a call.
UDPTL Redundancy Level	Select number of the previous package(s) that will be sent again. This function only takes effect when T.38 is enabled.
Input/Output gain	Voice amplification or attenuation in dB scale to adjust input/output volume.

[5] Please refer to 4.1 for details.

Advanced Settings	Select to see more optional settings shown below.
Selective Call Blocking	(Optional) Select **Block Anonymous Calls** to block all calls without a Caller ID
	(Optional) Block one or more calling numbers by typing the calling numbers and clicking . Removing the blocked numbers by clicking the number from the list, and then click .
Forward Options	(Optional) Select Unconditional Call Forward and click a default destination in the list, e.g. Voicemail or Phone Number. ☞ If selecting Phone Number, 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.
Timeout Before Forward	(Optional) Timeout before trying next forwarding number in the list. Note that if the forwarded number has personal setting of forwarding policy, this timeout guards the total duration allowed before a call is connected by the personal setting. As long as the call does not go through, eventually it returns to the hunt list of forwardings.
Play Unavailable Forward Prompt	(Optional) Notify the caller that callee is not available and the call is being forwarded to another extension.
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 Forward	(Optional) Unconditional call forwarding according to the calling number. Enters one or more calling numbers and a forwarding number, and click . E.g., forward only calls from 101 to a cellular number, while let the rest enter the voice mail by default. Selects a forwarding and click when the 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 (including "-"), "*", "#", digit set, and wildcard characters like ".", "X", "Z", and "N". **Table 4.4.1** explains digit set and wildcard characters.

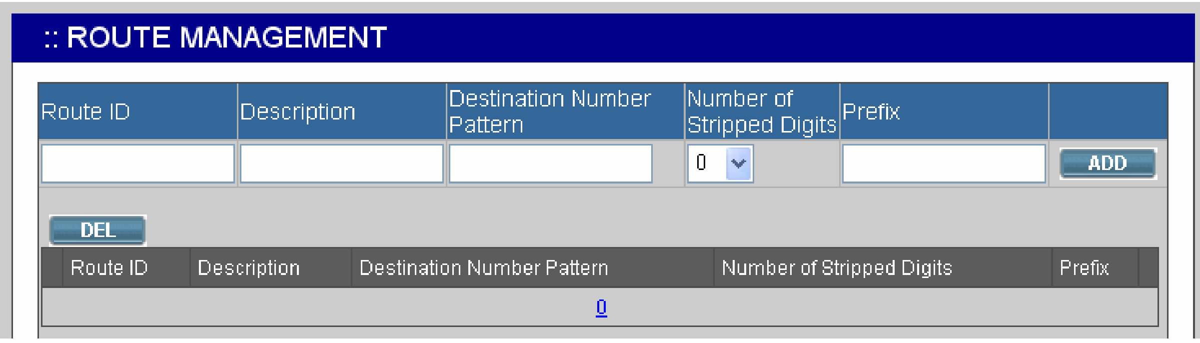


Table 4.4.1 Digit Set and Wildcard Characters for Route Patterns

Expression	Description
[<*digits*>]	Match any single digit listed explicitly. E.g., digit set [13579] match odd digits. One may use '-' to indicate a range of digits, e.g. [2-8].
. (dot)	Match any digit in any length. Usually given in the end of a pattern to include all numbers matched a specific prefix. ☞ . (dot) can not be used alone or at the beginning of the route patterns.
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 Management** -> **Route**, the administrator can add, edit, and delete routes in the Route Management page. Go to **Service** -> **IP PBX Service**, and click **RELOAD** to activate changes.

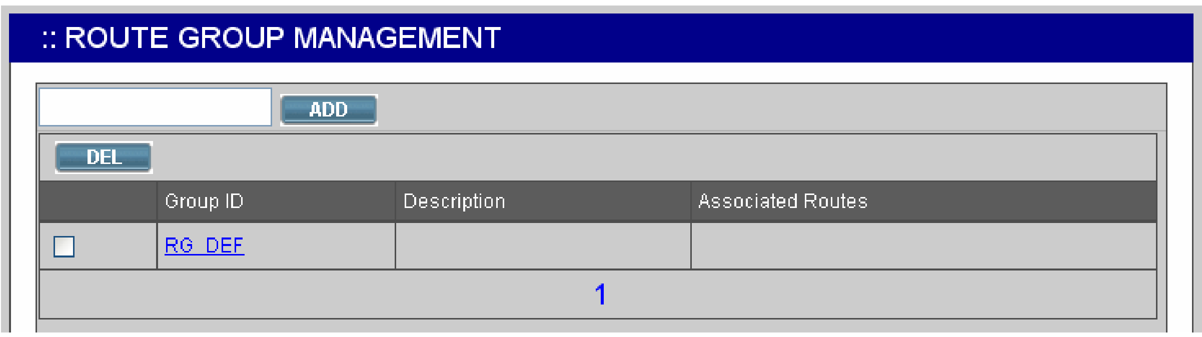
Add a Route	1. Enter settings shown in **Table 4.4.2**. 2. Click **ADD** to see the newly added route in the **Route ID**.
Edit a Route	1. Edit settings shown in **Table 4.4.2** in a row. 2. Click **APPLY** in the row to update the settings.
Delete a Route	1. Select a **Route ID**. 2. Click **DEL** to remove the route from the **Route ID**.

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 information.
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 Route Group 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 must be included into at least one routegroup in order to take the route pattern into effect.



Select **Route Management->** **Route Group**, and the administrator can add, edit and delete routegroups in the ROUTE GROUP MANAGEMENT page. Go to **Service -> IP PBX Service**, and click **RELOAD** to activate changes.

Add a Route Group	1. Type a route group name and click **ADD**. 2. Click the route group in **Group ID** to see the settings. 3. Enter settings shown in **Table 4.5**, and click **BACK**. The newly added route group should be displayed in the **Group ID**.
Edit a Route Group	1. Click a route group name in **Group ID**. 2. Edit settings shown in **Table 4.5**. 3. Click **SET**, if there is any update in the **Description** box. 4. Click **BACK** to see the updated information.
Delete a Route Group	1. Select a **Group ID**. 2. Click **DEL** to remove the route group from the **Group ID**.

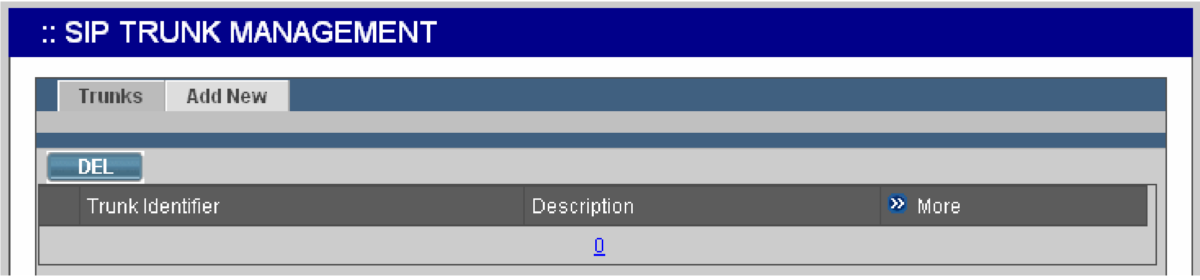
Table 4.5 Routegroup Configuration Settings

Field	Description
Group ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description information.
Associated Routes[6]	Select routes belonged to this routegroup. Click **ADD/DEL** button to add or remove a route to or from the routegroup. The right box lists current selected routes. Note the order of the selected routes is important since it decides which route would be matched first for an outgoing call. ☞ If there is no appropriate routes to select initially, one can come back later to revise it, once the expected routes are added.

[6] Please refer to 4.4 for details.

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 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 PLANET IP PBX. Select **Trunk -> SIP Trunk**, and one can add, edit, and delete SIP trunks. Go to **Service -> IP PBX Service**, and click **RELOAD** to activate changes.

Add a SIP Trunk	1. Click the **Add New** tab. **2.** Enter settings shown in 3. **Table 4.6**. 4. Click **ADD** to see the newly added SIP trunk in the **Trunk Identifier**.
Edit a SIP Trunk	1. Click the **Trunks** tab, and **More** to see more information. **2.** Edit settings shown in 3. **Table 4.6** in a row. 4. Click **APPLY** in the row to update the information.
Delete a SIP Trunk	1. Click the **Trucks** tab, and select a trunk identifier. 2. Click **DEL** to remove the SIP trunk from the **Trunk Identifier**.

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 information.
Dynamic Peer	Select 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 **SIP Proxy Port**	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.
Auth. Name	Specify the name for authentication if different to the **Trunk Identifier**.
Auth. Password	Give the password used for authentication on the remote SIP proxy or registrar. Usually this is given by the ITSP.
Registration Required	Select 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.
SIP Registrar **SIP Registrar Port**	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).
Outbound Routegroup[7]	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. ☞ If there is not any 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, clicks an extension in the list to be an unconditional destination for incoming calls to this trunk. Or click **bynumber** and then enter configurations in **DID Prefix** and **DID Stripping** to have the incoming calls directed to the corresponding extension derived by number manipulation. The SIP trunk numbers is therefore regarded as the direct line of the extension. ☞ If you set a DID extension in a trunk, then only that extension can use this trunk to call out, and all incoming calls to this trunk will connect to that extension directly.
DID Prefix	A digit string to be prefixed to the incoming called number

[7] Please refer to 4.5 for details.

