SMC Networks

USER GUIDE

SMCPBX10

TigerVoIP™
IP PBX Telephony System

IP PBX Telephony System User Guide

The easy way to make all your network connections

SMC®
Networks

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Irvine, CA 92618
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E112007-EK-R01

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* SMC will provide warranty service for one year following discontinuance from the active SMC price list. Under the limited lifetime warranty, internal and external power supplies, fans, and cables are covered by a standard one-year warranty from date of purchase.

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COMPLIANCES

Federal Communication Commission Interference Statement

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with instructions, may cause harmful interference to radio communications. However, there is no guarantee that the interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

• Reorient the receiving antenna
• Increase the separation between the equipment and receiver
• Connect the equipment into an outlet on a circuit different from that to which the receiver is connected
• Consult the dealer or an experienced radio/TV technician for help

FCC Caution: Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

Note: In order to maintain compliance with the limits of a Class B digital device, you are required to use a quality interface cable when connecting to this device. You may use unshielded twisted-pair (UTP) for RJ-45 connections - Category 3 or better for 10 Mbps connections, Category 5 or better for 100 Mbps connections.

FCC - Part 68

This equipment complies with Part 68 of FCC Rules. On the base unit of this equipment is a label that contains, among other information, the FCC Registration Number and Ringer Equivalence Number (REN) for this equipment. If requested, this information must be given to the telephone company.

This equipment uses the following USOC jacks: RJ-11.

The REN is useful to determine the quantity of devices you may connect to your telephone line and still have those entire devices ring when your telephone number is called. In most, but not all areas, the sum of the REN of all devices connected to one line should not exceed five (5.0). To be certain of the number of devices you may connect to you line, as determined by the REN, you should contact your local telephone company to determine the maximum REN for your calling area.

If your equipment causes harm to the telephone network, the telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But if advance notice is not practical, you will be notified as soon as possible. You will be informed of your right to file a complaint with the FCC. Your telephone company may make changes in its facilities, equipment, operations or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted telephone service.

If you experience trouble with this telephone equipment, please contact please contact our company at the numbers shown on back of this manual for information on obtaining service or repairs. The telephone company may ask that you disconnect this equipment from the network until the problem has been corrected or until you are sure that the equipment is not malfunctioning.

This equipment may not be used on coin service provided by the telephone company. Connection to party lines is subject to state tariffs.

CE Mark Declaration of Conformance for EMI and Safety (EEC)

SMC contact for these products in Europe is:

SMC Networks Europe,

Edificio Conata II,

Calle Fructuós Gelabert 6-8, 2º, 4ª,

08970 - Sant Joan Despí, Barcelona, Spain.

This information technology equipment complies with the requirements of the Council Directive 89/336/EEC on the Approximation of the laws of the Member States relating to Electromagnetic Compatibility and 73/23/EEC for electrical equipment used within certain voltage limits and the Amendment Directive 93/68/EEC. For the evaluation of the compliance with these Directives, the following standards were applied.

RFI Emission	• EN 55022:1998 +A1:2000 +A2:2003, Class B
	• EN 61000-3-2:2000, Class A
	• EN 61000-3-3:1995 +A1:2001
Immunity:	• EN 55024:1998 +A1:2001 +A2:2003
	• IEC 61000-4-2: 2001
	• IEC 61000-4-3:2002 +A1:2002
	• IEC 61000-4-4:2004
	• IEC 61000-4-5:2001
	• IEC 61000-4-6: 2003 +A1:2004
	• IEC 61000-4-8:2001
	• IEC 61000-4-11: 2004
LVD:	• EN 60950-1:2001

Warning: Do not plug a phone jack connector in the RJ-45 port. This may damage this device. Les raccordeurs ne sont pas utilisé pour le systéme téléphonique!

Table of contents:

1 Introduction

1.1 Overview

The *SMCPBX-10 IPPBX Administration Guide* provides instructions for administering the IPPBX system. IPPBX is an embedded call-processing server communicating with client stations with Session Initiation Protocol (SIP). It migrates the telephony network and the data network of a small-to-medium business (SMB) company into a manageable converged network. IPPBX works with various IP phones (desktop, WiFi, Bluetooth, and DECT), VoIP gateways, and analog telephone adapters to route calls among client phones, analog phones, and PSTN network. Additional voice features such as conferencing, auto attendant, and voicemail are seamlessly enabled to all phones. IPPBX also provides Internet access to all LAN devices through Network Address Translation (NAT).

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

- Configurable multiple layers IVR with office-hour setting
- Fax relay and pass through (T.38 and T.30)
- Voicemail IVR system
- Meet-me conference
- SIP registrar
- SIP outbound proxy for signaling and media
- SIP gateway (FXO/FXS)
- SIPPBX for extension calls

IPPBX has a built-in suite of voice applications for supplementary services, and no special-purpose hardware is required. Therefore, the total cost of ownership of a converged network enabled by IPPBX is lower than building separated infrastructures for legacy telephony network and data network. Moreover, it comes with a web-browsable[1] interface to the data network configuration and voice service provisioning, which brings both local and remote manageability of networks together to facilitate administration. IPPBX web configuration also provides Wizard to help administrator easily configure information of user, usergroup, route, routegroup, device and extensions step by step.

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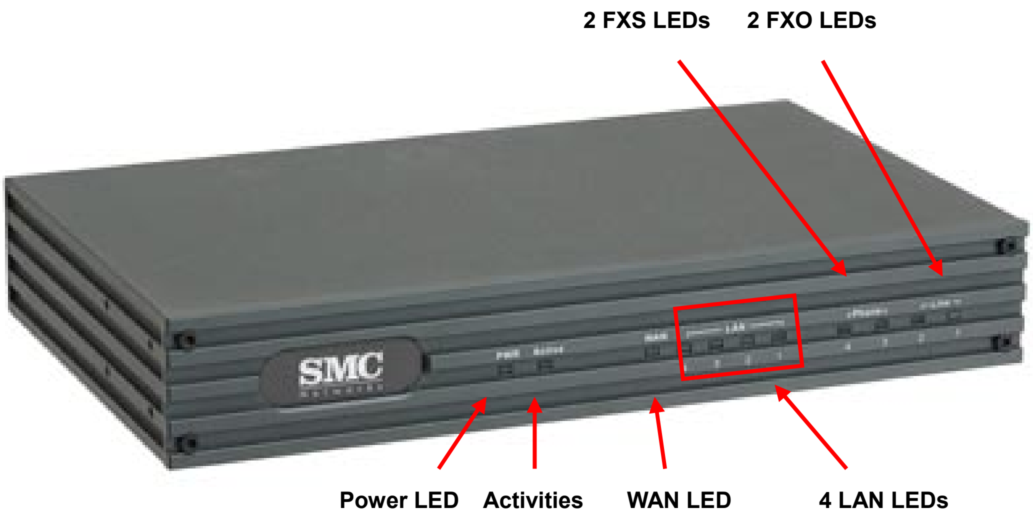
[1] It is highly recommended to use Internet Explorer 6.0 or later. The FireFox browser is not supported.

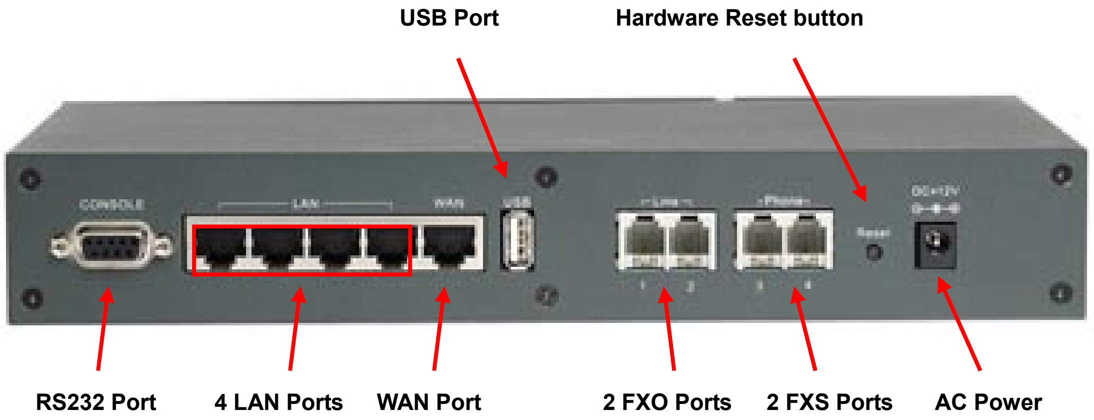
1.2 Installation

Front panel:



Power LED	Green, on: The power is working properly
Activities LED	Green, flashing: Transmitting or receiving data
WAN LED	Green, on: Connect to internet properly
LAN LEDs	Green, on: PC/NB to LAN is on line. (LAN1 to 4)
FXO LEDs	Green, on: Connect to PSTN line properly (FXO1~2)
FXS LEDs	Green, on: Connect to Phone/Fax line properly (FXS3~4)

Rear panel:



AC Power	Connect power supply with power jack (Power in: 12VDC/1.66A)
Reset button	1. Press then unclasp immediately → The system will reboot. 2. Press more than 7 seconds then unclasp → The system will go back to factory default
FXS ports	Connect a phone cable with RJ-11 connectors to an analog phone or FAX machine
FXO ports	Connect a phone cable with RJ-11 connectors to a PSTN line from CO.
USB port	External port with compliance to USB 1.1/2.0. Plug in an USB hard drive for CDR/voicemail backup from the internal storage.
WAN port	Connect an Ethernet LAN cable with RJ-45 connector to a broadband modem or a WAN router.
LAN ports	Connect an Ethernet LAN cable with RJ-45 connectors to an Ethernet-equipped computer, hub, bridge, or switch.
RS232 port	For technical console only

2 Web GUI Management

The factory default of LAN IP address is ***192.168.2.1***. Connect to LAN port and the configuration Web interface is at https://192.168.2.1/. Once connected, the browser will ask for accepting a certificate. Click **Yes** to see the home page. Type in the default username and password (admin/smcadmin) to log in for administration. The administrator password can be changed in the **User Management → User**.

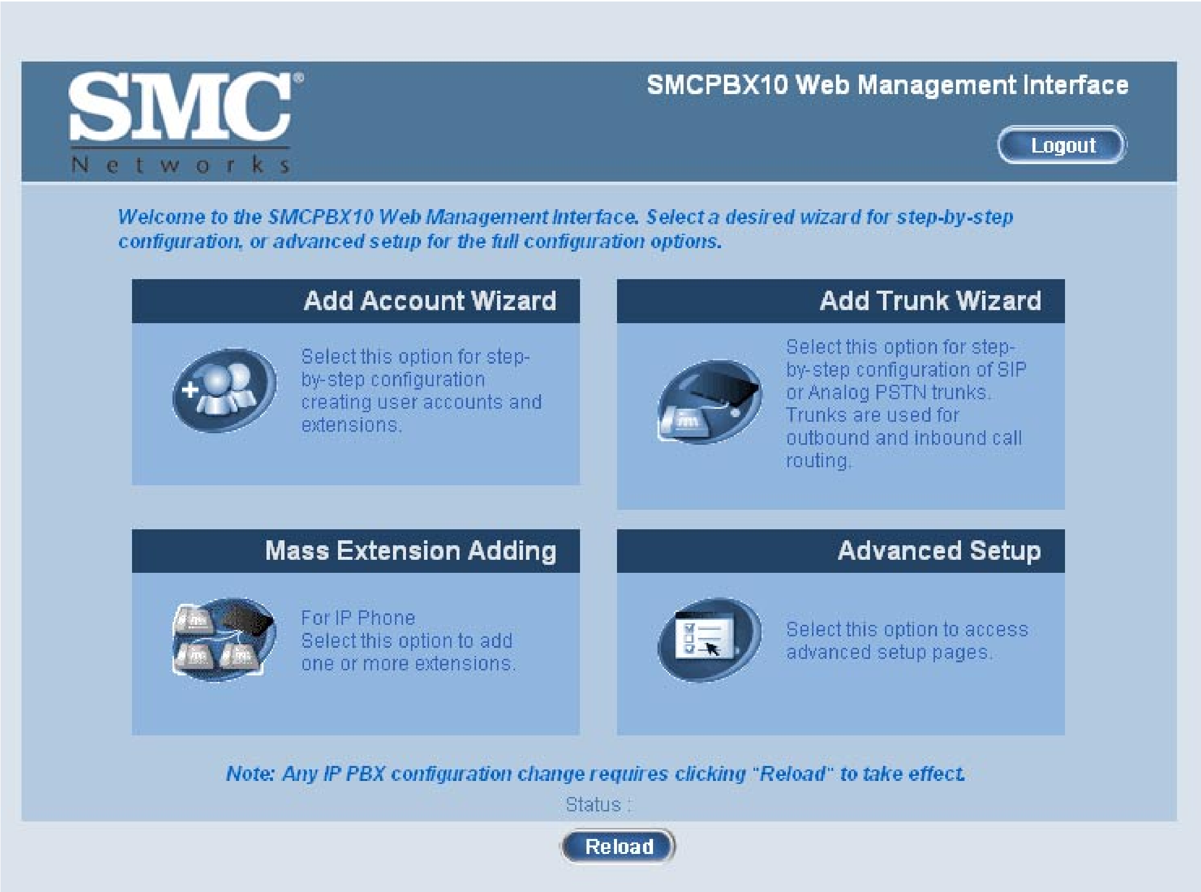
- Click User **admin** in the **Login ID** field.
- Change the password of User **admin** in **Password** field.
- Click **UPDATE** to change the password.

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

After login, you will see four icons, **Add Account Wizard**, **Add Trunk Wizard**, **Mass Extension Adding** and **Advanced Setup**. The first three icons can lead you step by step to configure some basic settings of IPPBX. Click the **Advanced Setup** icon to see all the PBX configurations in detail. Administrator can click on the top-right side of the webpage to go back to the home page of IPPBX Web Interface.

3 Wizard Configuration

With IPPBX Wizard configuration, the administrator can set basic configurations for IPPBX easily. With basic setup, IPPBX can function, and connect to the relevant devices and trunks. The Wizard Configuration includes Add Account Wizard, Add Trunk Wizard and Mass Extension Adding. When entering Wizard configuration, you will see Help at the bottom of each page that helps you to configure with Wizard. Any configuration change in Wizard requires clicking Reload at the bottom of the homepage.



3.1 Add Account Wizard

In **Add Account Wizard**, the administrator can setup usergroups, users and devices. You can follow the following steps to finish configuration. After finishing configuration, click Reload at the bottom of the homepage for configuration changes to take effect.

3.1.1 Add User Group

1. Enter a group ID and then click **ADD**.
2. The name will show in the table of the webpage.
3. Click the name to view the edit page.

4. Enter settings shown in **Table 3.1.1**.

5. Click **BACK** to return to the ADD USER GROUP page.

For deleting a usergroup, select a group ID and click **DEL**.

Note: Make sure there is no user associate with the usergroup, or it cannot be deleted.

Click **Next** to add user.

Table 3.1.1 Add Usergroup Settings

Field	Description
Description	Arbitrary description information. Click **SET** to add/update the information.
Associated Trunks[2]	Select routegroups and outbound trunks accessible by this usergroup. **Note:** the list order will determine hunting sequence in run-time. **Routegroup**: display available routegroups. **Trunk**: Display available trunks. **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. Click + or - to add or delete the associate trunks. After add all trunks, click **APPLY**. ☞ If there is not any appropriate SIP trunk and PSTN trunks to select, you may assign trunks at **Error! Reference source not found.** in Add Trunk wizard configuration after trunks are created in the previous step.
Reachable User Groups	Select a usergroup and click **ADD** that is 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,

[2] Please refer to **6.6**, **6.7** and **8.8** for details.
[3] Please refer to **7** for details.

	mm for Meet-me Conference, parked calls for Call Parking and vm stands for Voice Mail. ☞ 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.
Auth. Dial Passcode	Select and enter a password in number for caller to have the same privilege as this usergroup to dial out.

3.1.2 Add User

1. Enter settings shown in **Table 3.1.2**.
2. Click **ADD** to see the user information in the table of the webpage.

For deleting a user, select a Login ID and click **DEL**.
Click **Next** to choose a device.

Table 3.1.2 Add User 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 IPPBX 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 IPPBX 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.

3.1.3 Choose Device

Based on the devices you have, click **ADD IP PHONE** or **ADD ANALOG PHONE**, and **Next** to add/set the device.