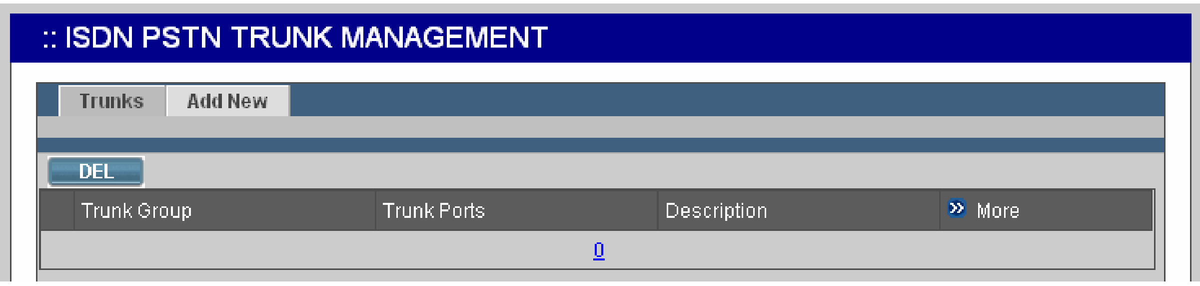
	after stripping.
DID Stripping	A number of leading digits to be stripped from the original called number. If prefix or stripping has been given but DID of Extension is not **bynumber**, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2nd dialing.
Language	Preferred language for system instructions heard from the trunk.
IVR List[8]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID.
Usergroup[9] of Privilege	When disabled DID, click a usergroup in the list whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this trunk. ☞ There may not be appropriate usergroups to select initially. One can come back later once the expected usergroup has been added.
Advanced Settings	Select to see more settings shown below.
DTMF Mode	Select a preferred DTMF mode, **RFC 2833** or **SIP INFO**, for this trunk in the list. This must match configuration on the server side. If the user does not know the DTMF mode on the server side, select **Not sure** from the list, and SDP will automatically detect the DTMF mode is Inband or RFC2833.
Try Peer-to-peer RTP	Click **NO** to disable or 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.
Bandwidth Limitation	Leave it blank to disable or, specifies a limit of bandwidth in kbps for call admission.

[8] Please refer to 0 for details.
[9] Please refer to 4.2 for details.

SIP Domain	Specify the SIP domain used by the proxy and registrar. If not specified, IP address will be used as the domain by default.
User-agent Content	Override default User-Agent header content.
Clear Bindings Prior Registration	Select if failed to the registration, and cannot identify any abnormal settings.
Disable NAT Traversal	IP PBX uses NAT traversal for outgoing traffics by default. Select to disable NAT traversal if there is a machine that could handle NAT issues.

4.7 ISDN PSTN Trunk Configuration

An ISDN PSTN trunk group is a logical group of one or more ISDN subscriber lines connecting to ISDN ports (RJ45) on PLANET IP PBX. Currently only Basic Rate Interface (BRI) ISDN service is supported. BRI consists of two 64 kb/s B channels and one 16 kb/s D channel for a total of 144 kb/s. This basic service is intended to meet the needs of most individual users.



The ISDN PSTN TRUNK MANAGEMENT page allows the administrator to configure ISDN trunks. Select **Trunk** -> **ISDN PSTN Trunk**, and one can add, edit and delete ISDN trunks. Go to **Service** -> **IP PBX Service**, and click **RELOAD** to activate changes.

Add an ISDN PSTN Trunk	1. Click the **Add New** tab. 2. Enter settings shown in **Table 4.7**. 3. Click **ADD** to see the newly added ISDN PSTN trunk in the **Trunk Group**. The newly added ISDN Trunk shall display in the **Trunk Group**.
Edit an ISDN PSTN Trunk	1. Click the **Trunks** tab, and **More** to see more information. 2. Enter settings shown in **Table 4.7** in a row. 3. Click **APPLY** in the row to update the information.

Delete an ISDN PSTN Trunk	1. Click the **Trunks** tab, and select a trunk group. 2. Click **DEL** to remove the ISDN PSTN trunk from the **Trunk Group**.

Table 4.7 ISDN Trunk Configuration Settings

Field	Description
Trunk Group	ID number of this ISDN trunk group. A valid number ranges from 1 to 31. It should not overlap with existing FXO PSTN trunk groups.
Trunk Ports	The **Trunk Ports** is the logical range of the sum of B and D channels. Each physical ISDN port occupies three Trunk Ports, two B and one D channels. User only needs to specify the B channel number here, since D channel is reserved in the 3rd trunk port for each physical ISDN port. E.g. Assume there are four ISDN ports in the PBX and no other FXO/FXS modules installed, then one can set each pair of numbers here, like 1,2 but excluding 3,6,9,11. ☞ If a four-port FXO/FXS module is also installed, then the Trunk Ports here should be numbered from 5 to 16 instead of 1 to 12. Make sure to specify the indices of ports correctly, or PBX will not start. One can refer to the **POTS Setting** page before configuration.
Description	Arbitrary description information.
Port Selection	Select to search for an available port in the group. **Rotating** means to force ports being selected by turns to even cost.
Signalling	Select **Point to point** or **Point to multipoint** depends on the link type between ISDN service provider and your device.
Switch Type	Supports European switch type by default.
Outbound Routegroup[10]	Selects a routegroup to associate routes with this trunk. Outbound calls match included route patterns could employ this trunk to access ISDN. ☞ There may not be any appropriate routegroup to select initially. One can come back later to revise it, once the expected routegroup is added.

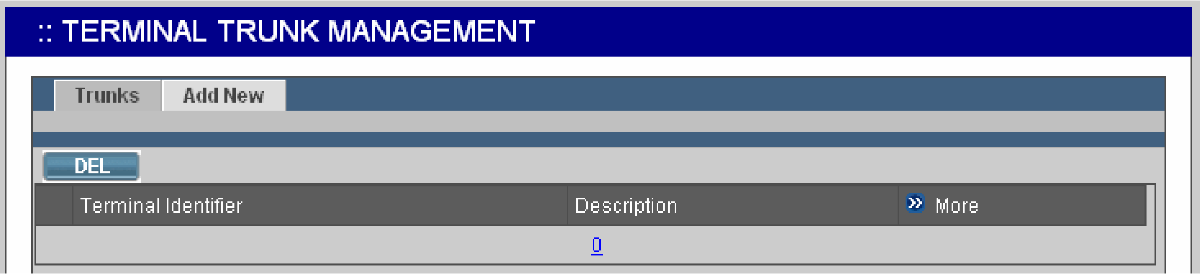
[10] Please refer to 4.5 for details.

DID of Extension	When enabled DID, selects an extension from the list to be an unconditional destination for incoming calls to this trunk. Or click **by number** and then enter configurations in **DID Prefix** and **DID Stripping** to have the incoming calls directed to the corresponding extension derived by number manipulation. The ISDN numbers of the included ports are therefore regarded as the direct line of the extension. ☞ If you set a DID extension in trunk, then only that extension can use this trunk to call out, and all other user's call in this trunk will connect to that extension.
DID Prefix	A digit string to be prefixed to the incoming called number after stripping.
DID Stripping	A number of leading digits to be stripped from the original called number. If prefix or stripping has been given but DID of Extension is not **bynumber**, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2nd dialing.
Language	Preferred language for system instructions heard from the trunk.
IVR List[11]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID.
Usergroup[12] **of Privilege**	When disabled DID, clicks a usergroup in the list whose reachability to other usergroups and trunks will use as the privilege of inbound calls from this trunk. ☞ There may not be any appropriate usergroups to select initially. One can come back later to revise it, once the expected usergroups are added.
Caller ID	Leaves it blank to have the default caller ID, or enters a caller ID that is provided by your ISDN service provider.

[11] Please refer to 0 for details.
[12] Please refer to 4.2 for details.

4.8 Terminal Trunk Configuration (IPX-2000, IPX-1803 and IPX-1804 only)

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 terminal trunk is the account for a SIP trunk to authenticate with, exact the same identifier and password must be used for both.



The TERMINAL TRUNK MANAGEMENT page allows the administrator to configure trunk terminals used by PLANET IP PBX. Select **Trunk** -> **Terminal Trunk**, and one can add, edit and delete terminals. Go to **Service** -> **IP PBX Service**, and click **RELOAD** to activate changes.

Add a Terminal Trunk	1. Click the **Add New** tab. 2. Enter settings shown in **Table 4.8**. 3. Click **ADD** to see the newly added terminal trunk in the **Terminal Identifier**.
Edit a Terminal Trunk	1. Click the **Trunks** tab, and **More** to see more information. 2. Edit settings shown in **Table 4.8** in a row. 3. Click **APPLY** in the row to update the information.
Delete a Terminal Trunk	1. Click the **Trunks** tab, and select a terminal identifier. 2. Click **DEL** to remove the terminal trunk from the **Terminal Identifier**.

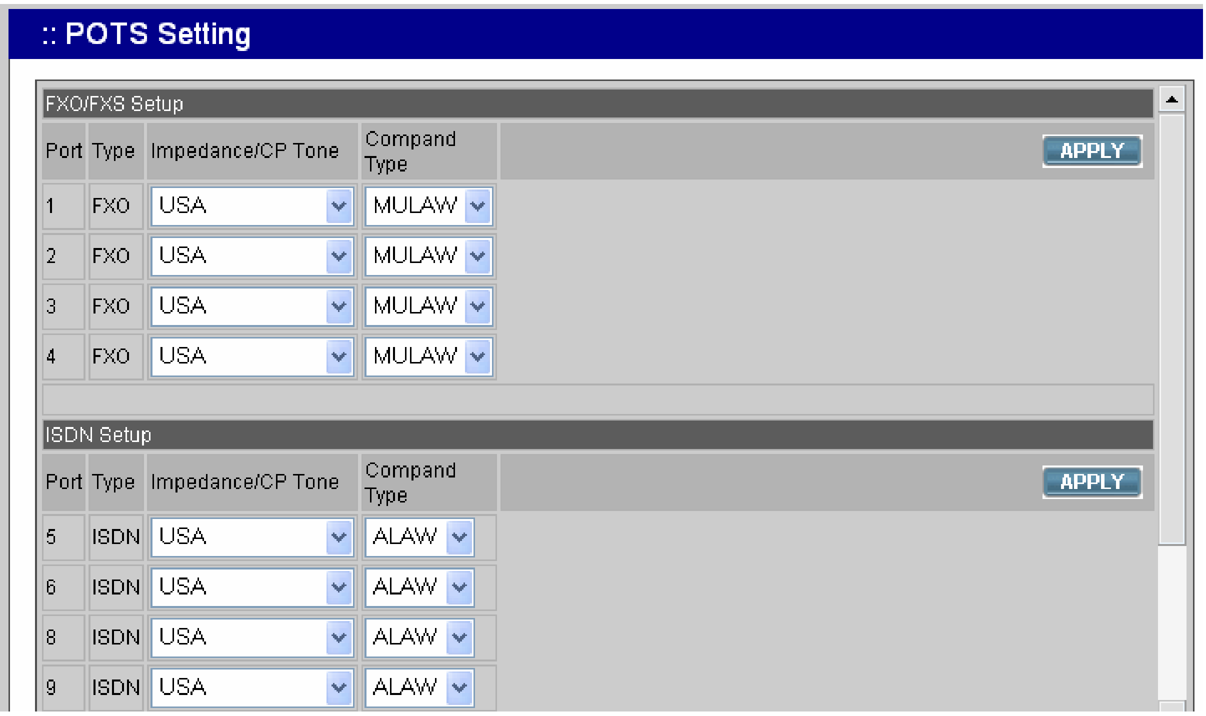
Table 4.8 Trunk Terminal Configuration Settings

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 information.