Note: If selecting **ADD ANALOG PHONE**, the wizard will skip to Step 5.

3.1.4 Add Device

1. Enter a device name in the **Device ID** box.

2. Select **Auto Provision** if you want to enable Automatic Client Configuration.

3. Click **ADD** to see the newly added device in the table of the webpage, or to see the Enable Automatic Client Configuration (ACC) page if **Auto Provision** is selected.

4. Enter settings shown in **Table 3.1.3**, and click **ENABLE**.

Note: Auto Provision feature is only supported with SMC's IP Phone's (SMCDSP-200 SMCDSP-205 and SMCWSP-100)..

Note: For deleting a device, select a device ID and click **DEL**. Make sure there is no extension associate with the device, or it cannot be deleted.

5. Click **Next** to set a device.

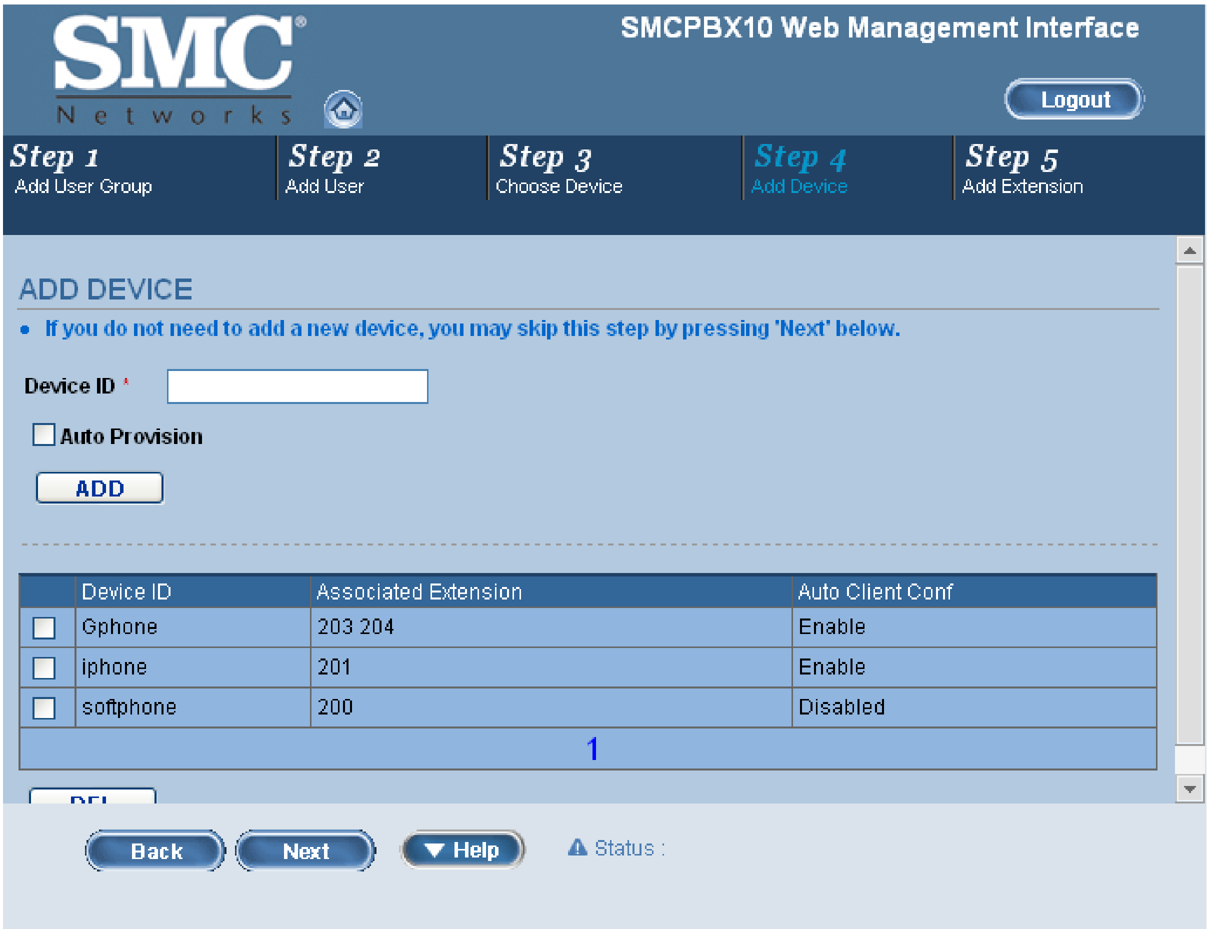


Table 3.1.3 ACC (Automatic Client Configuration) Settings

Field	Description
Vendor Prefix	The vendor Prefix's for SMC's IP Phones are as follows: SMCDSP-200 = dsp200 SMCDSP-205 = dsp205 SMCWSP-100 = wsp100
MAC Address	MAC address of the device.
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.

3.1.5 Add Extension

3.1.5.1 Add Extension (for IP Phone)

1. Enter settings shown in **Table 3.1.4**.

2. Click **ADD** to see the newly added extension in the table of the webpage.

For deleting an extension, select an extension number and click **DEL**.

Click **Finish** to finalize all the settings, and go back to the homepage.

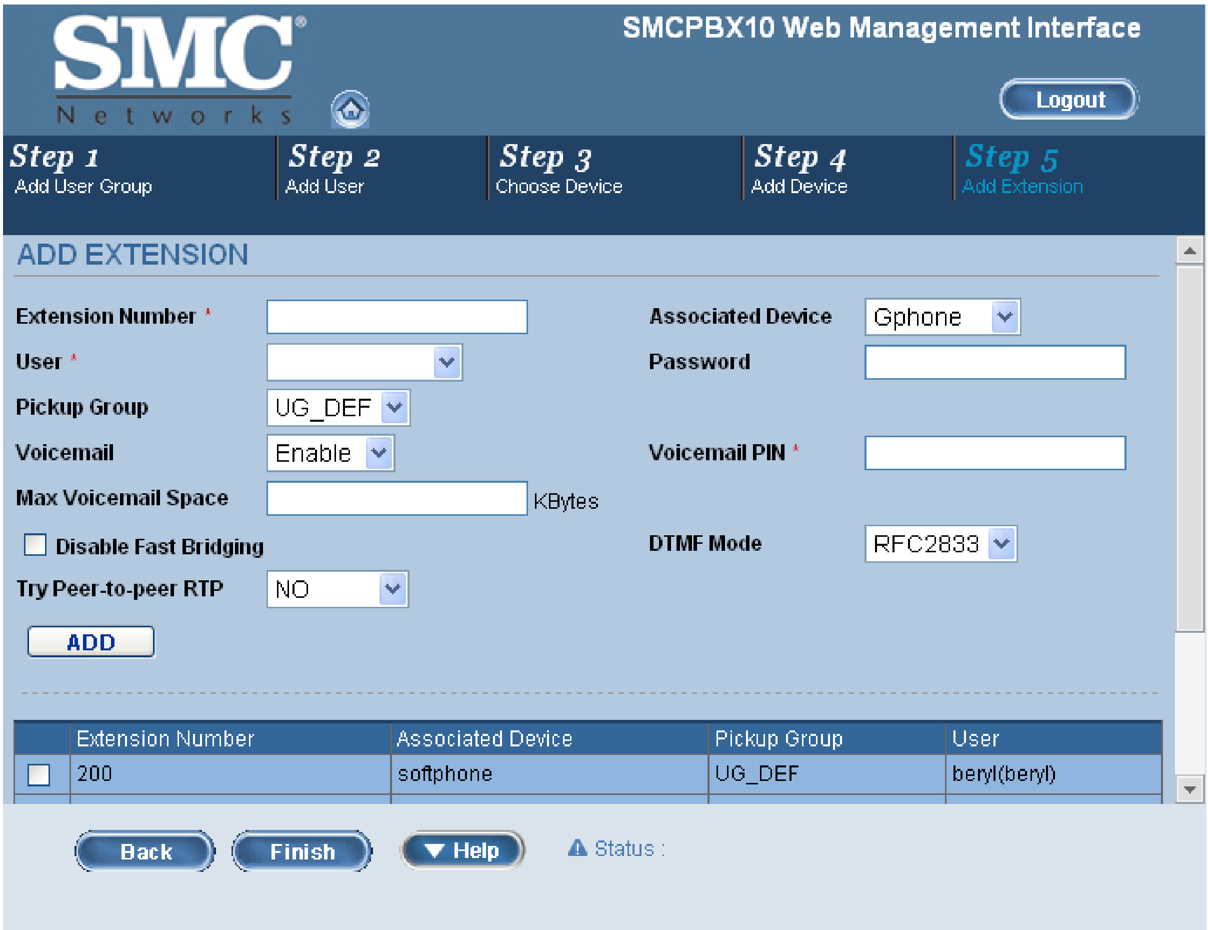


Table 3.1.4 Add Extension of IP Phone Settings

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.
User[4]	Select the user this extension associates with.

[4] Please refer to **6.1** for details.

	☞ If there is not any appropriate users to select, one can come back later once the expected user has been added.
Password	Password of this extension. Same password must be configured on the device side as well.
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.
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.
Max Voicemail Space	Enter maximum space in KBytes for voicemail.
Disable Fast Bridging	Select to disable media relay.
Try Peer-to-peer RTP	If click **YES**, IPPBX 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 IPPBX. However, only SIP INFO DTMF mode phones should enable this since other DTMF modes require IPPBX 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. ☞ In-band DTMF mode consumes the limited DSP resource when using a highly compressed codec, such as G.729 or G.723.1. Therefore, calls will not connect with such setting if DSP is not installed. Although using a low-complexity codec such as G.711 does not require DSP, DTMF detection still takes considerable CPU resource and impacts several system specs. Be cautious when configuring an extension with in-band DTMF mode.

3.1.5.2 Add Extension (for Analog Phone)

1. Enter settings shown in **Table 3.1.5**.
2. Click **ADD** to see the newly added analog phone in the table of the webpage.
3. Click **Finish** to finalize all the settings, and go back to the homepage.

For deleting an analog phone, select a POTS port and click **DEL**.

Table 3.1.5 Add Analog Phone Settings

Field	Description
POTS Port	FXS port index. The value should be either 3 or 4 dependant on which phone (FXS) port the analog phone is connected.
Pickup Group	The pickup group that the extension belongs to.
Extension Number	A unique line number composed of digits only, e.g. 101; 32 digits maximum.
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.
Max Voicemail Space	Enter maximum space in KBytes for voicemail.

3.2 Add Trunk Wizard

In Add Trunk Wizard, the administrator can setup routes, routgroups and trunks. Moreover, worktime and IVR are included in this part for assigning to trunks. You can follow the following steps to finish configuration. After finishing configuration, click 

at the bottom of the homepage to take the configuration effect.

[5] Please refer to **6.1** for details.

3.2.1 Add Route

1. Enter settings shown in **Table 3.2.1**.
2. Click **ADD** to see the newly added route in the table in the webpage.
3. Click **Next** to set a routegroup.

For deleting a route, select a route ID and click **DEL**.

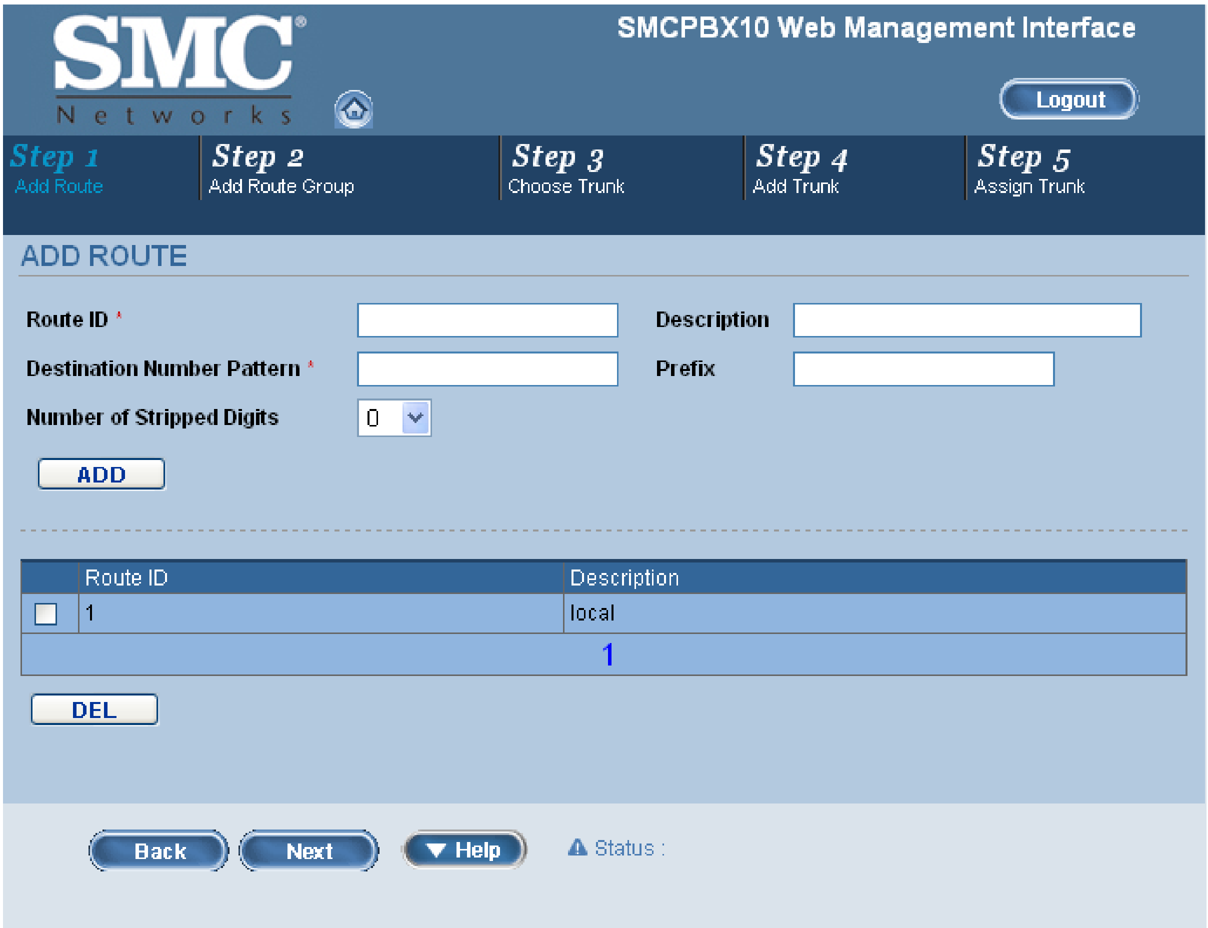


Table 3.2.1 Add Route Settings

Field	**Description**
Route ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 16 characters maximum.
Description	Arbitrary description information.
Destination Number Pattern[6]	A destination number pattern consisting of digits, digit set, and wildcard characters, e.g. 9NXXXXXX matches

[6] For more information about the available digit set and wildcard characters, please refer to **Table 6.4**.

	any 7-digit called number starting from a digit larger or equal to 2 and with an extra prefix digit 9.
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.
Number of Stripped Digits	Select 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.

3.2.2 Add Route Group

1. Enter a group ID and then click **ADD**.
2. Enter settings shown in **Table 3.2.2**.
3. Click **BACK** to return to the ADD ROUTE GROUP page.
4. Click **Next** to choose trunk.

For deleting a routegroup, select a group ID and click **DEL**.

Note: Make sure there is no route associate with the routegroup, or it cannot be deleted.