Terminal Password	Password of SIP trunk given on the other IP PBX for authentication.
Language	Preferred language for system instructions heard from the terminal.
Usergroup[13] of Privilege	When disabled DID, click a usergroup in the list whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this terminal. ☞ There may not be any 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.
Bandwidth Limitation	Leaves this blank to disable or, specifies a limit of bandwidth in kbps for call admission.

4.9 POTS Setting (IPX-2000, IPX-1803 and IPX-1804 only)

This page allows selection of country-based progress tones and/or impedance and/or compand type of POTS ports. Click **APPLY** to save modifications. Go to **Service** -> **IP PBX Service**, and click **RESTART** to active new settings.



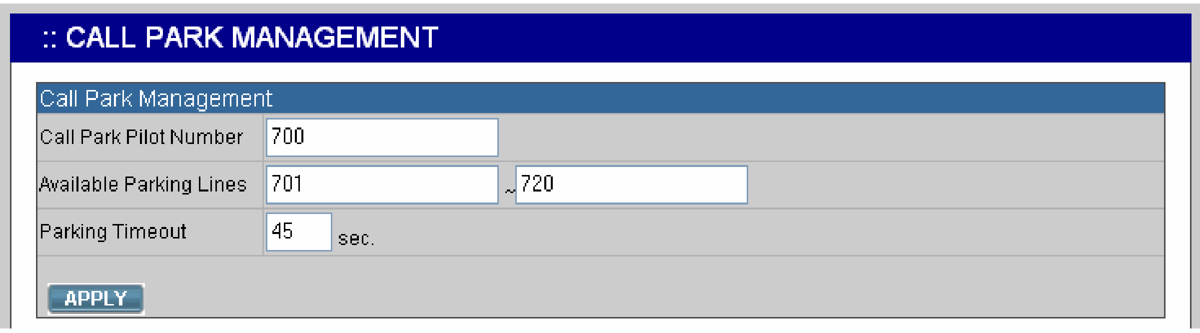
13 Please refer to 4.2 for details.

5 Feature Configuration

A feature is a logical entity presenting a function module of IP PBX, e.g. meet-me conference, auto attendant, voice mail, music on hold, etc. Any configuration change to a feature requires clicking **RELOAD** in **Service** -> **IP PBX Service** to take effect.

5.1 Call Park

During a call, the callee may want to continue the conversation using another phone. The call park feature enables so by letting the callee transfer the call to the call park pilot number. IP PBX will respond an available park line from the pool of call park numbers to the callee. After that the callee may hang up current phone, move to another phone, and dial 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 retrieve his call before timeout, IP PBX will ring the original extension where the callee answered the call. To configure Call Park feature, select **Feature** -> **Call Park**.



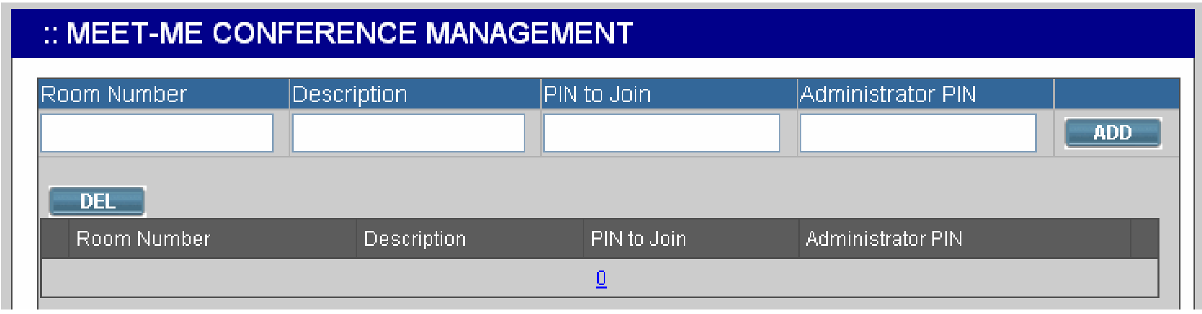
1. Enter settings shown in **Table 5.1**.
2. Click **APPLY**.

Table 5.1 Call Park Configuration Settings

Field	Description
Call Park Pilot Number	A unique extension number for call parking, e.g. 700.
Available Parking Lines	An extension pool for call parking, e.g. 701-720 forms a 20-line pool available for system to park calls.
Parking Timeout	Timeout waiting for picking up the parked call

5.2 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 behind another IP PBX. PLANET IP PBX allows multiple conference rooms going concurrently using different room numbers. Before entering a meeting room, the caller has to enter the correct PIN of the room number.



Note: The administrator who invited another meet-me conference room must drop all parties by pressing *5 when the meeting ends.

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

Add a Meet-me Conference	1. Enter settings shown in **Table 5.2**. 2. Click **ADD** to add a new conference room. The newly added room should display in the **Room Number**.
Edit a Meet-me Conference	1. Edit settings shown in a row. 2. Click **APPLY** at the end of the row to update the information.
Delete a Meet-me Conference	1. Select a room number. 2. Click **DEL** to remove the conference room from the **Room Number**.

Table 5.2 Meet-me Conference Configuration Settings

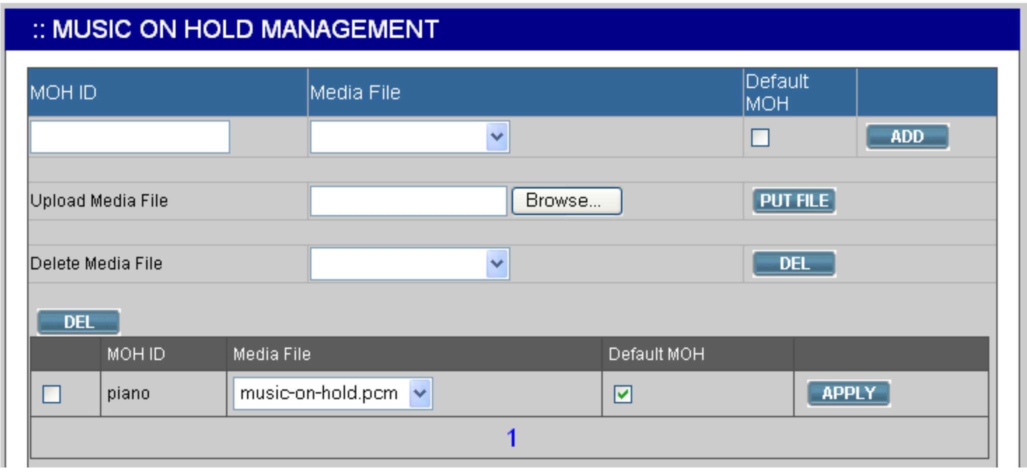
Field	Description
Room Number	Meeting room number, e.g. 8000.
Description	Arbitrary description information.
PIN to Join	PIN for normal users to join the conference. During a conference, a normal user has following options: - # to quit conference - *1 to mute/unmute - *9 to log in as the administrator if there is no administrator dialed in yet.
Administrator PIN	PIN for the administrator of the conference.

	During a conference, the administrator has following options: - # to quit conference - *1 to mute/unmute - *2 to lock/unlock the conference - *3 to invite a user into the conference - *4 to drop a party from the conference - *5 to drop all parties in the conference - *6 to drop the last invited party by *3 - ** to send DTMF string to the last invited party by *3. This is useful when the invited party is behind an IVR system.

5.3 Music On Hold

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

Select **Feature** -> **Music On Hold** to manage MOH files.



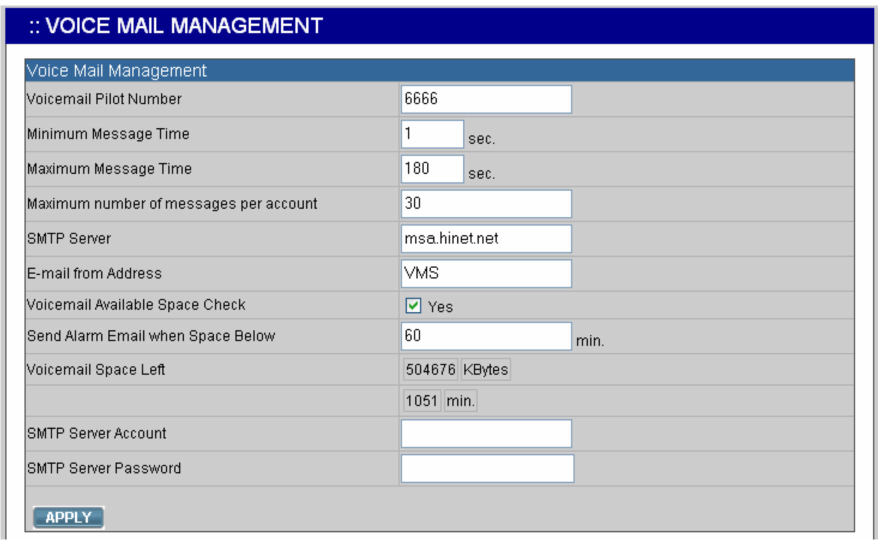
Add a MOH File	1. Enter settings shown in **Table 5.3**. 2. Click **ADD** to see the newly added file in the **MOH ID**.
Edit a MOH File	1. Edit settings shown as a table at the bottom of the page. 2. Click **APPLY** in the row.
Delete a OH File	1. Select a MOH ID. 2. Click **DEL** at the top-left the table to remove the MOH file from the **MOH ID**.

Table 5.3 MOH file Configuration Settings

Field	Description
MOH ID	A unique ID containing only alphabets, numbers, and underscore without spaces; 32 characters maximum.
Media File	Candidate music files in the repository. To upload a new music file, click **Browse** to find a Windows PCM (8000 Hz, 16-bit) file from the local host and click **PUT FILE**. On successful uploading, the filename will appear in the **Media File** list. To delete a media file from the list, choose a file from the **Delete Media Fil**e list, and click **DEL** to remove it.
Default MOH	Select to use this music file for system default MOH globally.