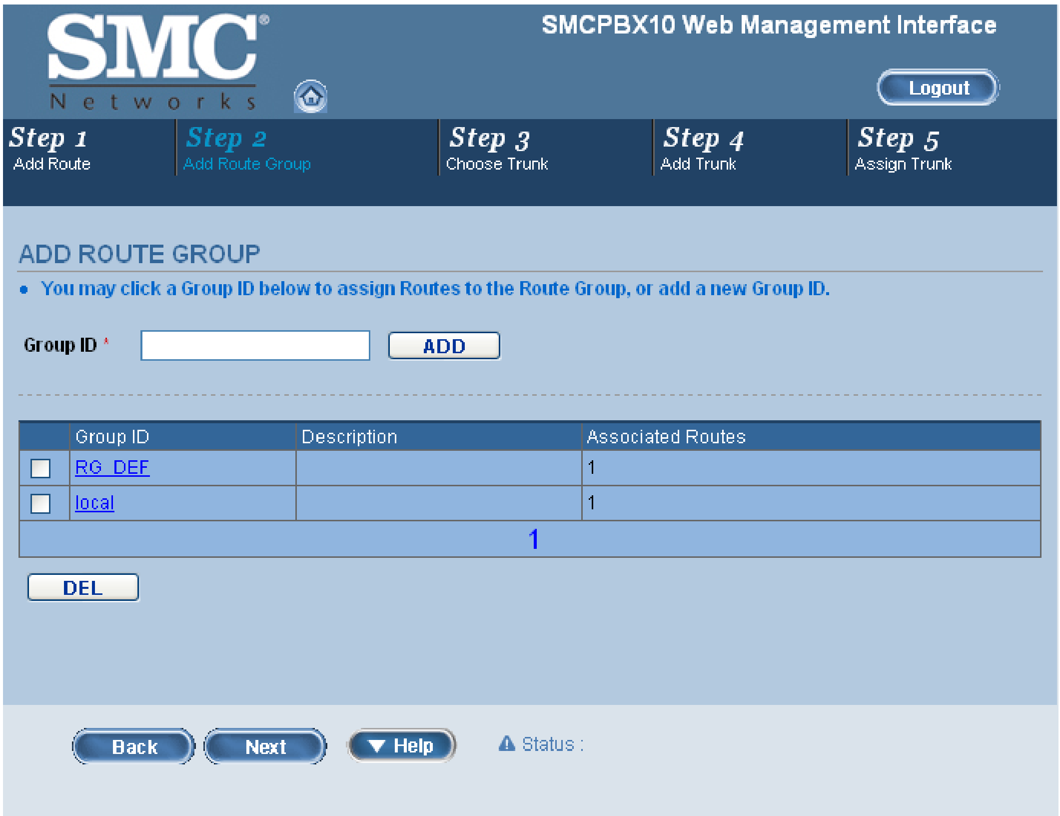


Table 3.2.2 Add Routegroup Settings

Field	Description
Description	Arbitrary description information. Click **SET** to add/update the information.
Associated Routes[7]	Select routes belonged to this routegroup. Click + or - button to add or remove a route to or from the routegroup. The right box lists current selected routes. Click **SET** to update the information. Note the order of the selected routes is important since it decides which route will 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.

3.2.3 Choose Trunk

In the Choose Trunk page, click **SIP TRUNK** or **ANALOG PSTN TRUNK** to select which kind of the trunks you want to add.

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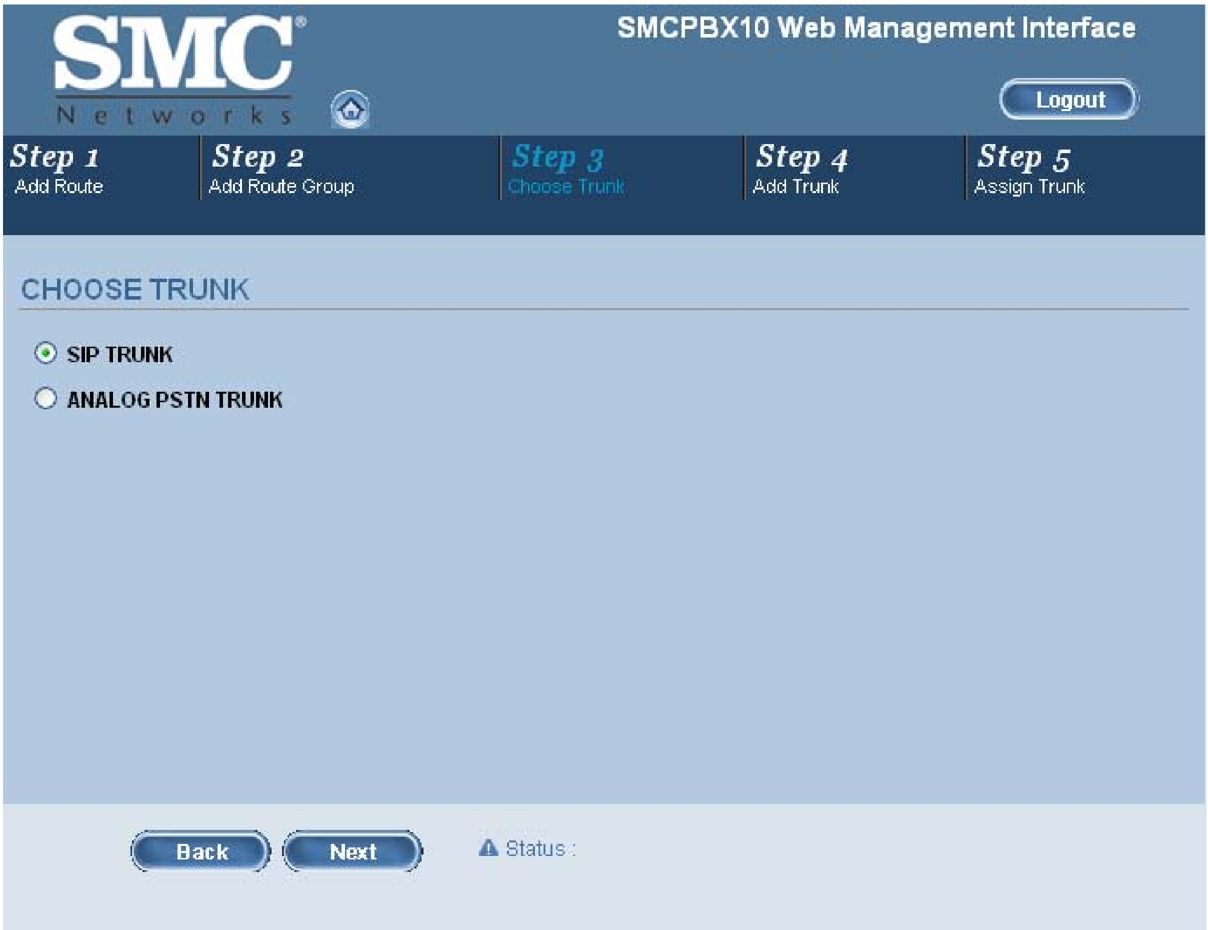
[7] Please refer to **6.4** for details.



3.2.4 Add Trunk

3.2.4.1 SIP Trunk

1. Enter settings shown in **Table 3.2.**.
2. Click **ADD** to see the newly added SIP trunk in the table in the webpage.
3. Click **Next** to assign trunks to usergroups.

For deleting a SIP trunk, select a trunk identifier and click **DEL**.

Table 3.2.4.1 Add SIP Trunk 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.
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.
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 IPPBX at a remote site with dynamic IP address. Once the remote IPPBX registers, calls from local to remote can be made reversely over the trunk.
SIP Proxy IP 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.
Registration Required	Select if registration to a registrar is required to activate the trunk. This is true for a remote IPPBX or an ITSP account, however, may be not required in case of a SIP gateway.
SIP Registrar IP 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).
IVR List[8]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID. Leave it blank and the system will automatically create an IVR for the trunk.
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.

3.2.4.2 Analog PSTN Trunk

1. Enter settings shown in **Table 3.2.3**.
2. Click **ADD** to see the newly added analog PSTN trunk in the table in the webpage.

[8] Please refer to **7.11** for details.
[9] Please refer to **6.2** for details.

3. Click **Next** to assign trunks to usergroups.

For deleting an analog PSTN trunk, select a trunk identifier and click **DEL**.

Table 3.2.3.2 Add Analog PSTN Trunk Settings

Field	**Description**
Trunk Group	ID number of this PSTN trunk group. A valid number ranges from 1 to 32. It should not overlap with existing ISDN PSTN trunk groups.
Trunk Type	Select the port type, FXO or FXS. If selecting FXS, users can see **By Number** and **By Privilege** in the **DID of Extension** list, and be able to configure **DID Prefix** and **DID Stripping**.
Trunk Ports	FXO and FXS port indices grouped by this PSTN trunk, such as 1 or 1, 2 or 1-3, etc. Maximum port index is 4.
Description	Arbitrary description information.
Port Selection	Click to search for an available port in the group. **Rotating** means to force ports being selected by turns to even cost.
DID of Extension	When enabled DID, clicks an extension in the list to be an unconditional destination for incoming calls to this trunk. The PSTN numbers of the included ports are therefore regarded as the direct line numbers of the extension. If **FXS** is selected in the **Trunk Type** list, you can also click **By Number** or **By Privilege**, and then enter configurations in **DID Prefix** and **DID Stripping** to have the incoming calls directed to the corresponding extension or trunk derived by number manipulation. The PSTN trunk numbers is 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. ☞ If selecting **By Number**, the "number" being manipulated for extension DID is the called (destination) number. As a result, one should confirm what prefix, usually the area code, would

	be given by the service provider side so that a correct stripping could be configured accordingly.
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 **By Number** or **By Privilege**, 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. Click **All** to strip all digits of the original called number.
IVR List[10]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID. Leave it blank and the system will automatically create an IVR for the trunk.
Usergroup[11] 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 any appropriate usergroups to select initially. One can come back later to revise it, once the expected usergroups are added.
Caller ID Detection	Select to detect the Caller ID calling from PSTN lines.
Answering by Battery Reversal Detection	If the PSTN service provides battery reversal, select to count billable time starting from the call is answered. ☞ Clear the check box, if you are not sure whether the PSTN service provides the function.

3.2.5 Assign Trunk

In ASSIGN TRUNK page, all usergroups display here. The administrator can assign trunks to any usergroup at this stage.

1. Click a group ID to see the Assign Trunk Management.
2. Enter settings shown in **Table 3.2.4**.
3. Click [+] or [-] to add or delete the associate trunks. After add all trunks, click **APPLY**.

---

[10] Please refer to **7.11** for details.
[11] Please refer to **6.2** for details.

4. Click **Finish** to finalize all the settings, and go back to the homepage.

Note: The order of the assigning trunks impacts the hunting sequence in run-time.

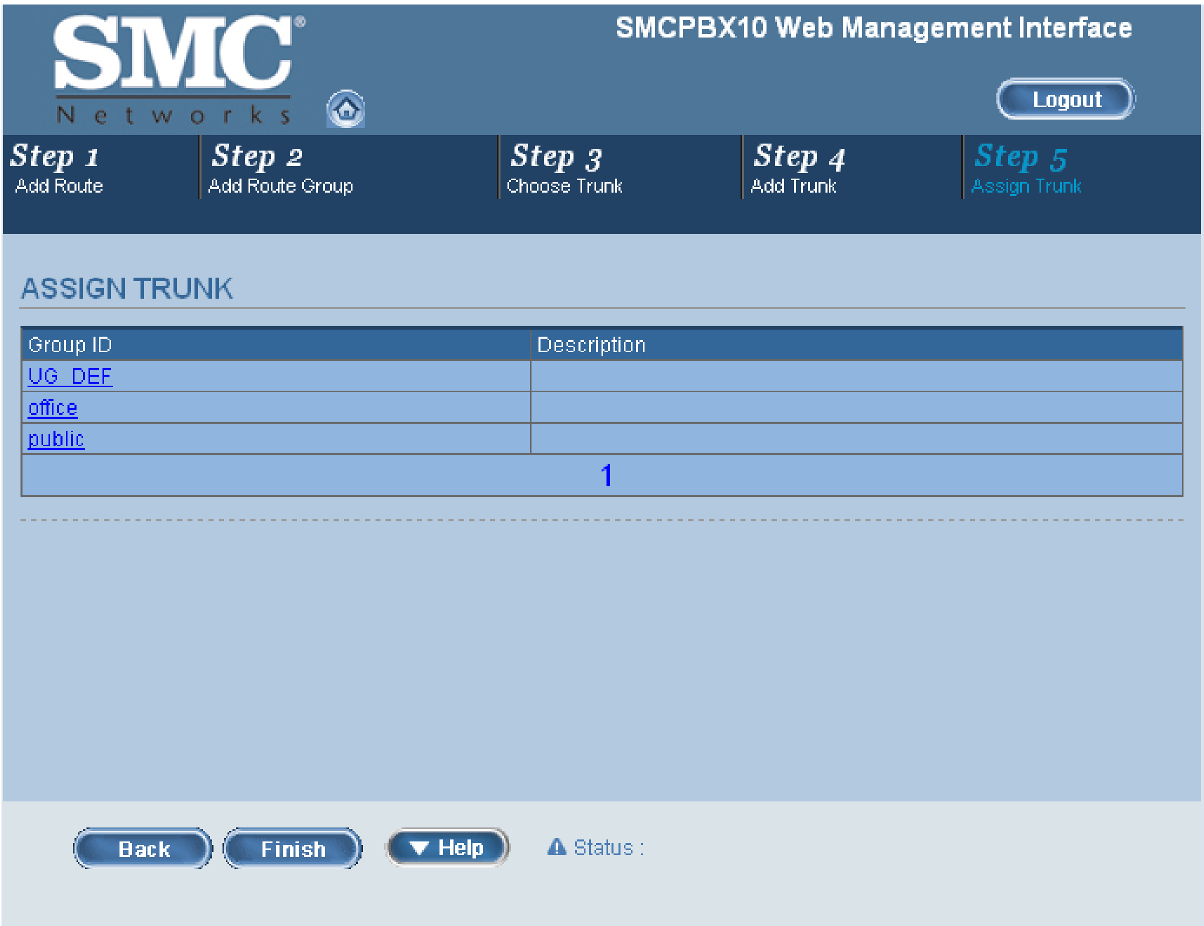


Table 3.2.4 Assign Trunk Settings

Field	Description
Route Group	Click to select available routegroups.
Trunk	Click to select available trunks.
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.

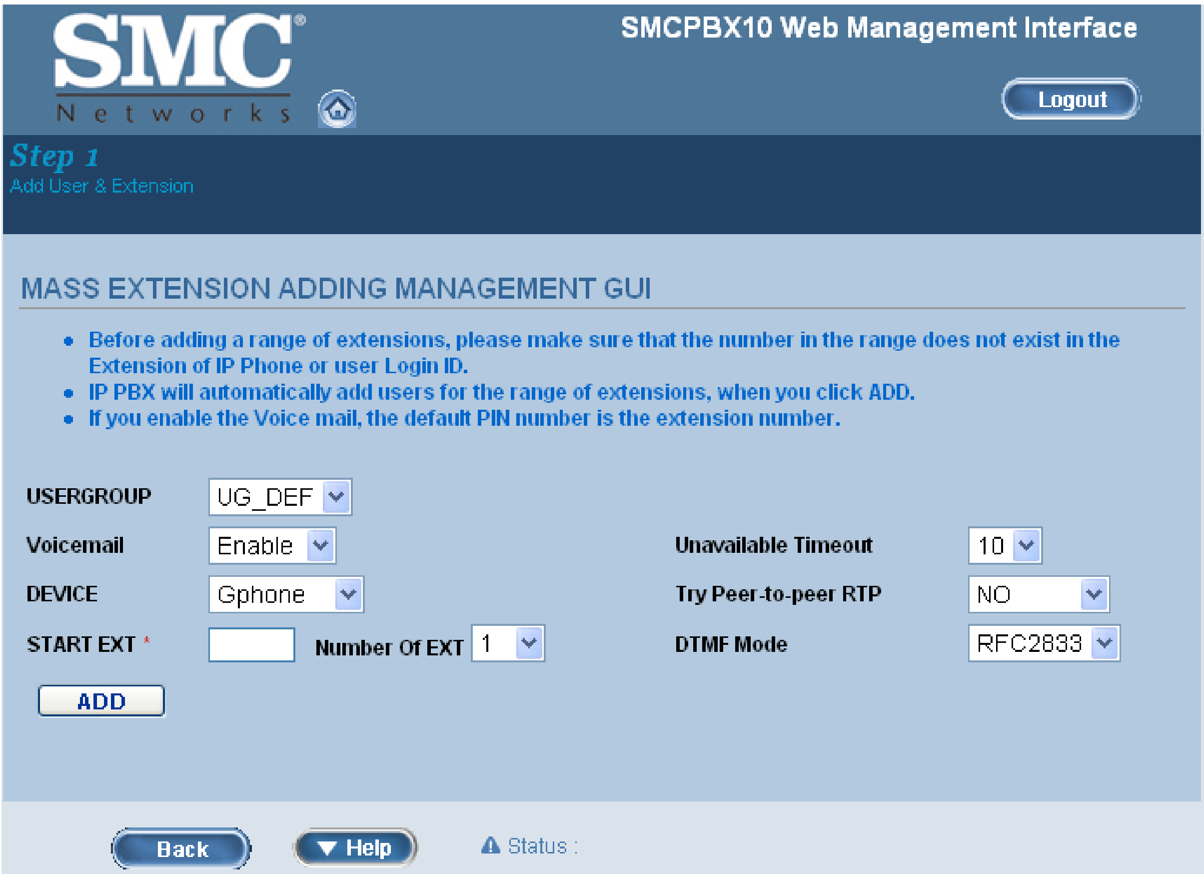
3.3 Mass Extension Adding

The Mass Extension Adding page helps the administrator to add many extensions and assign users for these extensions under a usergroup and a device. After finishing configuration, click Reload at the bottom of the homepage to take the configuration effect.