5.4 Voicemail

PLANET IP PBX has a built-in voice mail subsystem with a sophisticated IVR menu. 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 IP phones (if any) will be lit. For analog phones, the user will hear six short beeps before the normal dial tone when picking up the analog phone. 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. In addition to indicating current voice mail capacity on the management page, IP PBX can send an alarm email to the administrator when the available voice mail space reaches the threshold. To configure Voicemail feature, select **Feature** -> **Voicemail**.



1. Enter settings shown in **Table 5.4**.
2. Click **APPLY**.

Table 5.4 Voice Mail Configuration Settings

Field	Description
Voicemail Pilot Number	Number to access voice mail system IVR.
Minimum Message Time	Messages less than this duration will not be notified by e-mail. E.g., 3 (sec).
Maximum Message Time	Maximum duration allowed for a single message. E.g., 60 (sec).
Maximum number of 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.
Voicemail Available Space Check	Select to enable the Alarm Email function described below.
Send Alarm Email when Space Below	Set a threshold in minutes to send an alarm email to the administrator when the space left is below it.
Voicemail Space Left	Show the available space in Kbytes and minutes. ☞ The storage inside IP PBX saves not only voice mails but also some other stuff, such as CDR and logs. The remained disk space is all for voice mails, and it is the "maximum" available voice mail space.
SMTP Server Account	Specify an account ID if the SMTP server requires authentication for outgoing mails.
SMTP Server Password	Specify the account password if the SMTP server requires authentication for outgoing mails.

5.5 Meet-me Prompts

This page allows replacing built-in meet-me conference prompts with user recordings.

1. Click a language and a prompt in the corresponding lists.
2. Find a corresponding recording in the local storage.
3. Click **PUT FILE** to complete the replacement.
4. To reset a prompt back to default, leave the **Upload** box in blank and directly click the **PUT FILE**.

Note that the replacement is done for the selected Language and Prompt only. Currently only following prompts could be replaced.

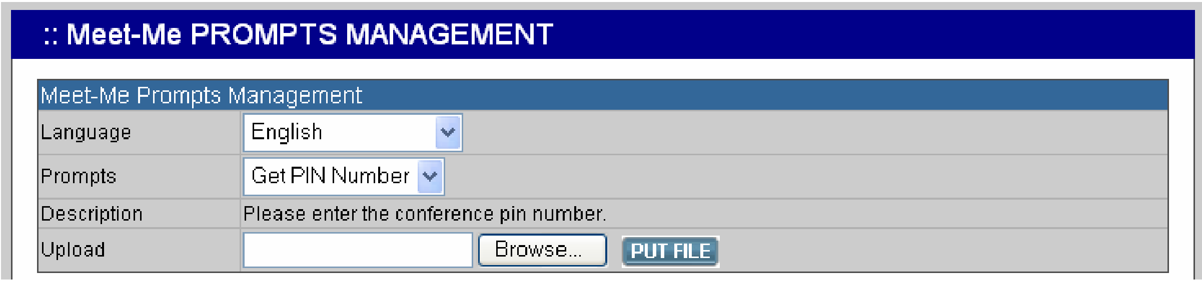


Table 5.5 Replaceable Meet-me Prompts

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.

5.6 Voicemail Prompts

This page allows replacing built-in voicemail system prompts with user recordings.

1. Click a language and a prompt in the corresponding lists.
2. Find a corresponding recording in the local storage.
3. Click **PUT FILE** to complete the replacement.
4. To reset a prompt back to default, leave the **Upload** box in blank and directly click **PUT FILE**.

Note that the replacement is done for the selected Language and Prompt only. Currently only following prompts could be replaced.

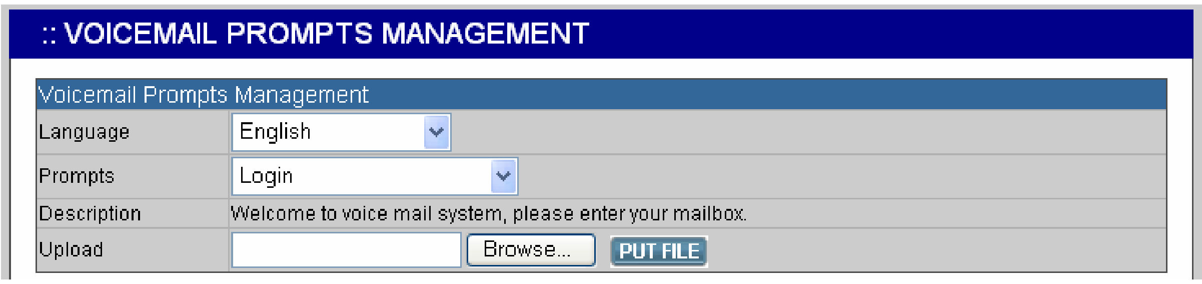
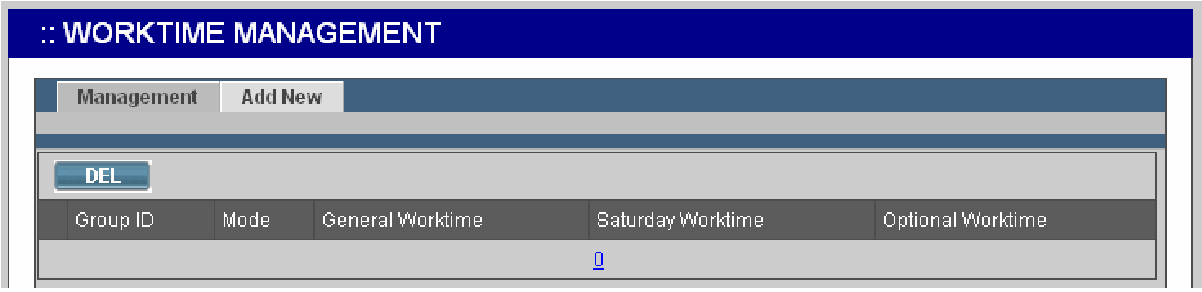


Table 5.6 Replaceable Voicemail System Prompts

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

5.7 Worktime

Worktime defines holidays and business hours for generic IVR application. Several groups of date/time could be defined for different IVR menus. Select **Feature** -> **Worktime** to configure Worktime features.



Add a Worktime	1. Click the **Add New** tab. 2. Enter settings shown in **Table 5.7**. 3. Click **ADD** at the bottom of the page. The newly added worktime should display in the **Group ID**.
Edit a Worktime	1. Click the **Management** tab. 2. Click a **Group ID**. 3. Edit settings shown in **Table 5.7**. 4. Click **UPDATE** to update the information.
Delete a Worktime	1. Click the **Management** tab. 2. Select a **Group ID**. 3. Click **DEL**. The deleted worktime shall disappear from the **Group ID**.

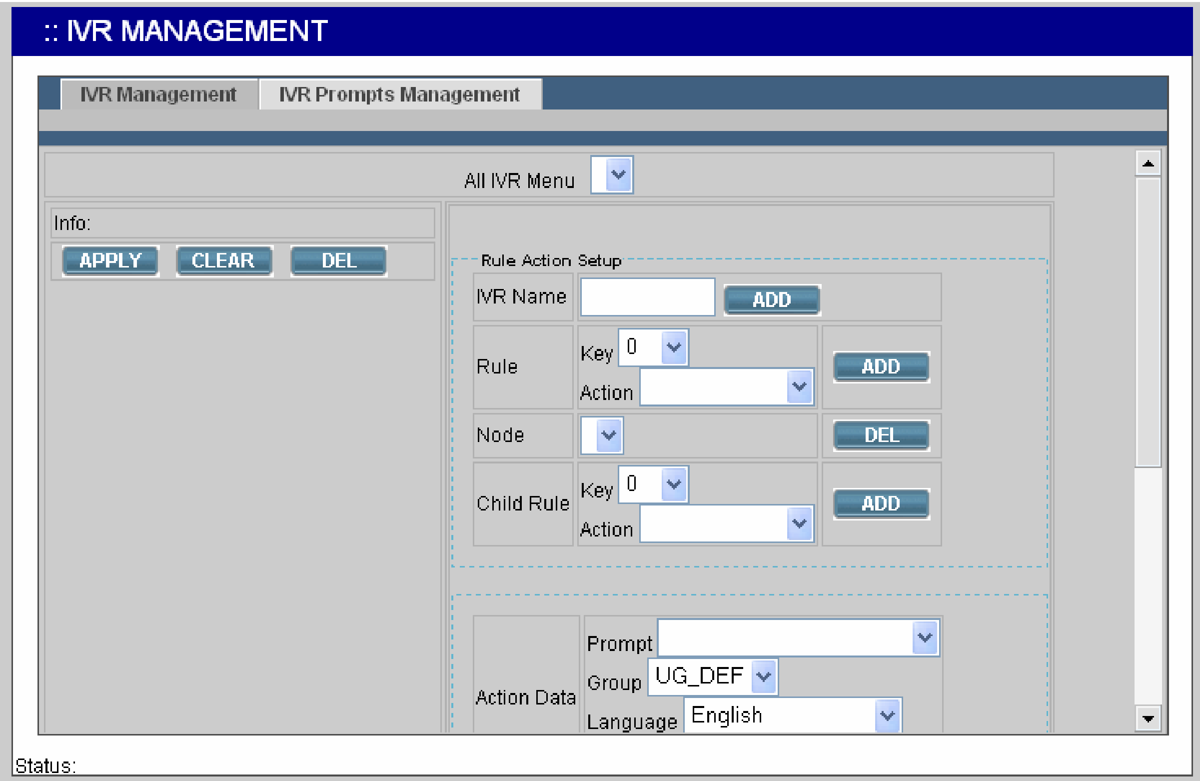
Table 5.7 Worktime Configuration Settings

Field	**Description**
Group ID	A unique ID containing numbers only.
Mode	Select one of the three modes: **1:** No work on weekends. **2:** Work off and on by turns on Saturdays. **3:** Work half-day on Saturdays.
General Worktime	The work time from Monday to Friday.
Saturday Worktime	The work time for Saturdays, this field only active when mode is set to 2

	or 3.
Optional Worktime	Special holidays or work day. User can set date and its work time, or set it to a whole-day holiday.