Note: Make sure the range of extension numbers does not exist in Extension of IP Phone or user Login ID, or the configuration will fail to continue.

3.3.1 Add User & Extension

1. Click a usergroup in the **USERGROUP** list.
2. Click an IP phone device in the **DEVICE** list.
3. Enter the digits for the starting extension.
4. Click a number in the **Number of EXT** list for the mass extension adding.
5. Click **ADD**, and IPPBX will start to add these extensions automatically.
6. Click **Back** to the homepage.

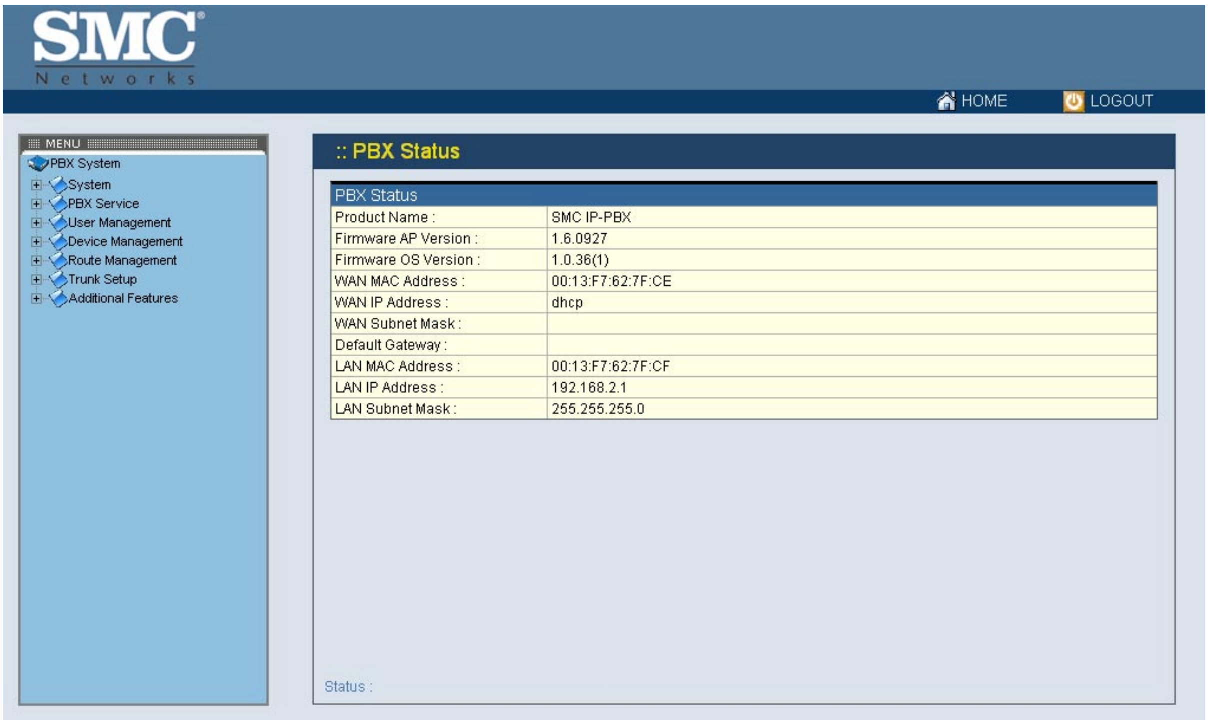


4 System Configuration

This section describes how to configure system parameters used by IPPBX. Click **Advanced Setup** after login the web interface to configure the following system parameters.

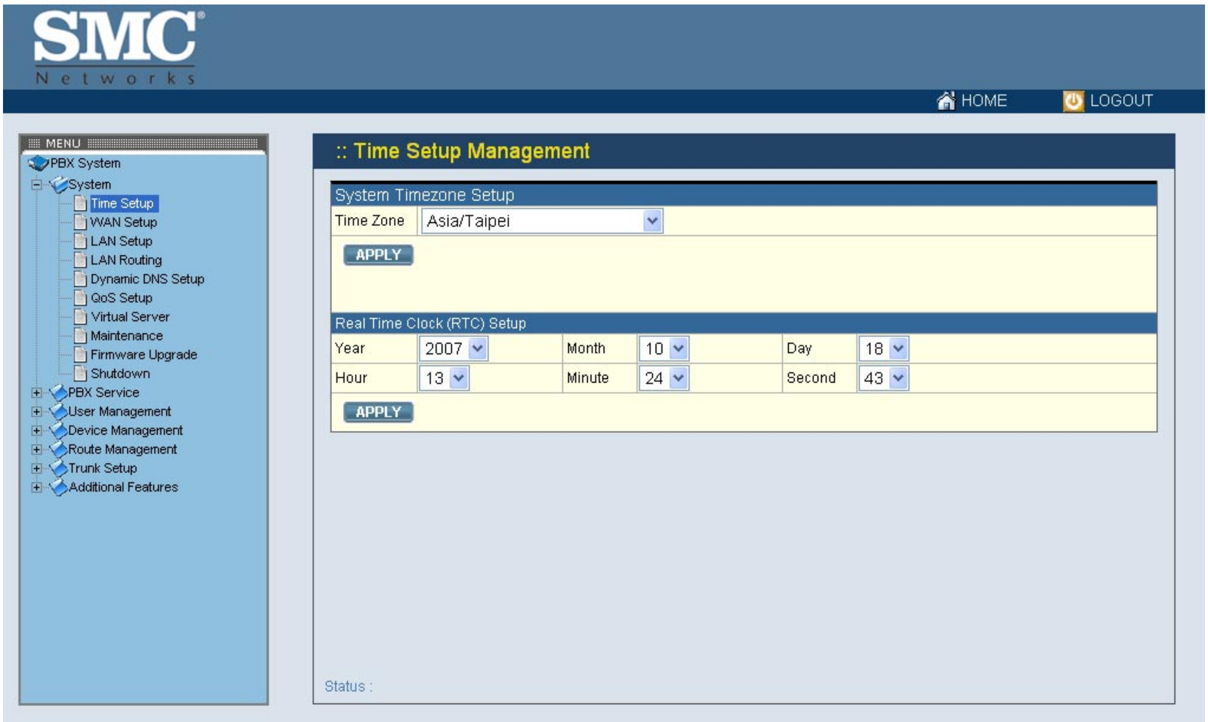
4.1 PBX System

The PBX System page briefs IPPBX 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 IPPBX.



4.2 Time Setup

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



4.2.1 Time Zone Setup

Click a region/country in the **Time Zone** list, and click **APPLY** in **System Timezone Setup**.

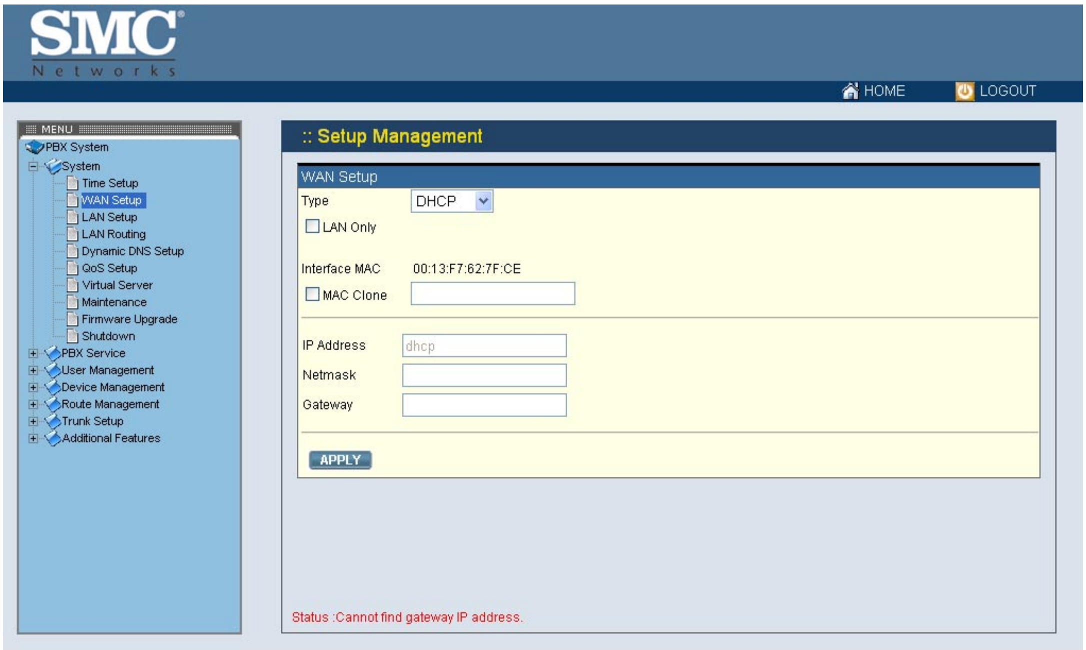
4.2.2 Real Time Clock (RTC) Setup

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.

4.3 WAN Setup

The On-board WAN Setup page allows administrator to configure WAN network interface for IPPBX. 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. Select **MAC Clone** to change the WAN MAC address.



4.3.1 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.

4.3.2 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.

4.3.3 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**.

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.

4.3.4 LAN only

Select **LAN Only** to disable WAN IP settings but allow the configuration of default gateway and primary/secondary/third DNS servers.

4.3.5 MAC Clone

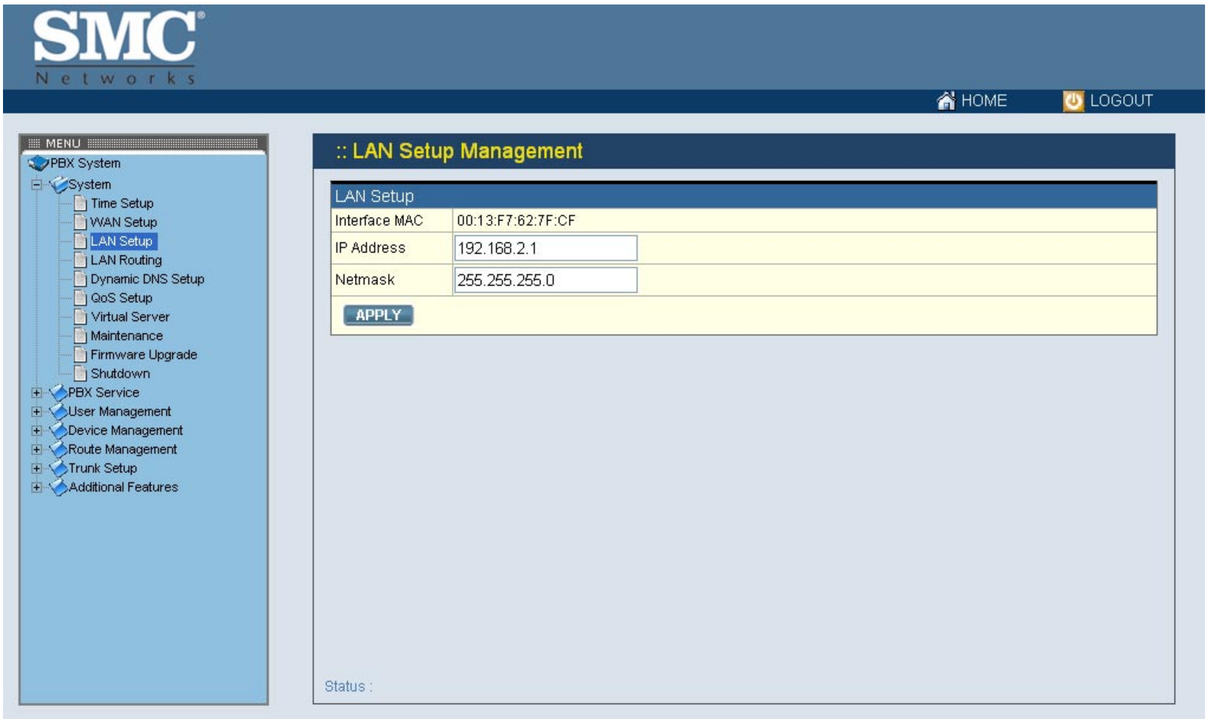
Select **MAC Clone** and enter a MAC/physical address to change the WAN MAC address.

4.4 LAN Setup

The On-board LAN Setup page allows administrator to configure LAN network interface for IPPBX.

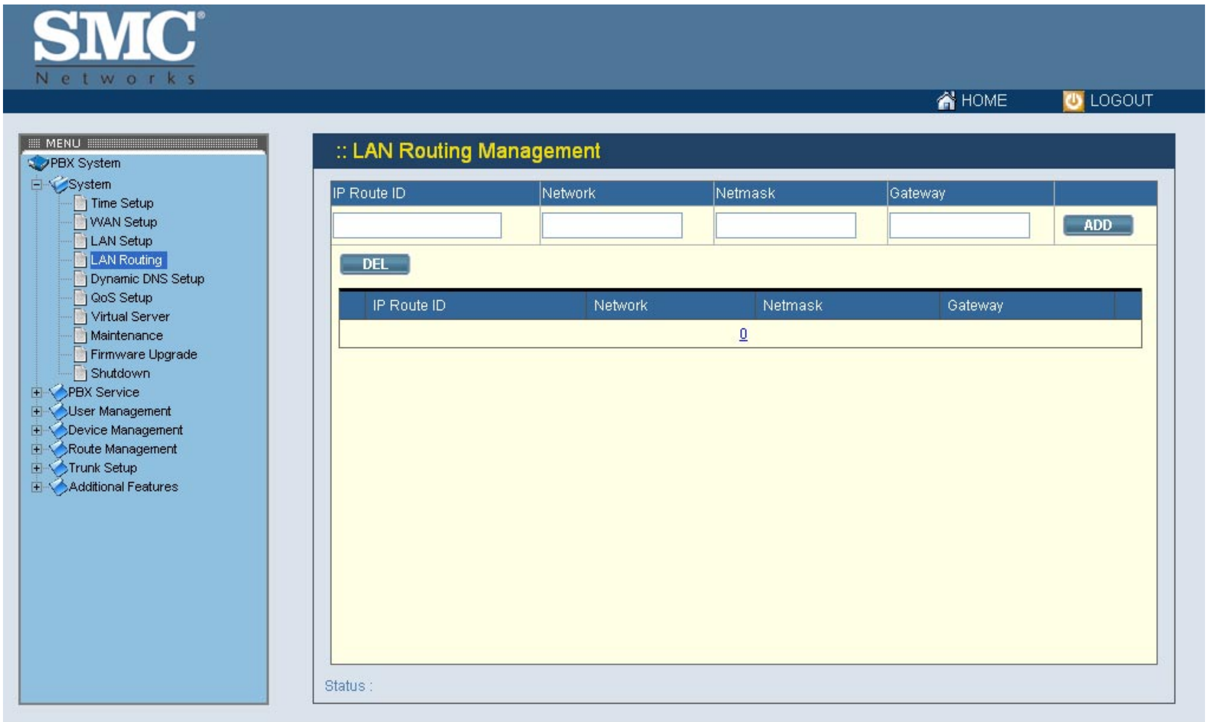
1. Select **System** --> **On-board LAN Setup** to see the current settings of LAN network interface.
2. Enter a new IP address and network mask.
3. Click **APPLY** to change the settings.

Note: By default IPPBX 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** --> **IPPBX Service**, and click **Restart** to activate the changes.



4.5 LAN Routing

To enable static routing among LAN subnets, enter network information and the IP address of the corresponding gateway in the IPPBX's LAN. It is important to assure that the given gateway IP address sits in the IPPBX'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 IPPBX Service needs to be restarted to regard calls from designated LAN subnets as LAN traffic. Go to **Service** --> **IPPBX Service**, and click **Restart** to regard calls as LAN traffic.



4.5.1 Add a Route

1. Enter the **IP Route ID**, **Network**, **Netmask**, and **Gateway**.
2. Click **ADD** to have the newly added route in **IP Rout ID**.

4.5.2 Edit a Route

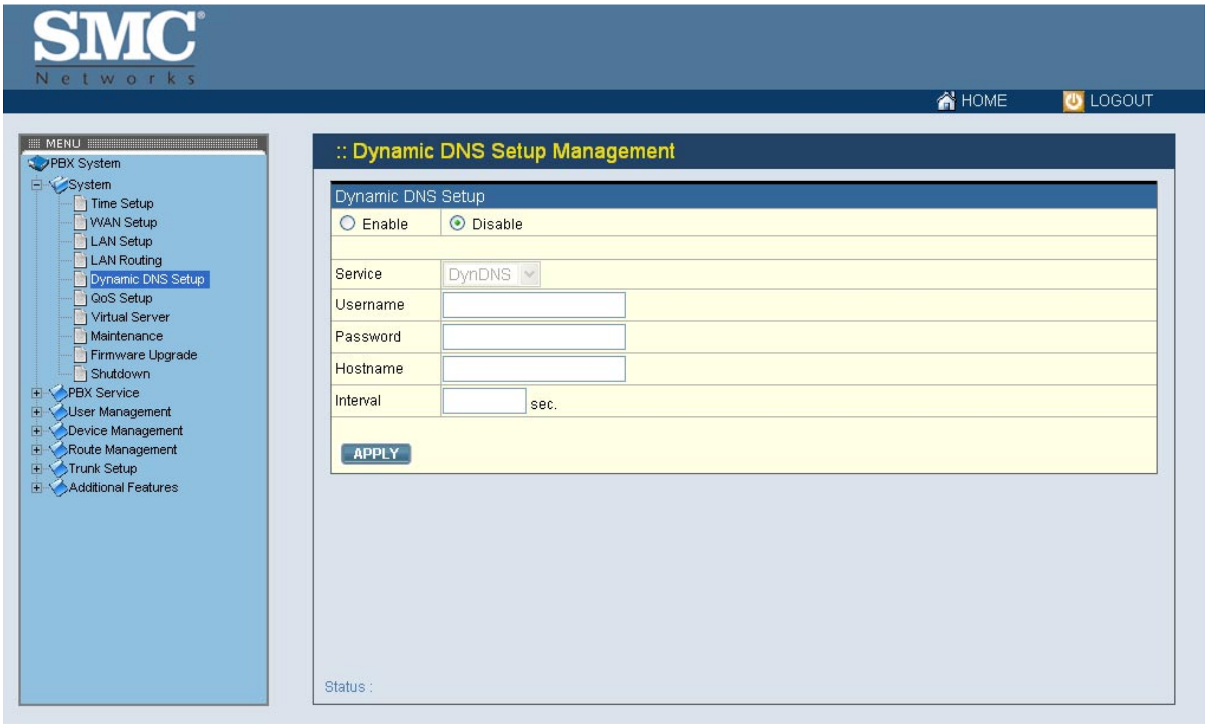
1. Edit the information in a row.
2. Click **APPLY** in the row to update the information.

4.5.3 Delete a Route

1. Select a route ID.
2. Click **DEL** to remove the route ID from the **IP Route ID** column.

4.6 Dynamic DNS Setup

Dynamic WAN IP address causes difficulty for inbound connections from remote clients or IPPBX 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 IPPBX). 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.



4.6.1 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.

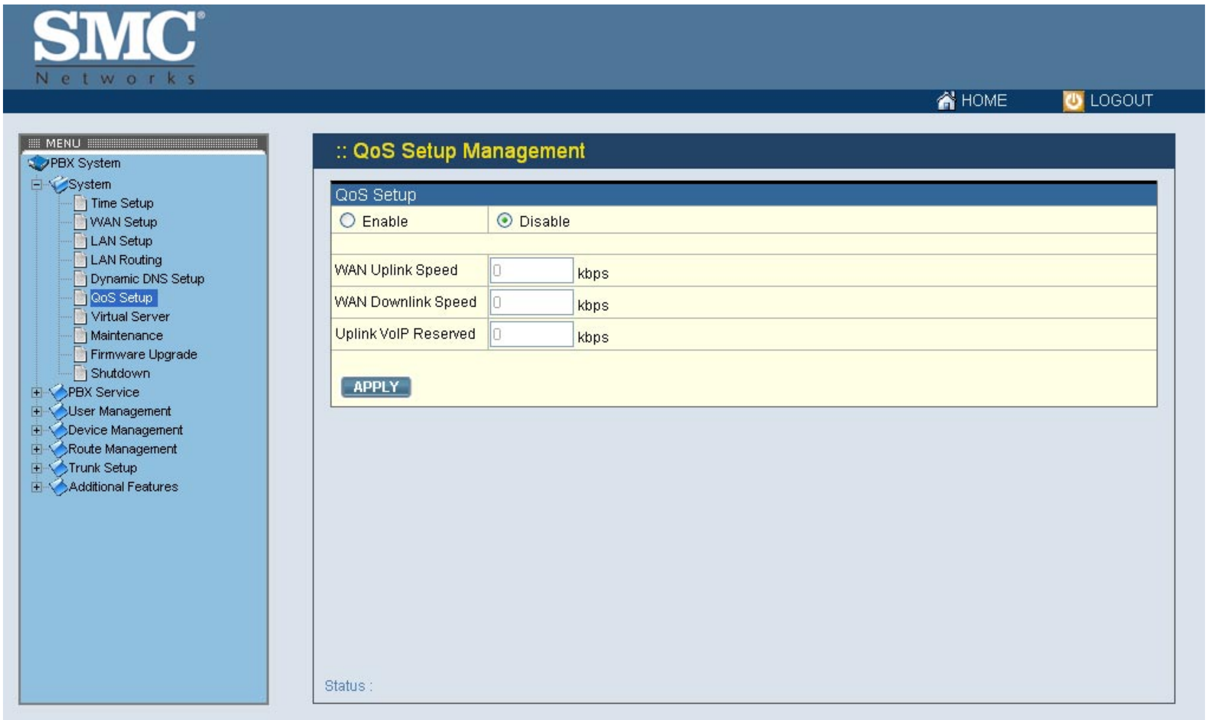
- Click **Enable**.
- Click **DynDNS** or **3322.net** in the **Service** list.
- Enter the **Username**, **Password**, **Hostname**, and **Interval**.
- Click **APPLY**.

4.6.2 Disable Dynamic DNS

Click **Disable**, and then click **APPLY**.

4.7 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.



4.7.1 Enable QoS

1. Click **Enable**.

2. Enter the **WAN Uplink Speed**, **Downlink Speed**, and **Uplink VoIP Reserved** (bandwidth).

3. Click **APPLY**.

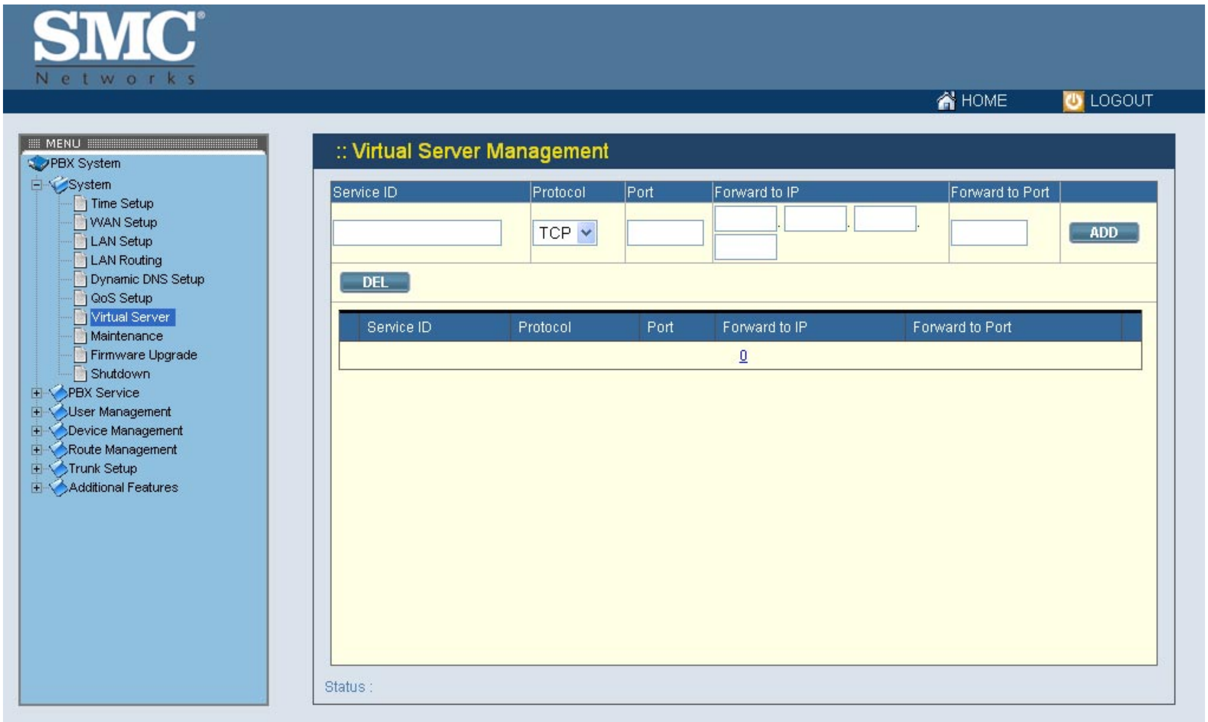
For a popular 2M/256K ADSL program, the WAN uplink speed would be 256 and the WAN downlink speed would be 2048. The Uplink VoIP reserved could be, say, 192 out of the total 256 kbps to allow 2 concurrent G.711 calls.

4.7.2 Disable QoS

Click **Disable**, and then click **APPLY**.

4.8 Virtual Server

You can configure IPPBX 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, IPPBX 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 IPPBX redirects the external service request to the appropriate internal server (located at one of your LAN's Private IP Address). To allow to 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.



4.8.1 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**.

4.8.2 Edit a Service

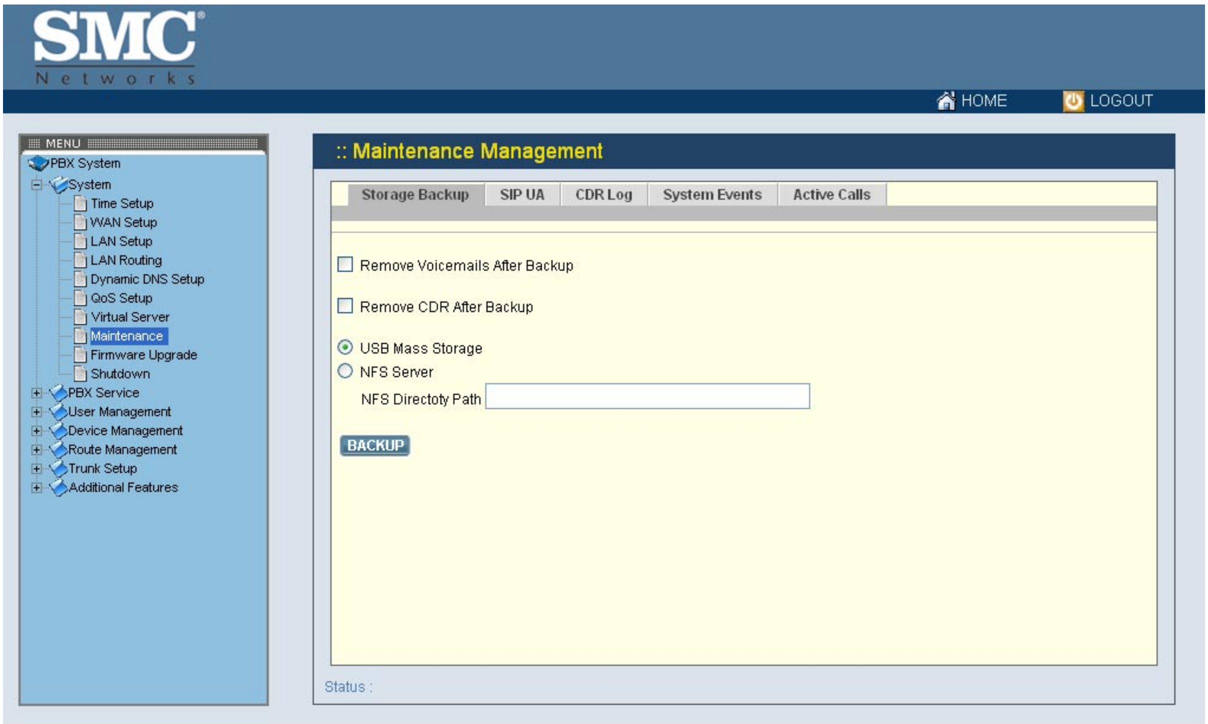
1. Change any information in a row.
2. Click **APPLY** in the row to update the information.

4.8.3 Delete a Service

1. Select a service ID.
2. Click **DEL** to remove the service from the **Service ID**.

4.9 Maintenance

This page includes maintenance functions of IPPBX, including Storage Backup, SIP UA, CDR Log, System Events, and Active Calls.



4.9.1 Storage Backup

To back up internal main storage, the administrator can back up the internal main storage to USB Mass storage or NFS server. You can select to keep or remove CDR and/or voicemails after backup.

4.9.1.1 Back up to USB Mass Storage

Click **USB Mass Storage**, and then **BACKUP** to follow the instructions to insert the USB connector of an external USB drive. 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.

4.9.1.2 Back up to NFS Server

Click **NFS Server**, enter a URL path in **NFS Directory Path**, and then click **BACKUP** to have the internal main storage stored in the NFS server.

4.9.2 SIP UA

SIP UA lists the registration status of each client and remote IPPBX, 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.

4.9.3 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, 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. Click **SHOW CDR** to see all the record. You can export the CDR as *.cvs file by clicking **GET FILE**. Click **DELETE CDR** to erase all the record in PBX.

4.9.4 System Events

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

4.9.5 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.

Table 4.9.5 Active Calls Record Description

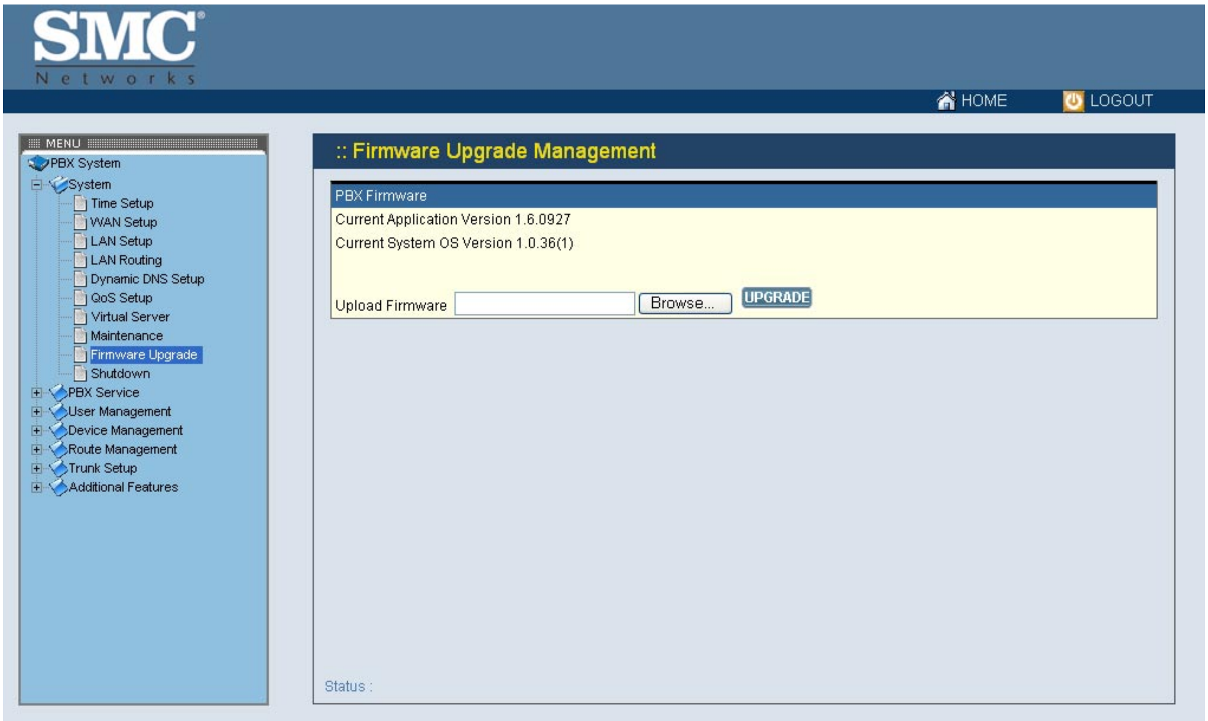
Field	Description	
Client	Show the caller or callee's extension number, port number, or SIP trunk ID.	
State	Connected	In the conversation.
	Ring	The client is a caller and is ringing a callee.
	Ringing	The client is a callee and is ringed by a caller.
	Reserved	FXS detects off-hook.