5.8 *Interactive Voice Response (IVR)*

Interactive Voice Response (IVR) helps a caller to select options from voice menus by pressing keys on a telephone keypad. With IVR, a caller can connect to an expected extension or a service promptly. PLANET IP PBX enables multiple configurable IVR menus in a single system, and each of them could have a hierarchy up to three layers. Select **Feature** -> **IVR** to add, edit and delete the IVR menus. You can also manage IVR prompts, used by IVR menus, in this page.



Add a new IVR Menu	1. Enter a name of an IVR menu in **IVR Name**, and click a file in the **Prompt** list. 2. Click **ADD** next to the **IVR Name** box to set the new IVR name in **Info**. System will prompt to ask for confirmation whether a Worktime setting is required or not. This is because Worktime setting can only be added when creating a new IVR. After creation, an IVR without Worktime setting cannot be associated with a Worktime setting later. If Worktime setting is indeed not required,

	click **Cancel** in the pop-up window. 3. Enter settings shown in **Table 5.8**. 4. Click **APPLY** to add the new IVR menu and see it as a tree view in **Info**. 5. For example, to create a basic Auto Attendant IVR for a trunk with Usergroup of Privilege *dial_in*: • Enter an **IVR Name**, say *Basic_AA* • Choose **/agent-newlocation.gsm* from **Prompt** list in **Action Data** block. • Choose a usergroup from **Group** under **Action Data**. • Click the **ADD** next to the **IVR Name** box. • Click **Cancel** in the pop-up window to confirm the Worktime setting is not required. • Now, *Basic_AA* should be available in the **IVR list** of Trunk pages.
Edit an IVR Menu	1. Click an IVR name in the **All IVR Menus** list. 2. Edit settings shown in **Table 5.8**. 3. Click **APPLY** to update the changes.
Delete an IVR Menu	1. Click an IVR name in the **All IVR Menus** list. 2. Click **DEL** to delete the IVR menu.

Table 5.8 Interactive Voice Response Configuration Settings

Field	Description	
All IVR Menus	Select a preferred IVR menu name.	
Info	View the IVR menu as a tree view.	
IVR Name	Specify the name of the IVR.	
Rule	Click a number in the **Keypad** list and one of the following actions in the **Action** list to associate an action with a key.	
	Hang Up	To cut off the call immediately.
	Play Back	To play the IVR prompt selected in **Prompt** list
	Call To	To call an extension.
	Go to Top	To go back to the root menu of the IVR.
	Next Layer	To go to the next layer of the IVR menu.
	Select Language	To choose a language.
	Return	To go back to the previous layer.

<table>
<tr><td>Node</td><td colspan="2">Information of the configured keys and actions. Click a node and DEL to delete the node and its underlying structure.</td></tr>
<tr><td>Child Rule</td><td colspan="2">If a Next Layer is selected, Child Rule sets the key-action associations with the next-layer menu.</td></tr>
<tr><td rowspan="5">Action Data</td><td colspan="2">Specify applicable parameter(s) for an action.</td></tr>
<tr><td>Prompt</td><td>Select a *.wav recording file that you add from the IVR Prompt tab, or select one of the following default voice file. The default file that marked */ in front of the file name means this voice file provides all languages that IP PBX has for you to select. Please click a language in the Languages list.
agent-newlocation.gsm: Please enter a new extension followed by the # key.
auth-thankyou.gsm: Thank you.
invalid.gsm: I'm sorry, that is not a valid extension, please try again.
transfer.gsm: Please hold while I try out that extension.
ss-busy.gsm: System is busy at this moment, please try again later.
ss-noservice.gsm: The number you have dial is not in service, please check the number and try again.
vm-goodbye.gsm: Good bye.
vm-sorry.gsm: I'm sorry, I do not understand you response.</td></tr>
<tr><td>Group</td><td>Select a usergroup.</td></tr>
<tr><td>Language</td><td>Select a language of the IVR.</td></tr>
<tr><td>Extension</td><td>Enter an extension number to be transferred to.</td></tr>
<tr><td>Active Worktime</td><td colspan="2">Select to set work time for the IVR.</td></tr>
<tr><td>Group</td><td colspan="2">Select a work time group set in Feature -> Worktime.</td></tr>
<tr><td rowspan="4">In-Hour Actions</td><td colspan="2">Select one action during business hours.</td></tr>
<tr><td>Play Back</td><td>To play the selected prompt.</td></tr>
<tr><td>Call To</td><td>To transfer to an extension</td></tr>
<tr><td>No Action</td><td>No action.</td></tr>
<tr><td>Prompt</td><td colspan="2">Select a *.wav file if Playback is selected in the In-Hour Actions list.</td></tr>
<tr><td>Extension</td><td colspan="2">Enter an extension number if Call To is selected in the In-Hour Actions</td></tr>
</table>

	list.	
Off-Hour Actions	Select one action during the off hours.	
	Play Back	To play the selected prompt.
	Call To	To transfer to an extension
	No Action	No action.
Prompt	Select a *.wav file if **Playback** is selected in the **Off-Hour Actions** list.	
Extension	Enter an extension number if **Call To** is selected in the **Off-Hour Actions** list.	

5.8.11 IVR Prompts Management

One can upload customized IVR prompts in **Feature** -> **IVR,** and click **IVR Prompts Management** tab.

Add an IVR Prompt	1. Select a language from the **Language** list. 2. Click **Browse** to find the expected recording in the local storage. 3. Click **PUT FILE** to upload the file add it to the **Prompt** list.
Delete an IVR Prompt	1. Select a *.wav file from the **All Files** list. 2. Click **DEL**. The deleted file shall disappear from the **All Files** list.

6 Voice communication samples

There are several ways to make calls to desired destination in IPX-1800N. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

6.1 Voice communication via IP PBX system – IPX-1800N

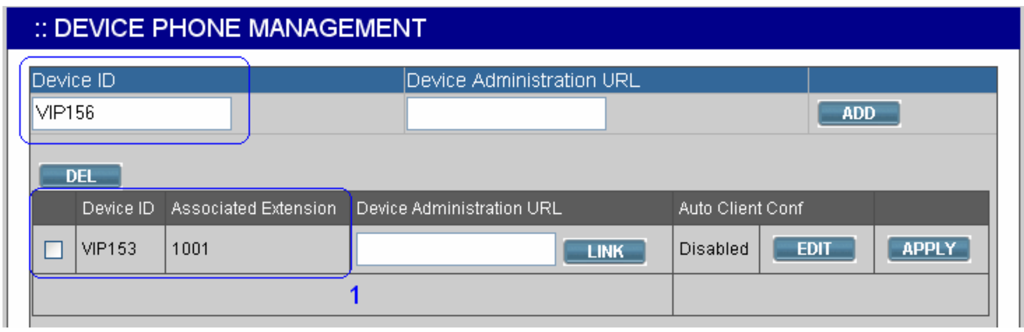
In the following sample, we'll introduce how to integrate the client with our IP PBX system IPX-1800N via general settings.

- VIP-153PT IP Address: 192.168.1.2
  Line Number: 1001
- VIP-156 IP Address: 192.168.1.3
  Line Number: 2002

Machine configurations on the IPX-1800N

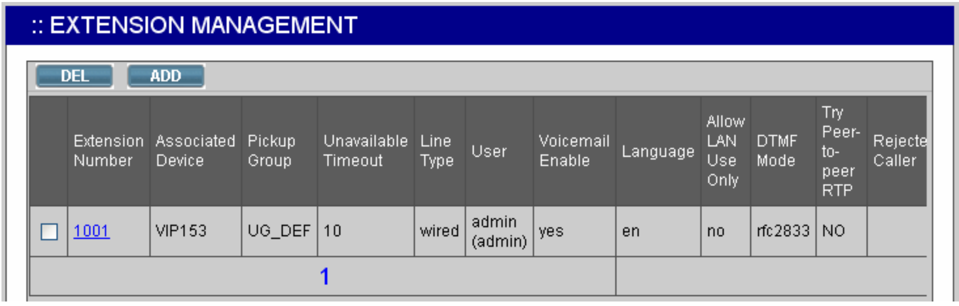
STEP 1:

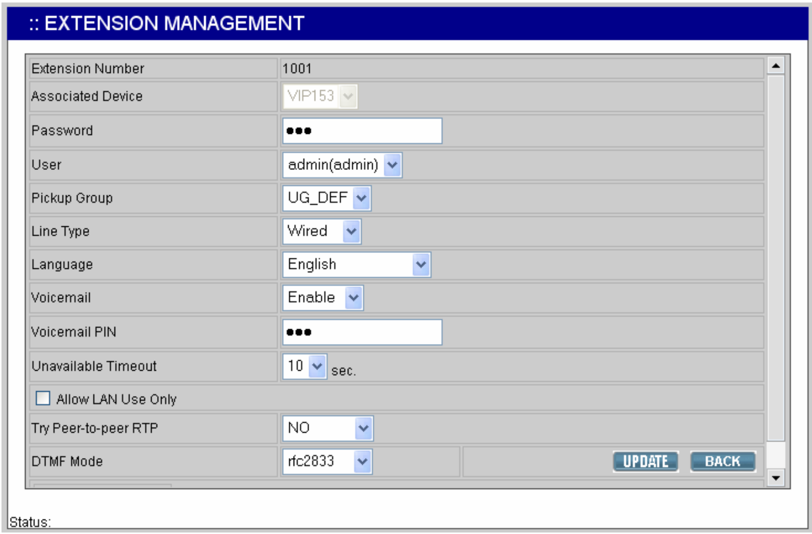
Please browse to the **"Device → IP Phone"** menu and create new device for the general configuration.



STEP 2:

Please browse to the **"Device → Extension of IP Phone"** menu and press the **ADD** button to create the two extension accounts/password: 1001/123 (for VIP-153PT), and 2002/123(for VIP-156) for the voice calls.

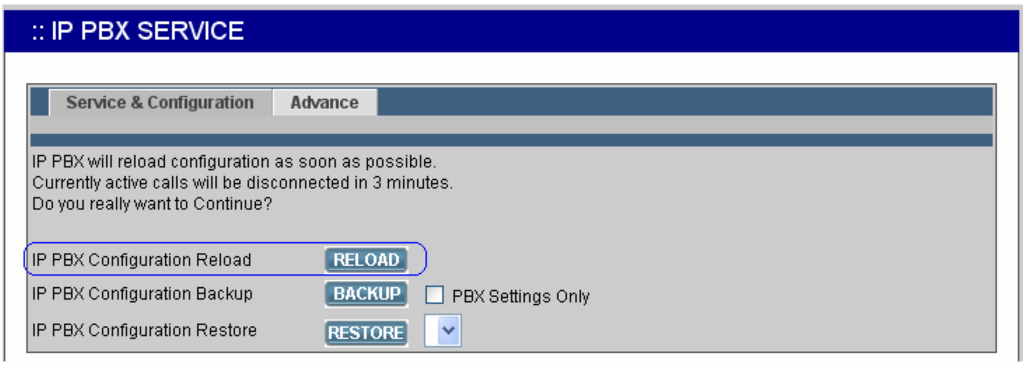




STEP 3:

After setting up the parameters, please refer to the path to activate the settings:

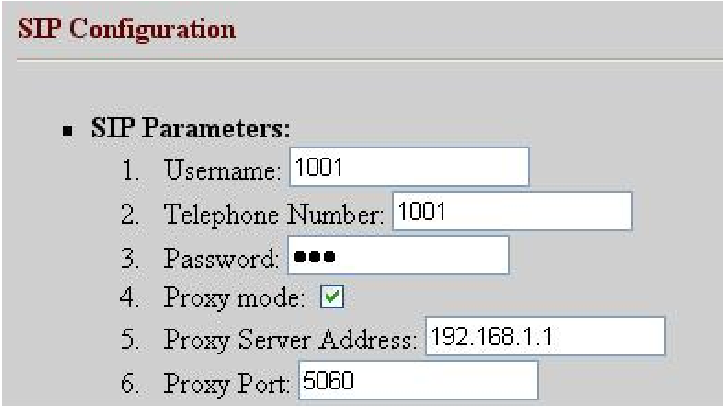
Service ---> **IP PBX service** ---> **IP PBX Configuration Reload**



Machine configurations on the VIP-153PT

STEP 1:

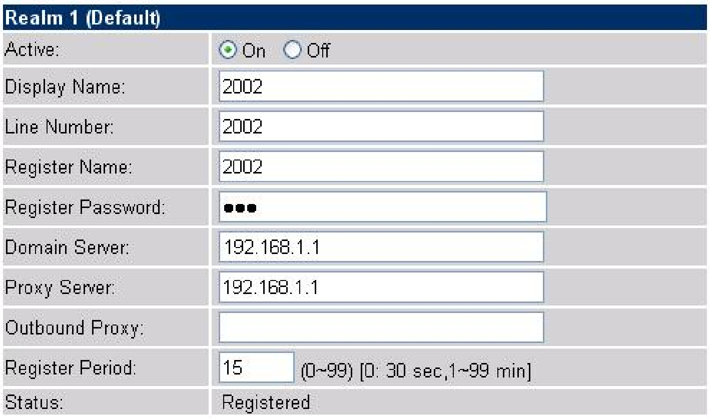
After creating accounts on the IP PBX system, please log in VIP-153PT via web browser, browse to the **SIP Configuration**, and refer to the account settings of the IP Extension to complete the SIP parameters. After these configurations, be sure to click the "**DONE**" button to apply settings and browse to "**System Configuration**" menu to reboot the machine to make the settings effective.



Machine configurations on the VIP-156

STEP 1:

Please log in VIP-156 via web browser, browse to the **SIP Settings** menu. In the setting page, please browse to the **Service Domain** page, and insert the SIP parameters for IP PBX system.



Test the scenario:

To verify the VoIP communication, you may make calls from extension side (VIP-153PT) 1001 to the number 2002 (VIP-156) or reversely make calls from extension client (VIP-156) 2002 to the number 1001 (VIP-153PT)

6.2 Voice communication via IP PBX system – IPX-1800N (Auto-config)

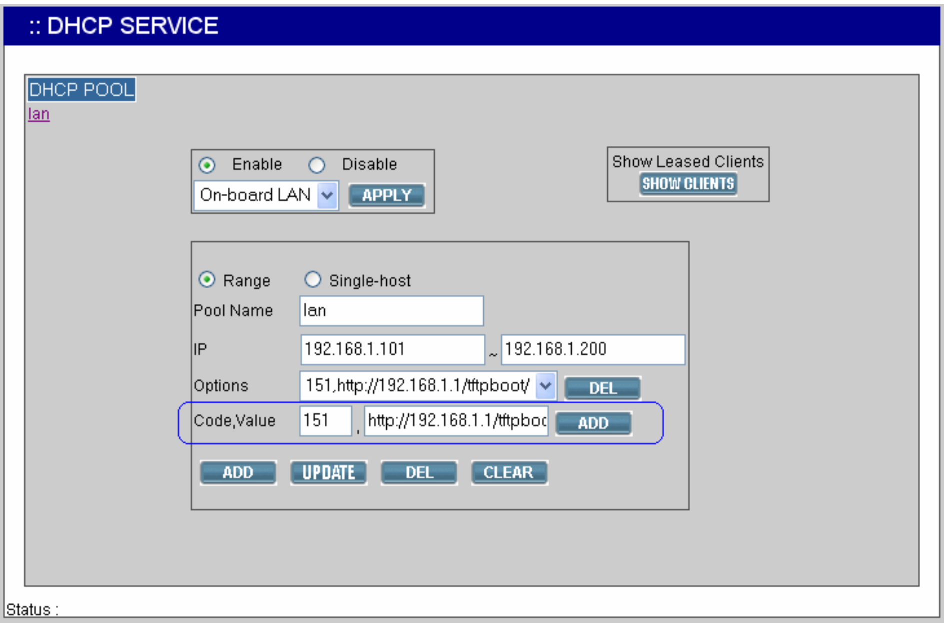
In the following sample, we'll introduce how to integrate the client with our IP PBX system IPX-1800N via Auto-config feature.

- VIP-153PT IP Address: 192.168.1.2
  Line Number: 1001
- VIP-156 IP Address: 192.168.1.3
  Line Number: 2002

Machine configurations on the IPX-1800N

STEP 1:

Log in IPX-1800N and browse to the DHCP menu and create new options list for the auto configuration.



Code: please insert 151 as the DHCP server option.

Value: http://LAN IP for IPX-1800N/tftpboot

If you'd like to enable auto-config for IP extension features in IPX-1800N, please be sure to setup the DHCP option code and the value information.

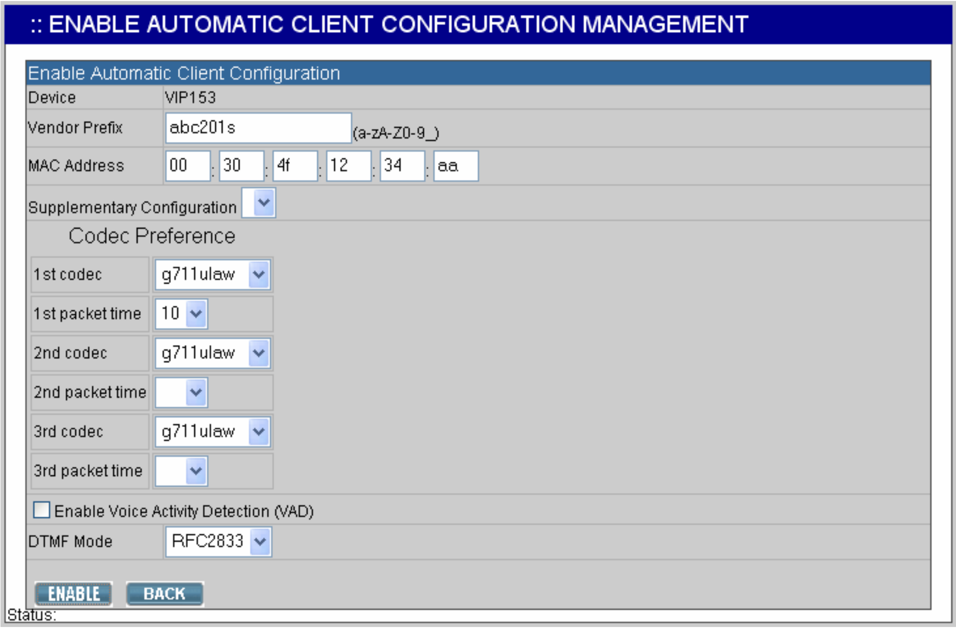
In most case, insert the optional code 151 and the value=http://192.168.1.1/tftpboot/

ⓘNote

- 192.168.1.1 is the IP address of IPX-1800

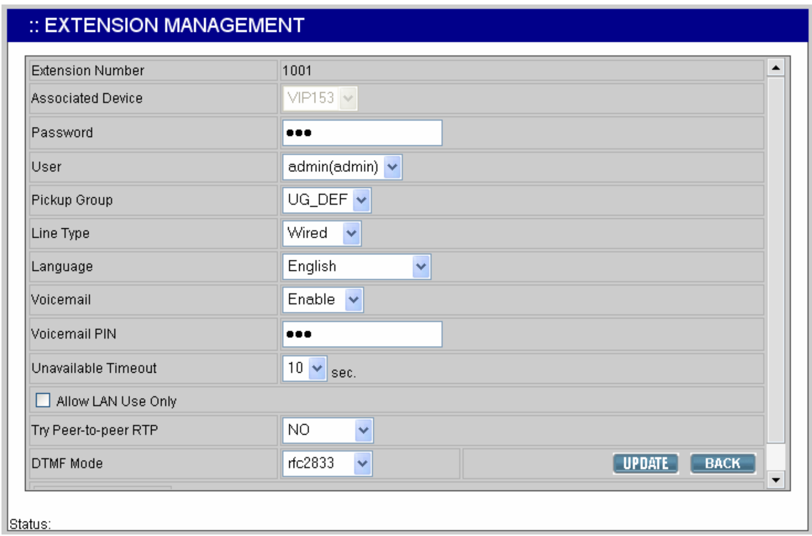
STEP 2:

Please browse to the Device → IP Phone menu and create new device for the auto configuration.