	Down	The client is dormant.
Service	Dial	The client is a caller.
	Answer	The client is a callee.
	Pause	The client is being paused for some time.
	Prompt	The client is being prompted for a pre-recorded audio.
	IVR	Calls from a trunk are picked up by Auto-Attendant.
	Meet-me	The client enters meet-me.
	Voicemail	The client enters voicemail.
	Busy-Callback	System is conducting a requested Line-in-Use callback for the client.
	ACD	The client is waiting in an ACD queue or is currently being served.
Party	Shows extension number, POTS number or SIP trunk ID that is talking to this client.	

4.10 Firmware Upgrade

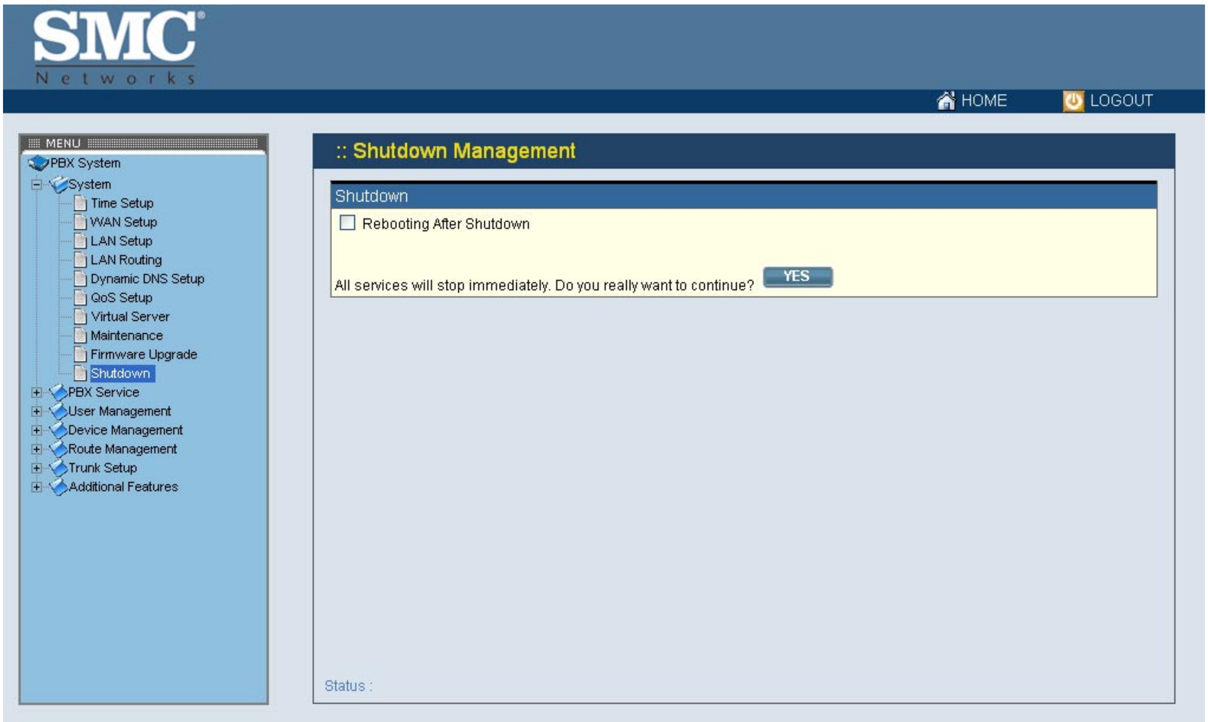
The version of the running PBX firmware could be found in **System** --> **Firmware Upgrade**. To upgrade current firmware, click **Browse** to locate a release file obtained from the vendor, and click **UPGRADE** to have the latest version of PBX firmware.

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



4.11 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 IPPBX system before a power-off.

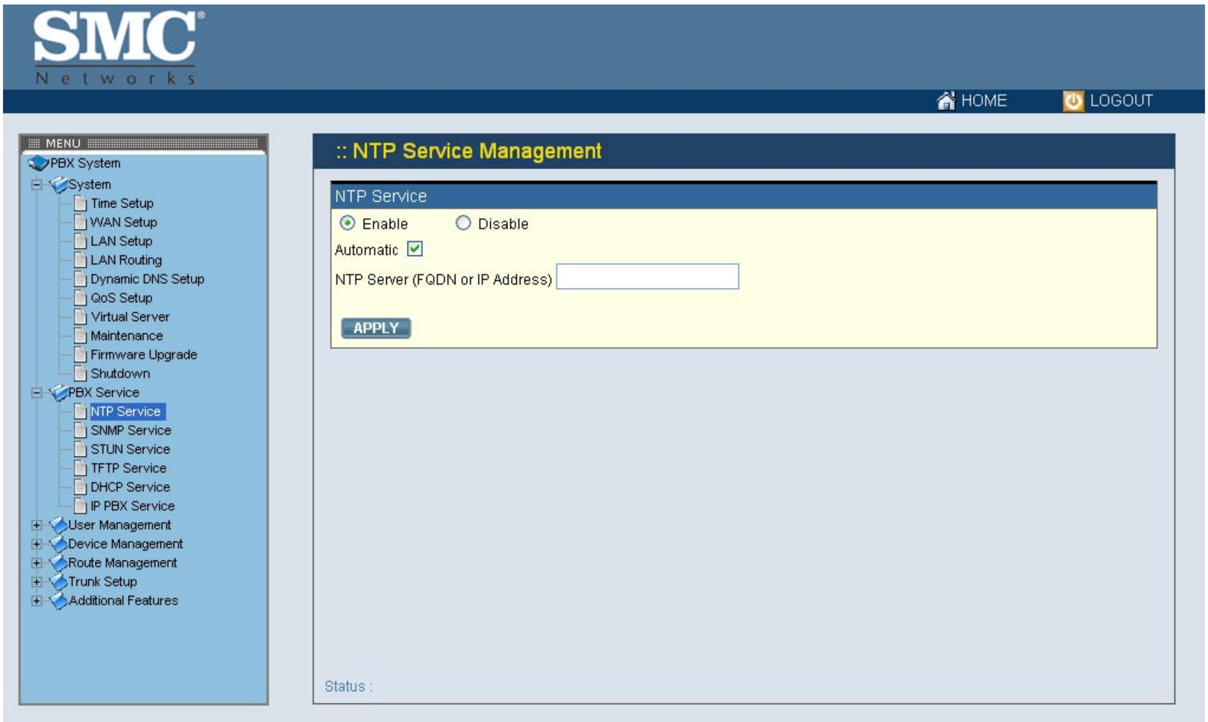


5 Service Configuration

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

5.1 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.



5.1.1 Enable NTP Service

1. Click **Enable**.

2. Select the **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.

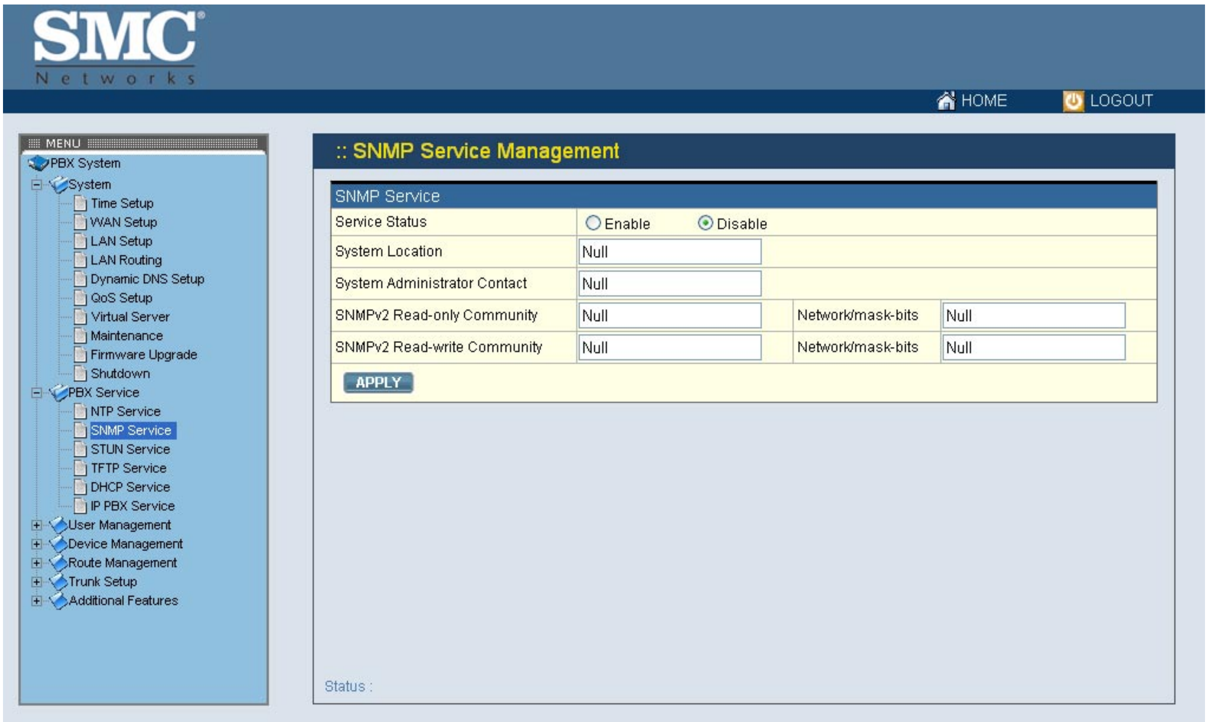
3. Click **APPLY**.

5.1.2 Disable NTP Service

Click **Disable**, and click **APPLY**.

5.2 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.



5.2.1 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 **SNMPv2 Read-write Community**.

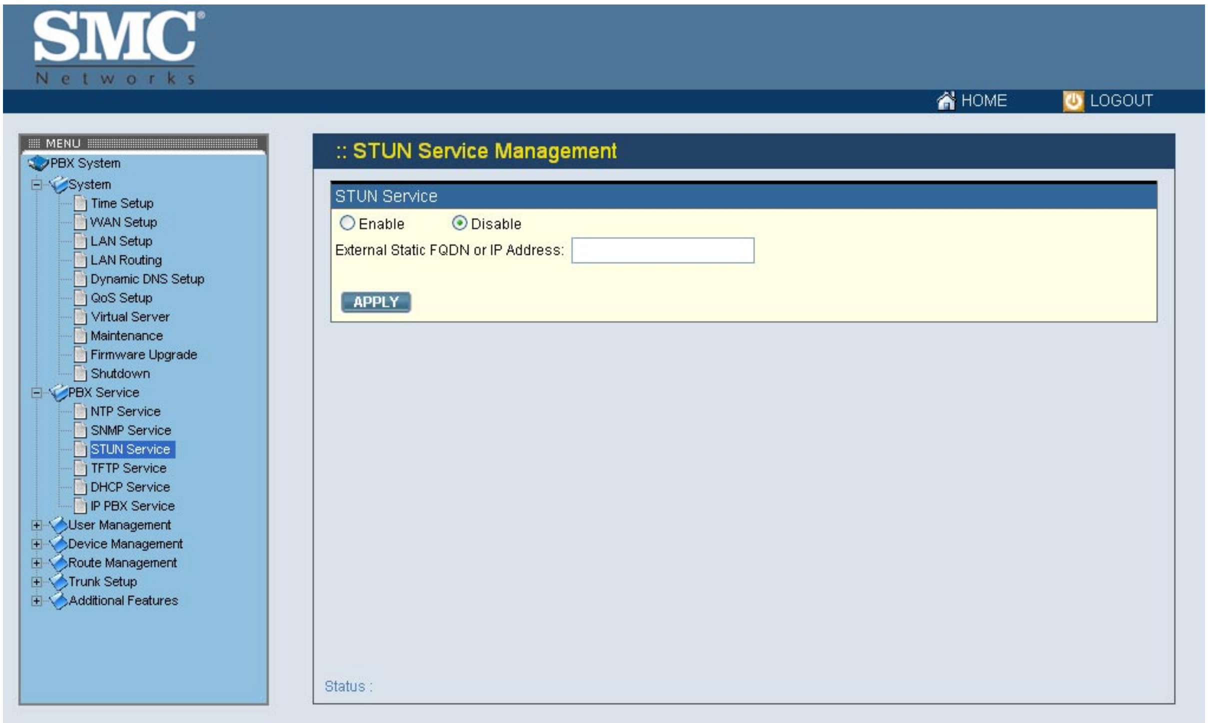
3. Click **APPLY**.

5.2.2 Disable SNMP Service

Click **Disable**, and click **APPLY**.

5.3 STUN Service

IPPBX 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.



5.3.1 Enable STUN Service

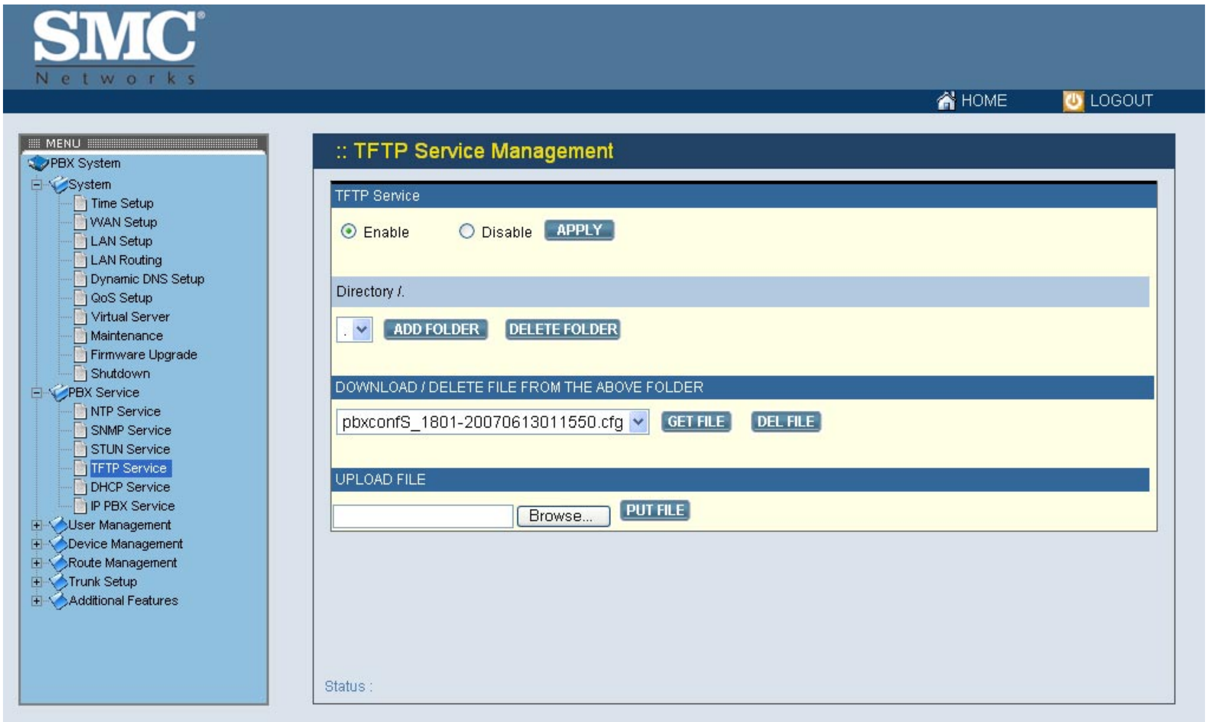
1. Click **Enable**.
2. Enter a fully qualified domain name or the IP address of a STUN server.
3. Click **APPLY**.
4. Go to **Service** --> **IPPBX Service**, and click **RESTART** to reflect the changes.