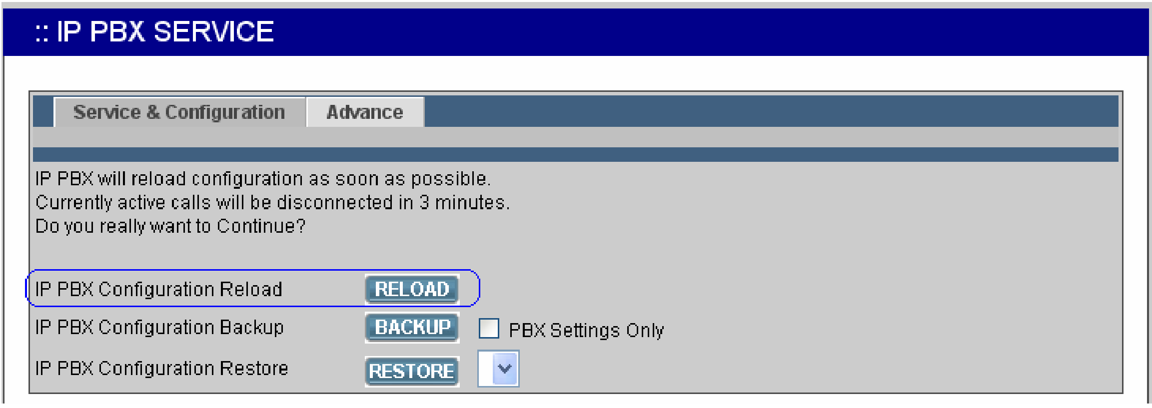
STEP 3:

Please press the Show extensions button to create the two extension accounts/password: 1001/123 (for VIP-153PT), and 1002/123(for VIP-156) for the voice calls.



STEP 4:

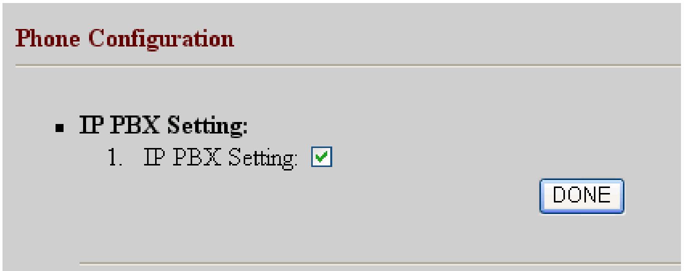
After setting up the parameters, please refer to the path to activate the settings: **Service ---> IP PBX service ---> IP PBX configuration reload**



Machine configurations on the VIP-153PT

STEP 5:

Please log in VIP-153PT via web browser, please browse to the Phone Configuration page, and enable the IPX PBX setting features for IP PBX system. After these configurations, be sure to click the "DONE" button to apply settings and browse to "System Configuration" menu to reboot the machine to make the settings effective.



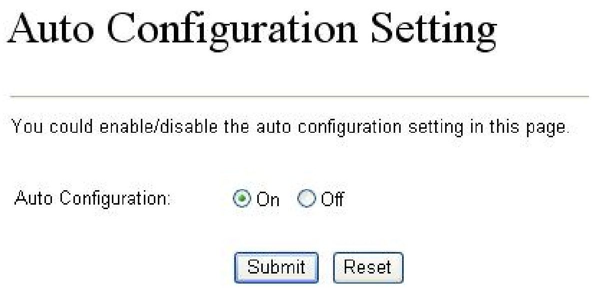
STEP 6:

After enabling the Auto-config feature, the VIP-153PT shall be able to obtain IP address and SIP extension information from IP PBX system IPX-1800N information. The VIP-153PT will perform registration to IPX-1800N after obtaining the extension config file.

Machine configurations on the VIP-156

STEP 7:

Please log in VIP-156 via web browser, browse to the Advanced Settings menu. In the setting page, please browse to the Auto-config page, and enable the Auto Configuration features for IP PBX system. (Your may connect telephone set to VIP-157, press #136 to enable the Auto configuration, or press #137 to disable the Auto Configuration setting.)



STEP 8:

After enabling the Auto-config feature, the VIP-156 shall be able to obtain IP address and SIP extension information from IP PBX system. To verify the auto-config results, you may connect telephone set to VIP-156; press #120# to check if the IP address is obtained from IPX-1800N. And #122# can be used to verify the extension number assigned by IPX-1800N.

Test the scenario:

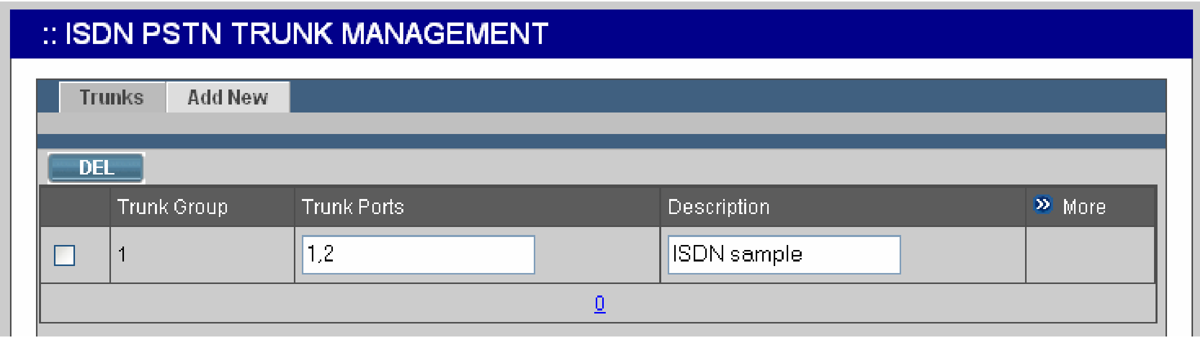
To verify the VoIP communication, you may make calls from extension side (VIP-153PT) 1001 to the number 1002 (VIP-156) or reversely make calls from extension client (VIP-156) 1002 to the number 1001 (VIP-153PT)

6.3 ISDN PSTN Trunk Procedure:

STEP 1:

Please browse to "**ISDN PSTN trunk**" page in "**Trunk**" menu, and refer to the following configuration steps for more understandings:

Press **<Add new>** button from the left panel to add a new ISND PSTN trunk.



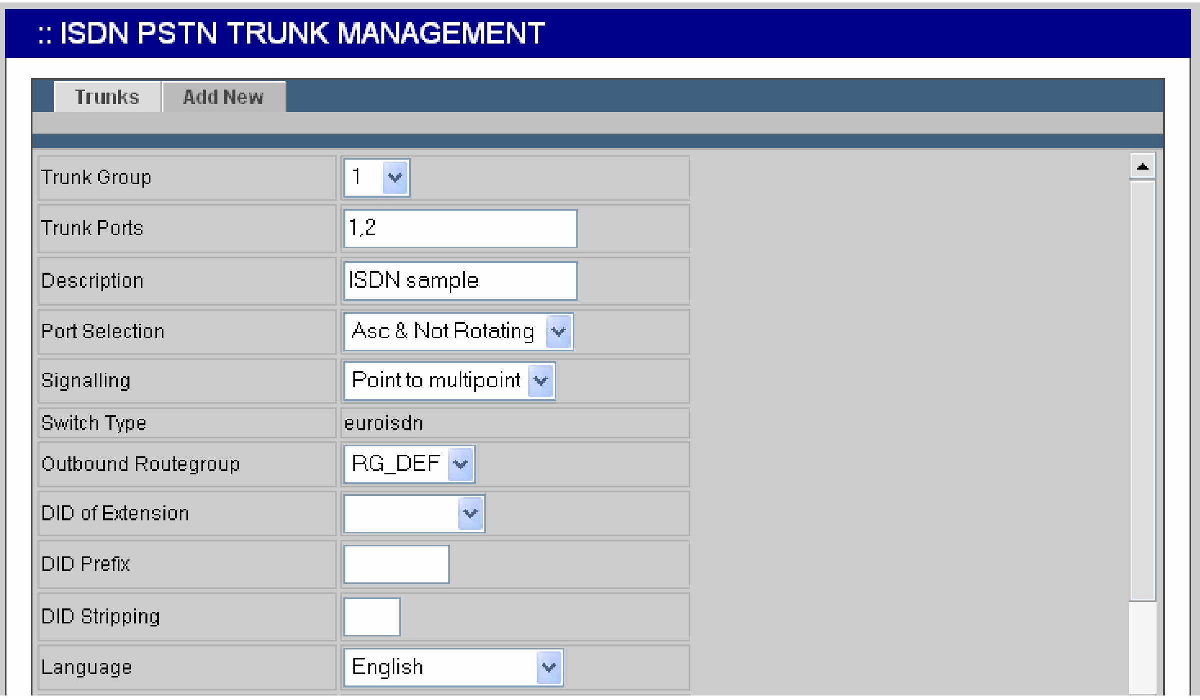
For example:

Trunk Group =**1**

Trunk ports = **1,2**

Port Selection = **Asc & Not Rotating**

Signalling = **Point to multipoint**



STEP 2:

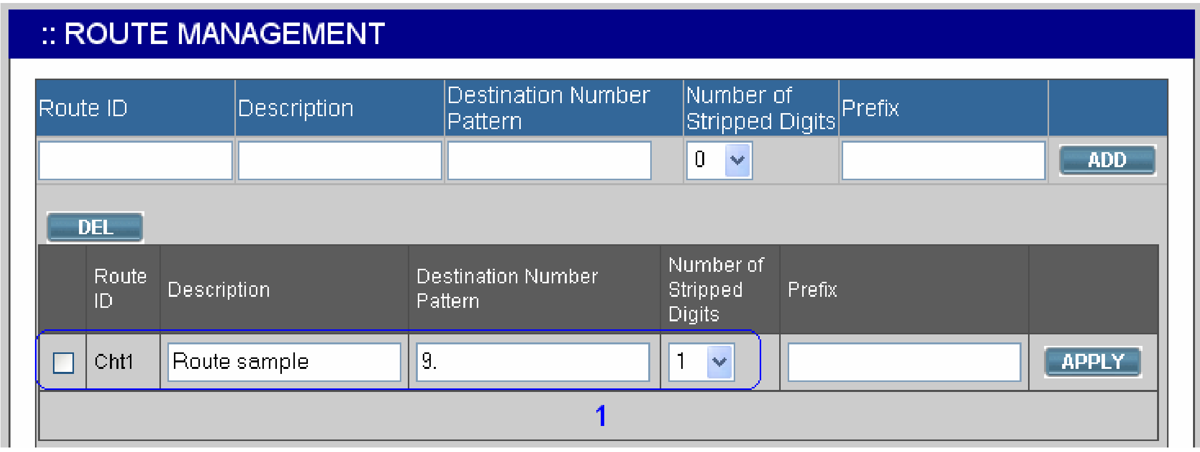
a) Please browse to "**Route management**" page in "**Route**" menu to add routes ID in IP PBX system.

b) Press **<Add new>** button from the left panel to add a new routes table and Insert following data:

Route ID: a unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.

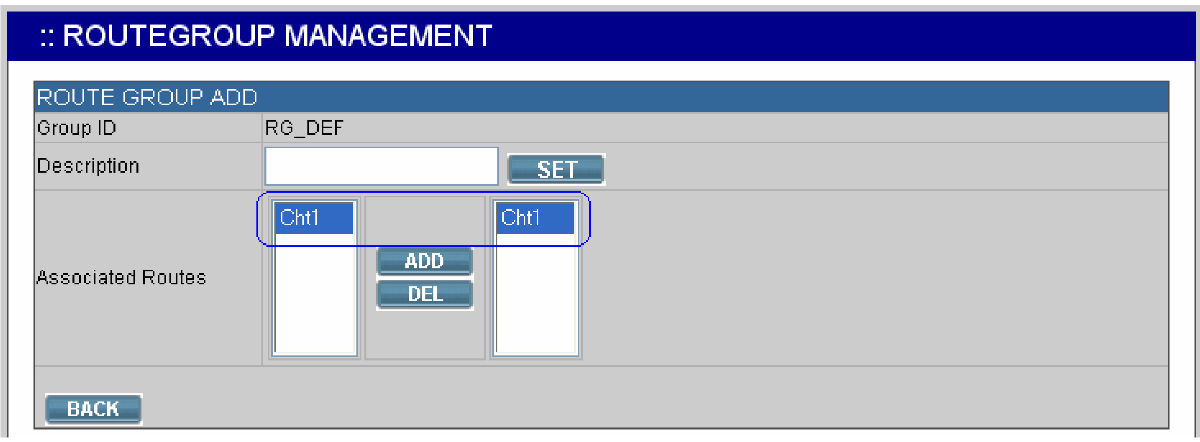
Destination number pattern: a destination number pattern consisting of digits, digit set, and wildcard character

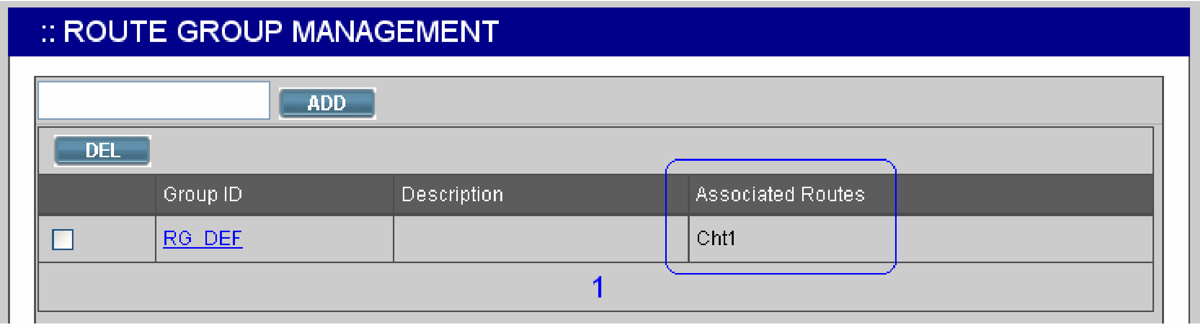
Number of stripped digits: number of leading digits to be stripped from the original dialed number when matches this route.



STEP 3:

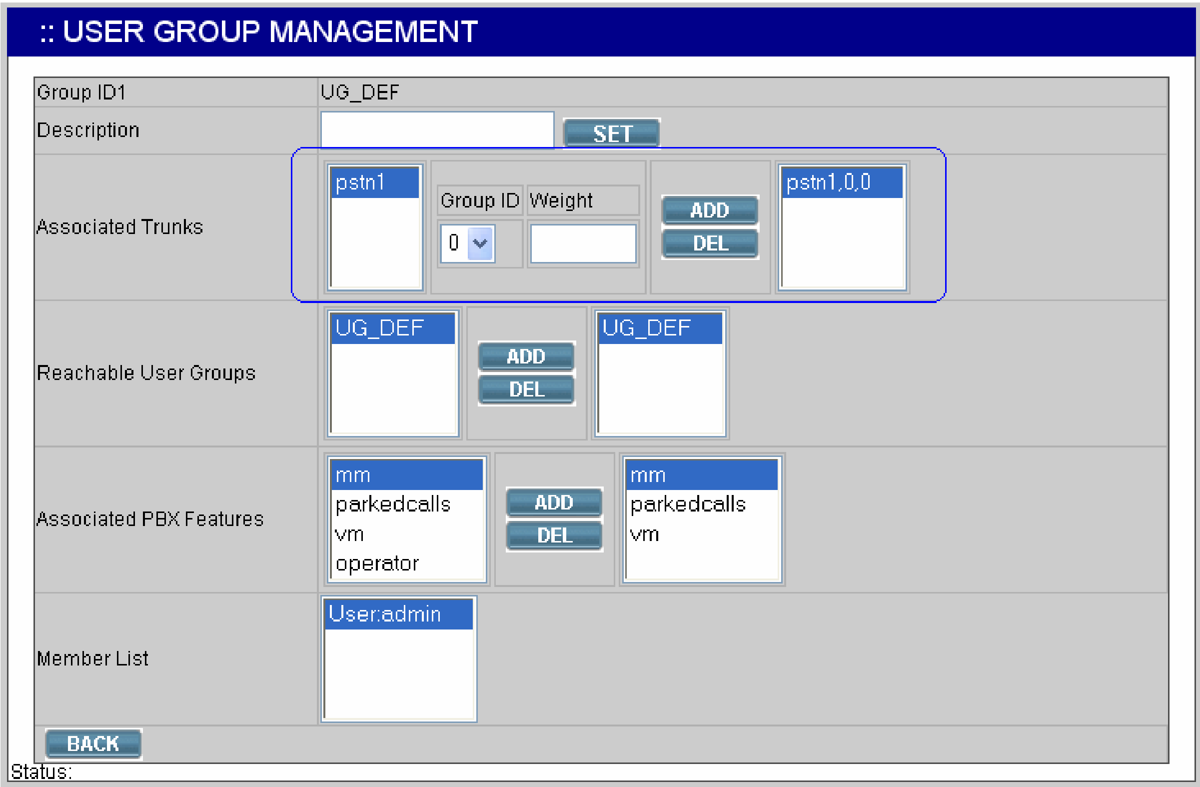
Please browse to "**Routegroup**" page in "**Route**" menu and select SIP route associated routes by this routegroup.

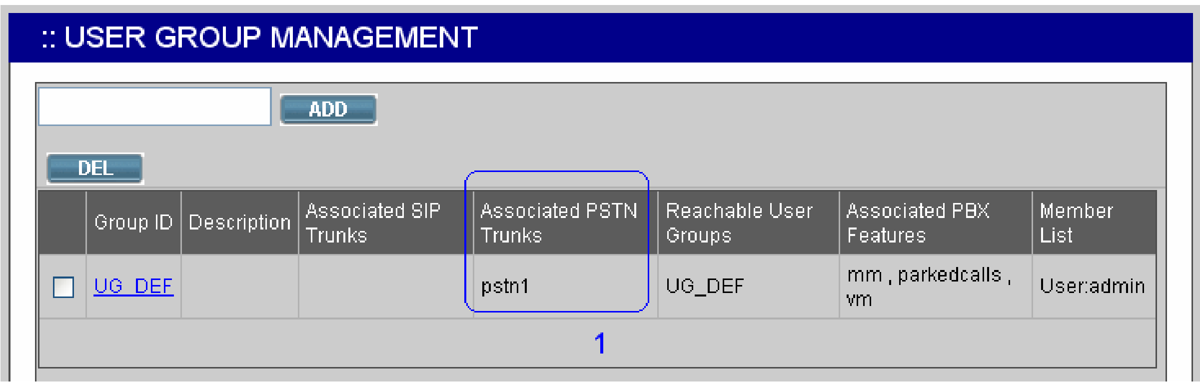




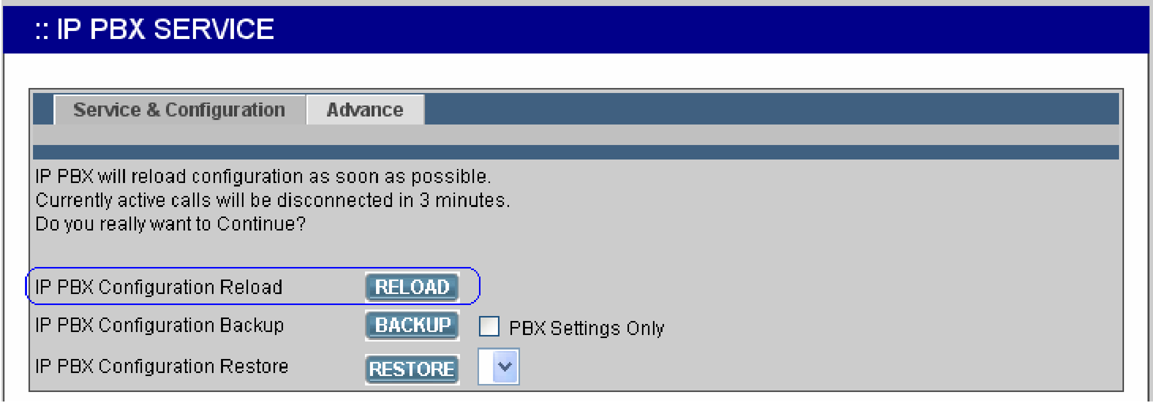
STEP 4:

Please browse to "**Usergroup**" page in "**User**" menu, and select outbound **"SIP accounts number**" trunks accessible by this usergroup.





After these configurations, be sure to press to "**Save**" button to apply settings and browse to "**IP PBX service**" page in "**Service**" menu to click the "**Reload**" button to make the settings effective.



IPX-1800N Usage:

IPX-1800N IP Ext 1001 calls to ISDN PSTN number

Human operation at IPX Caller side	Equipment operation	Human operation at VIP Receiver Side
Pick up phone 1001	1. IPX-1800N dial tone is heard.	
Dial 9 + phone number	1.Du Du is heard 2.IPX-1800N communication is going	
Ring back tone is heard		Phone number is ringing