5.3.2 Disable STUN Service

1. Click **Disable**, and click **APPLY**.
2. Go to **Service** --> **IPPBX Service**, and click **RESTART** to reflect the changes.

5.4 TFTP Service

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



5.4.1 Enable TFTP Service

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

5.4.1.1 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.

5.4.1.2 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/.

5.4.1.3 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.

5.4.1.4 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.

5.4.1.5 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.

5.4.1.6 Upload a File

1. Click a directory in the **Directory** list.
2. Click **Browse**.
3. Select a directory in the **Look in** 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.

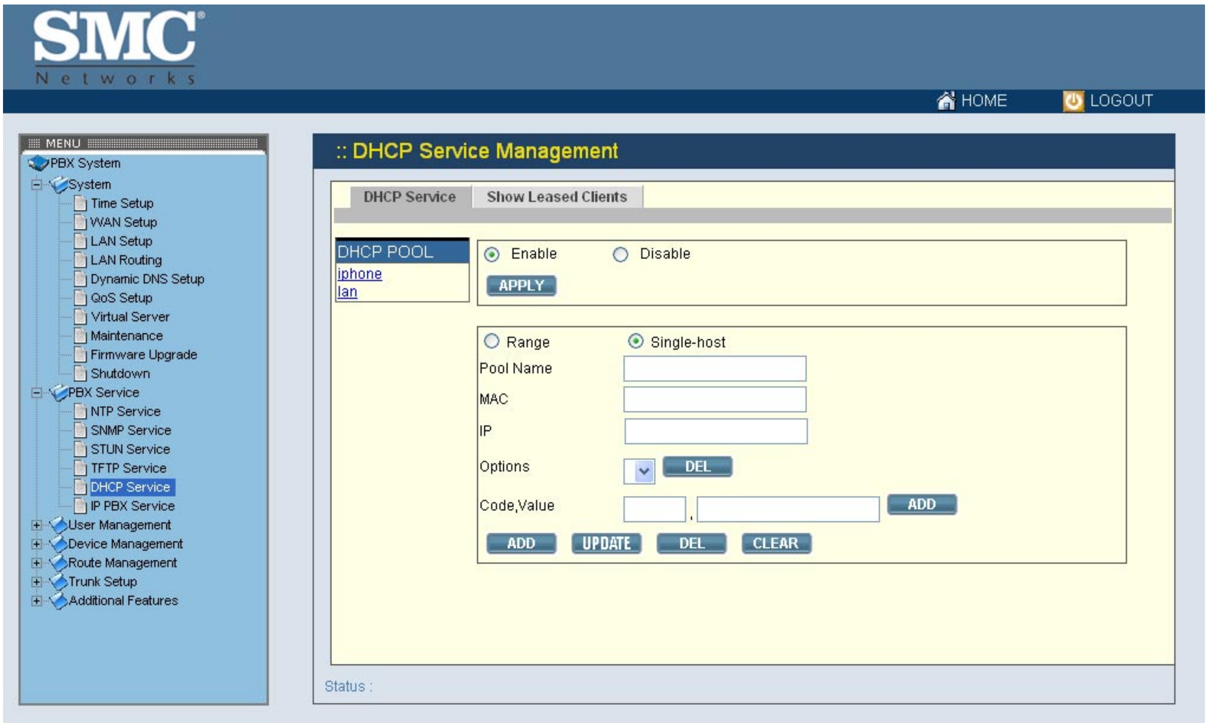
5.4.2 Disable TFTP Service

Click **Disable**, and then **APPLY**.

5.5 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.

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



5.5.1 Enable DHCP Service

Click **Enable**, choose the main interface offering addresses, and then **APPLY** to configure DHCP settings.

5.5.1.1 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[12] 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.

5.5.1.2 Edit DHCP Range

1. Click any pool name in the **DHCP POOL** list to see the settings on the right.

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

2. Edit the settings.

3. Click **UPDATE** to change the settings.

5.5.1.3 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.

5.5.1.4 Show Leased Clients

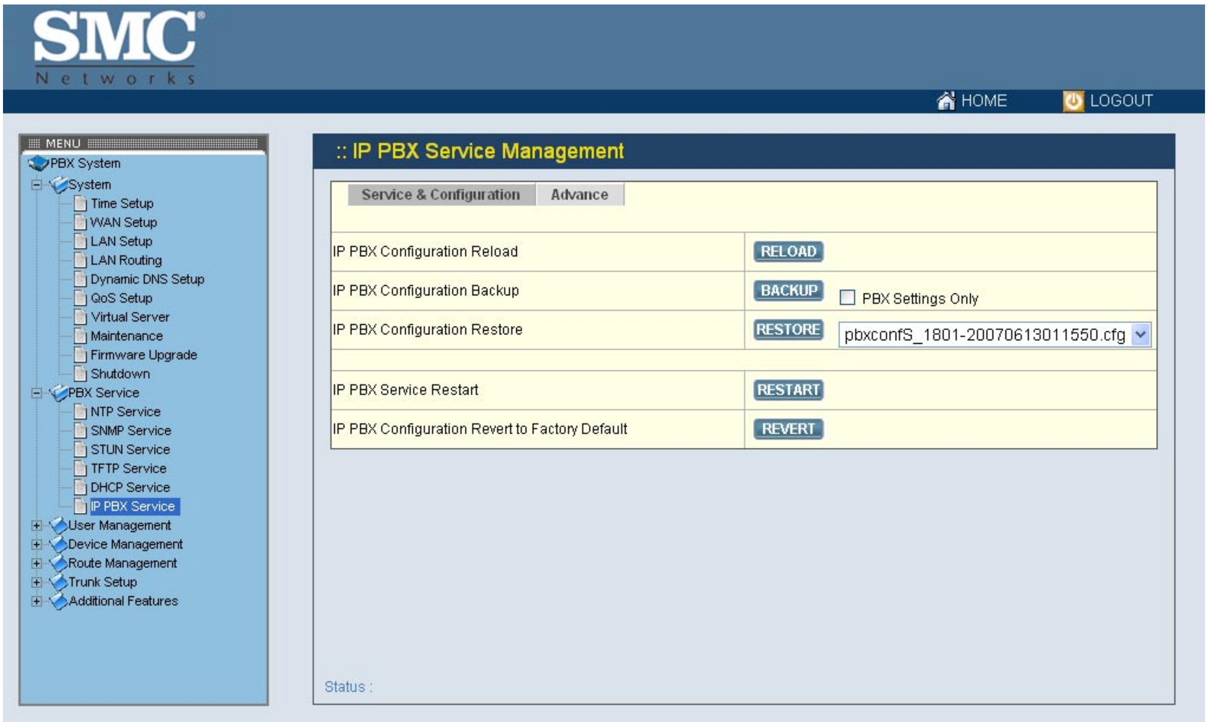
Click the **Show Leased Clients** tab to see all leased LAN IP addresses and client details.

5.5.2 Disable DHCP Service

Click **Disable**, and click **APPLY**.

5.6 IPPBX Service

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



5.6.1 Service & Configuration

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

5.6.1.1 Reload IPPBX Configuration

Click **RELOAD**, and IPPBX 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 IPPBX will reload. This is the most frequently used function in this page since any IPPBX configuration change has to be reloaded to take effect.

5.6.1.2 Backup IPPBX Configuration

Click **BACKUP**, and IPPBX 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.

5.6.1.3 Restore IPPBX Configuration

Click a configuration backup file in the list, click **RESTORE**, and IPPBX will restore the configuration as current setup. After restoring, the system will ask for reboot the PBX service, click **Yes** to reboot IPPBX.

5.6.1.4 Restart IPPBX Configuration

Click **RESTART**, and the IPPBX 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.

5.6.1.5 Revert IPPBX Configuration

Click **REVERT**, and IPPBX will erase current IPPBX settings and revert configuration back to the factory default. Note the reversion affects IPPBX service only, but not other system services such as DHCP, TFTP, and NTP. The backup IPPBX configuration files under TFTP remain intact after reversion, so that one can restore to a specific time if a backup file had been generated 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 one needs to restore the backup configuration after factory default, he/she must download the backup configuration file to the local computer in **Service→ TFTP Service** page.

5.6.2 Advance

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

Table 5.6.2 Advance Settings of IPPBX Services

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 IPPBX 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 terminal trunks)) * 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.
MAX CDR Size	Enter maximum space in KBytes available for CDR.
IP TOS Value	Set the TOS value in the IP header of RTP packets

	originated from IPPBX.
Call Keep Alive	Enter a time in second to confirm the call is ongoing. The time is between 10 and 180 seconds.
Registered Keep Alive	Enter a time in second to confirm the register still exists. The time is between 10 and 180 seconds.
NAT Traversal Keep Alive	Enter a time in seconds to confirm NAT traversal is alive. The time is between 10 and 180 seconds.
Disable WAN Bandwidth Saver	Select to disable attempts to use low-bit-rate codec (G.729A or G.723.1) for remote parties.
Disable DSP saver for LAN calls	Select to honor the preferred codec of caller's phone instead of overriding by DSP utilization concerns.
Enable DNS SRV Resolution	Select to enable looking up IP of dynamic clients or trunks by DNS Service records before their successful registrations.

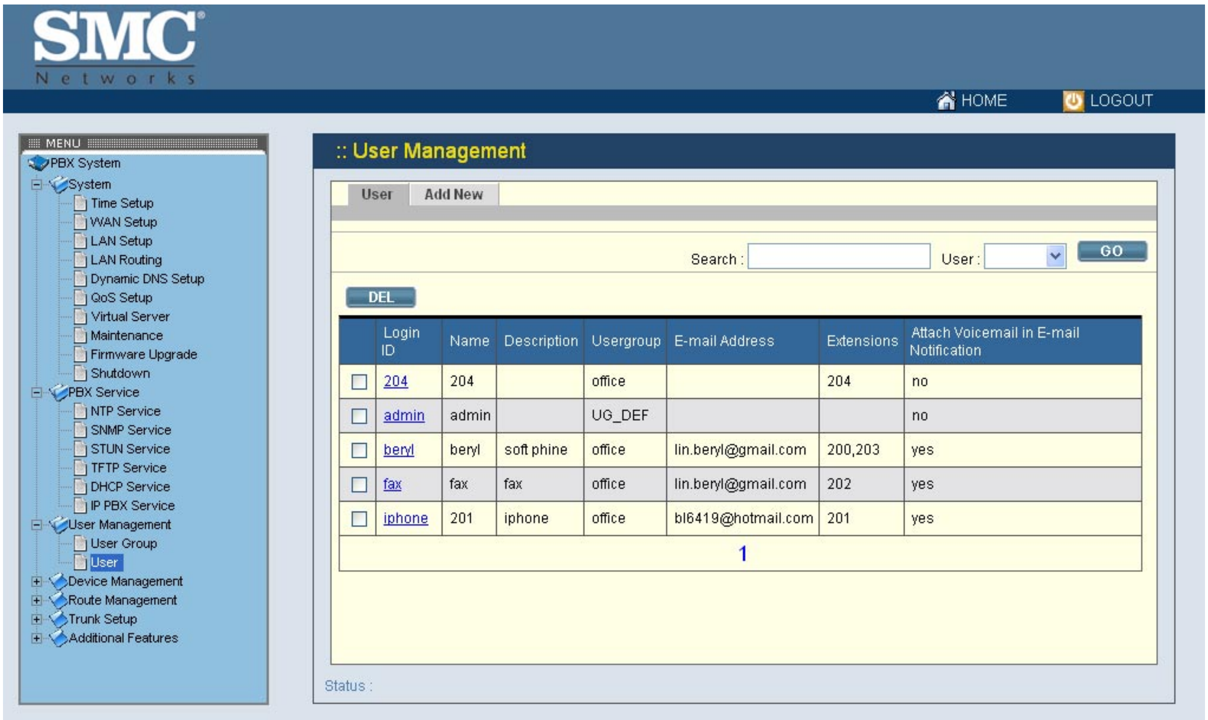
6 IPPBX Configuration

This section introduces steps to provision the IP telephony part of the IPPBX. Note that reloading configuration is required in order to make new configuration effective[13].

6.1 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, delete or search users. Go to **Service** --> **IPPBX Service**, and click **RELOAD** to activate changes.



6.1.1 Add a User

1. Click the **ADD New** tab.
2. Enter settings shown in **Table 6.1**.
3. Click **ADD**.
4. Click **BACK** to see the newly added user in the **Login ID**.

[13] Please refer to **5.6.1.1** for details.

6.1.2 Edit a User

1. Click a user in the **Login ID**.
2. Edit settings shown in **Table 6.1**.
3. Click **UPDATE**.

6.1.3 Delete a User

1. Select a **Login ID**.
2. Click **DEL** to remove the user from the **Login ID**.

6.1.4 Search a User

1. Type a login ID in the **Search** box, or click a login ID in **Users** list.
2. Click **GO** to see the Update page.

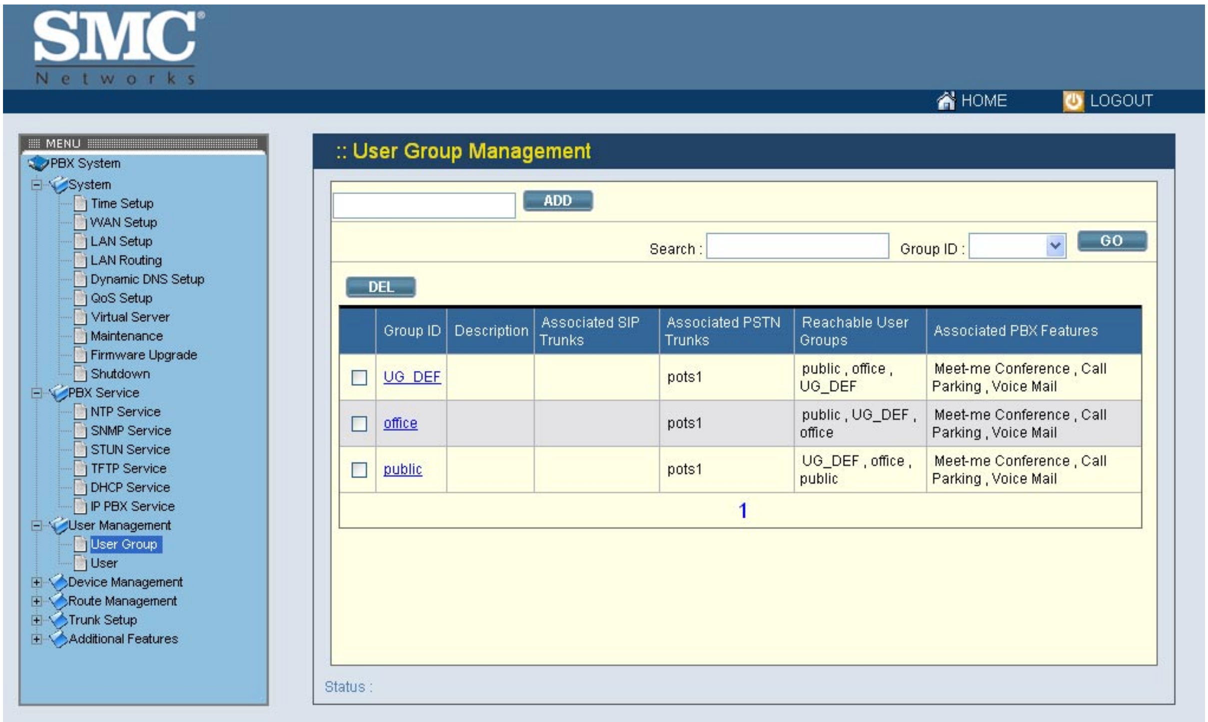
Table 6.1.4 User configuration Settings

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 IPPBX 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 IPPBX 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.

6.2 User Group Configuration

A usergroup is a logical 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, delete or search usergroups. Go to **Service** --> **IPPBX Service**, and click **RELOAD** to activate changes.



6.2.1 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 6.2**.
5. Click **BACK** to return to the USERGROUP MANAGEMENT page.

Now, you can see the newly added usergroup displayed in the **Group ID**.

6.2.2 Edit a User Group

1. Click a usergroup name in the **Group ID**.
2. Edit settings shown in **Table 6.2**.
3. Click **BACK** to see the updated information.

6.2.3 Delete a User Group

1. Select a **Group ID**.
2. Click **DEL** to remove the usergroup from the **Group ID**.

6.2.4 Search a User Group

1. Type a group ID in the **Search** box, or click a group ID in **Group ID** list.
2. Click **GO** to see the Update page.

Table 6.2.4 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. Click **SET** to add/update the information.
Associated Trunks[14]	Select routegroups and outbound trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time. **Routegroup**: display available routegroups. **Trunk**: Display available trunks. Select **blocking** to block calls that meet the route patterns in selected routegroup. Loop back is especially for hop on and off function. Select **loopback** to have the number after performing the route pattern in selected routegroup to go through the rules in Associate Trunks again. Once find the matched rule, the number will go to the relevant trunk. **Group ID**: The default number is “0”. A trunk with Group ID “0” does not form a balance group with any

[14] Please refer to **6.6**, **6.7** and **6.8** for details.

	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. Click or to add or delete the associate trunks. After add all trunks, click **APPLY**. ☞ 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 a usergroup and click that is 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[15]	Select PBX features enabled to this usergroup. Here vm stands for Voice Mail, mm for Meet-me Conference, and parkedcalls for Call Parking. ☞ 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.
Auth. Dial Passcode	Select and enter a password in number for caller to have the same privilege as this usergroup to dial out.

6.3 Device Configuration

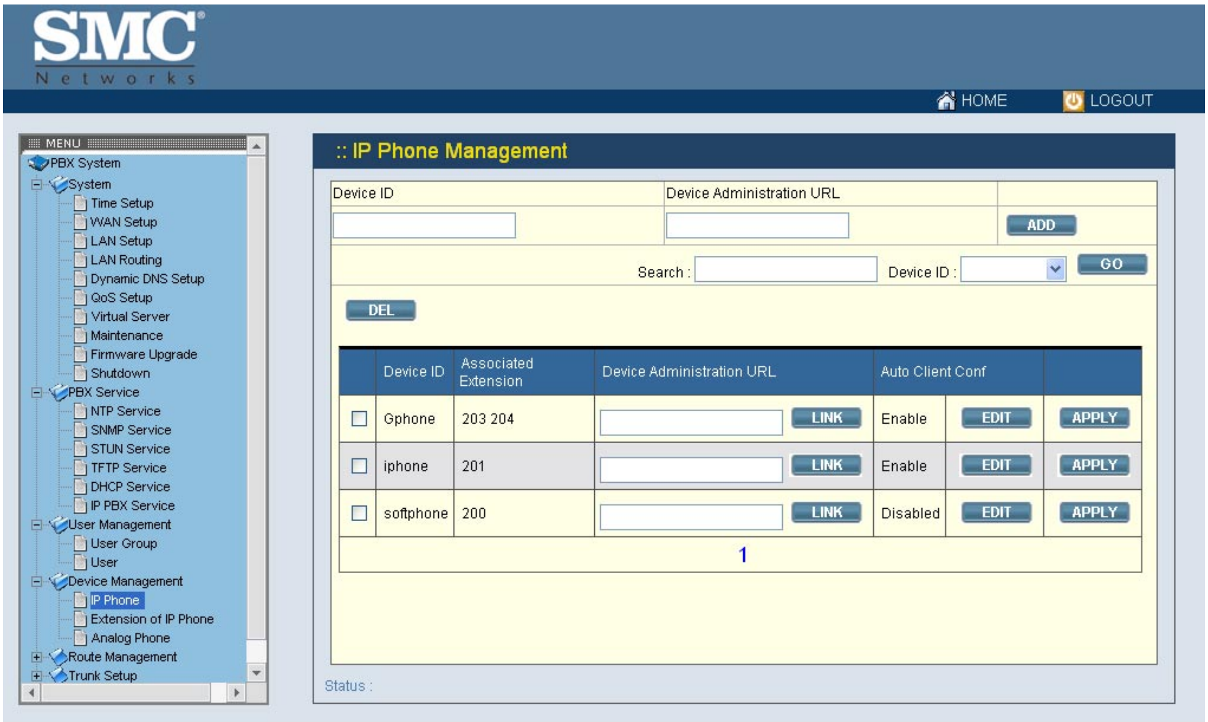
A device could be an IP phone, gateway, analog telephone adapter, or even another IPPBX, etc. It has one or more extensions to be registered to the IPPBX.

6.3.1 IP Phone

The DEVICE PHONE MANAGEMENT page lets the administrator to create IP Phone

[15] Please refer to **7** for details.

devices. Before a device can be reached from the IPPBX, 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, delete and search devices. Go to **Service** --> **IPPBX Service**, and click **RELOAD** to activate changes.



6.3.1.1 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**.

6.3.1.2 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 6.3.1** 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.

3. Click **ENABLE** to enable ACC function. **Enable** will be displayed in the **Auto Client Conf** column. Click **EDIT** and then **DISABLE** to disable the function.

6.3.1.3 Delete a Device

1. Select a **Device ID**.
2. Click **DEL** to remove the device from the **Device ID**.

6.3.1.4 Search a Device

1. Type a group ID in the **Search** box, or click a device ID in **Device ID** list.
2. Click **GO** to see the data.

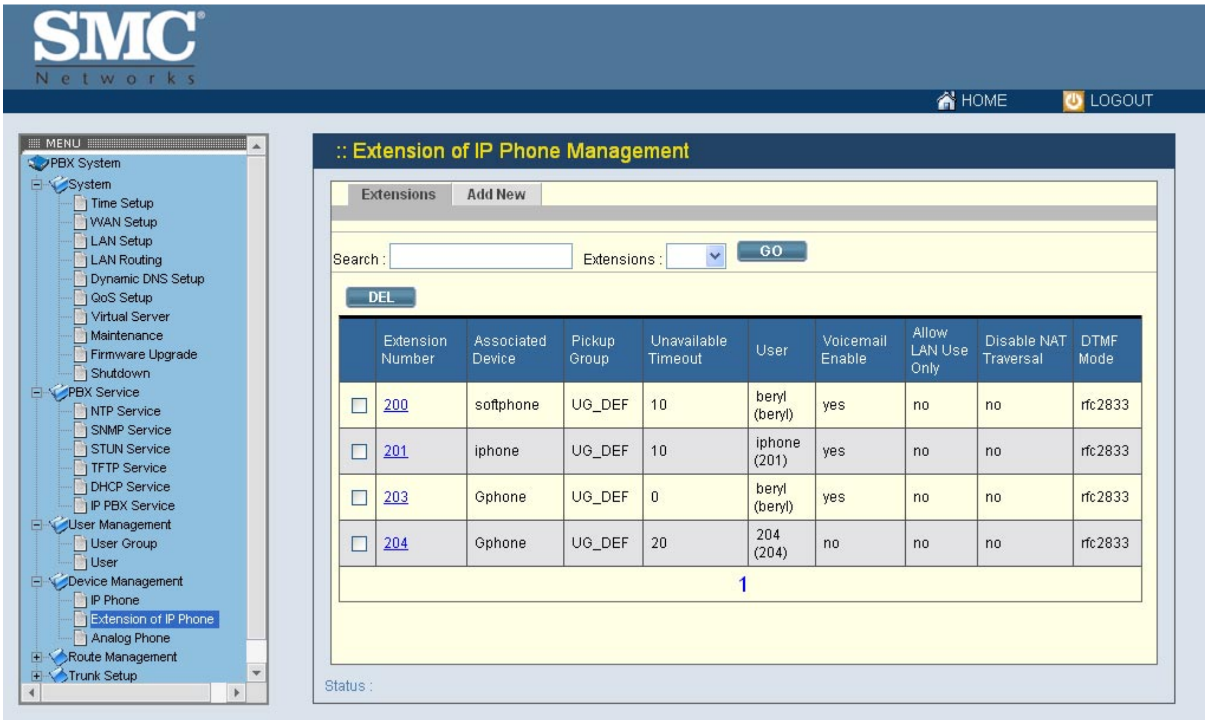
Table 6.3.1 ACC (Automatic Client Configuration) Settings

<table>
<tr><th>Field</th><th colspan="2">Description</th></tr>
<tr><td>Device</td><td colspan="2">A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.</td></tr>
<tr><td>Vendor Prefix</td><td colspan="2">The vendor Prefix's for SMC's IP Phones are as follows:
SMCDSP-200 = dsp200
SMCDSP-205 = dsp205
SMCWSP-100 = wsp100</td></tr>
<tr><td>MAC Address</td><td colspan="2">MAC address of the device.</td></tr>
<tr><td>Codec Preference</td><td colspan="2">Preference order of supported codec and packet times of the phone.</td></tr>
<tr><td rowspan="6">Redundant Servers</td><td colspan="2">Enter the information as follows.</td></tr>
<tr><td>Proxy IP/FQDN</td><td rowspan="2">Specify IP address (or fully qualified domain name) and UDP port of the remote IPPBX.</td></tr>
<tr><td>Proxy Port</td></tr>
<tr><td>Registrar IP/FQDN</td><td rowspan="2">Specify IP address (or fully qualified domain name) and UDP port of the remote IPPBX</td></tr>
<tr><td>Registrar Port</td></tr>
<tr><td>SIP Domain</td><td>Specify the SIP domain used by the proxy and registrar. If not specified, IP address will be used as the domain by default.</td></tr>
<tr><td>Enable Voice Activity Detection (VAD)</td><td colspan="2">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.</td></tr>
<tr><td>DTMF mode</td><td colspan="2">Choose a DTMF mode used by the phone.</td></tr>
</table>

6.3.2 Extension of IP Phone

The EXTENSION MANAGEMENT page allows the administrator create extensions. Select **Device** --> **Extension of IP Phone**, and one can add, edit, delete and search extensions. Go to **Service** --> **IPPBX Service**, and click **RELOAD** to activate changes. IPPBX also offers its own extensions Line-in-use Call Back function. For example, extension A and B are within one IPPBX or a stacked cluster. When extension A calls extension B, and extension B is in use. Extension A can press 1 to be in the waiting list. Once extension B finishes the call, IPPBX will ring extension B within a minute. When extension B is picked up, IPPBX will ring extension A to connect the call. Extension A call also press * to cancel, and will connect to voice mail.

Note: The IP Phone should disable call waiting (multi-line) function in order to activate the line-in-use call back function. If the extension B does not disable the call waiting function on IP Phone, when extension A calls to extension, and extension B is in use, the second line of extension B's IP Phone will ring.



6.3.2.1 Add an Extension

1. Click the **ADD New** tab to set an extension.
2. Enter settings shown in **Table 6.3.**.
3. Click **ADD** to see the newly added extension.

6.3.2.2 Edit an Extension

1. Click an extension in the **Extension Number**.

2. Edit settings shown in **Table 6.3.**.

3. Click **UPDATE** to see the updated information.

6.3.2.3 Delete an Extension

1. Select an extension numbers.

2. Click **DEL** to remove the extension from the **Extension Number**.

6.3.2.4 Search an Extension

1. Type an extension number in the **Search** box, or click an extension number in **Device ID** list.

2. Click **GO** to see the data.

Table 6.3.2 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[16]	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.
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.
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.
Max Voicemail Space	Enter maximum space in KBytes for voicemail.
Unavailable Timeout	Timeout for ringing before a call is answered.

[16] Please refer to **6.1** for details.

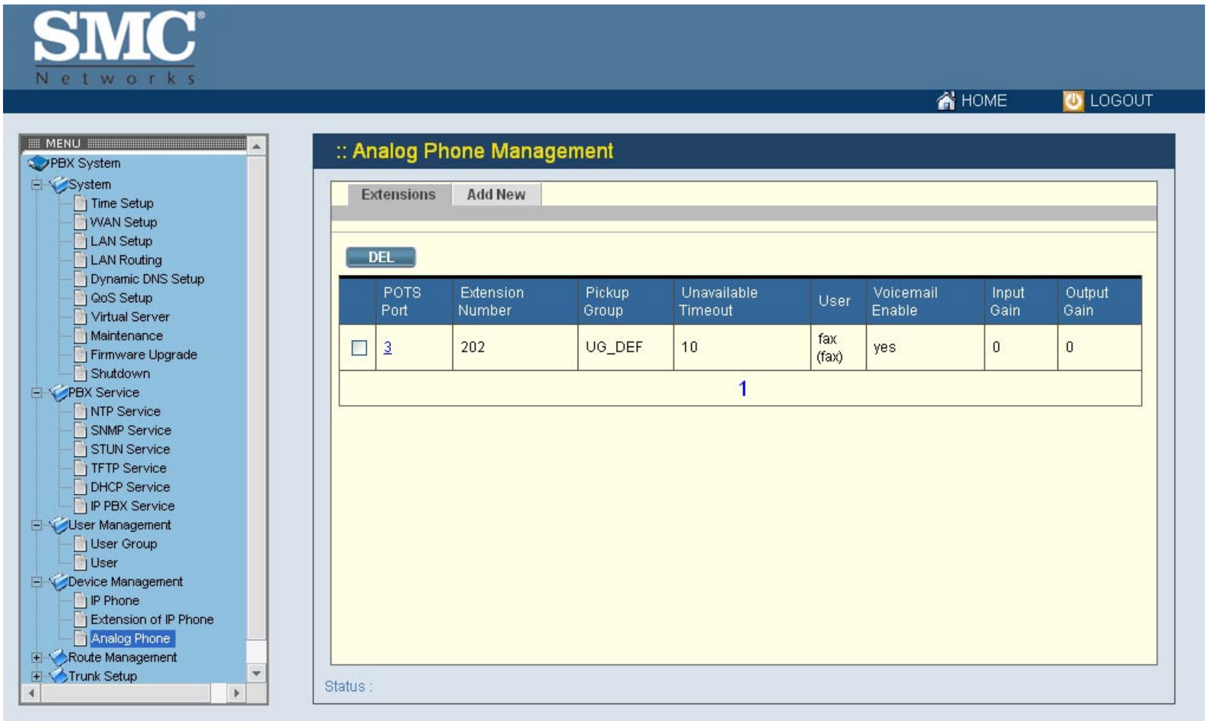
Allow LAN Use Only	Check to reject registration and calls from WAN in a SIP ID same as the extension number. I.e., this extension must be on LAN.
Disable NAT Traversal	IPPBX uses NAT traversal for outgoing traffics by default. Select to disable NAT traversal if there is a machine that could handle NAT issues.
Call Keep Alive	Select to check if the call is still ongoing in a certain time.
Registered Keep Alive	Select to check if the register still exists in a certain time.
Try Peer-to-peer RTP	If click **YES**, IPPBX 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 IPPBX. However, only SIP INFO DTMF mode phones should enable this since other DTMF modes require IPPBX 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. ☞ In-band DTMF mode consumes the limited DSP resource when using a highly compressed codec, such as G.729 or G.723.1. Therefore, calls will not connect with such setting if DSP is not installed. Although using a low-complexity codec such as G.711 does not require DSP, DTMF detection still takes considerable CPU resource and impacts several system specs. Be cautious when configuring an extension with in-band DTMF mode.
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 .
Block SIP redirection from the extension	Select to ignore the forward settings of IP phone and honor the forward settings of IPPBX.
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 ringing 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 . ☞ The time must be shorter than **Unavailable Timeout**, or the function will not work normally.
Play Unavailable/Line-in-use 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.

6.3.3 Analog Phone

The ANALOG PHONE MANAGEMENT page allows the administrator to create analog phones. Select **Device** --> **Analog Phone**, and one can add, edit, and delete analog phones. Go to **Service** --> **IPPBX 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 6.3.**.



6.3.3.1 Add an Analog Phone

1. Click the **ADD New** tab to see the detailed ANALOG PHONE MANAGEMENT page.
2. Enter settings shown in **Table 6.3.**.
3. Click **ADD** to see the newly added analog phone in the **Extension Number**.

6.3.3.2 Edit an Analog Phone

1. Click a port in **POTS Port**.
2. Edit settings shown in **Table 6.3.**.
3. Click **UPDATE** to see the edit information.

6.3.3.3 Delete an Analog Phone

1. Select a **POTS Port**.

2. Click **DEL** to remove the extension from the **POTS Port**.

Table 6.3.3 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[17]	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.
Max Voicemail Space	Enter maximum space in KBytes for voicemail.
T.38 Enabled	Click **Auto**, **Enable** or **Disable** 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.
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.

[17] Please refer to **6.1** for details.

	☞ 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 ringing 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 . ☞ The time must be shorter than **Unavailable Timeout**, or the function will not work normally.
Play Unavailable/Line-in-use 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.

6.4 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 6.4** explains digit set and wildcard characters.

Note: The “#” in route pattern is for some PSTN saver lines that may set “#” as their dial pattern. For most of the IP Phones, press "#" will immediately send out the dialed number.

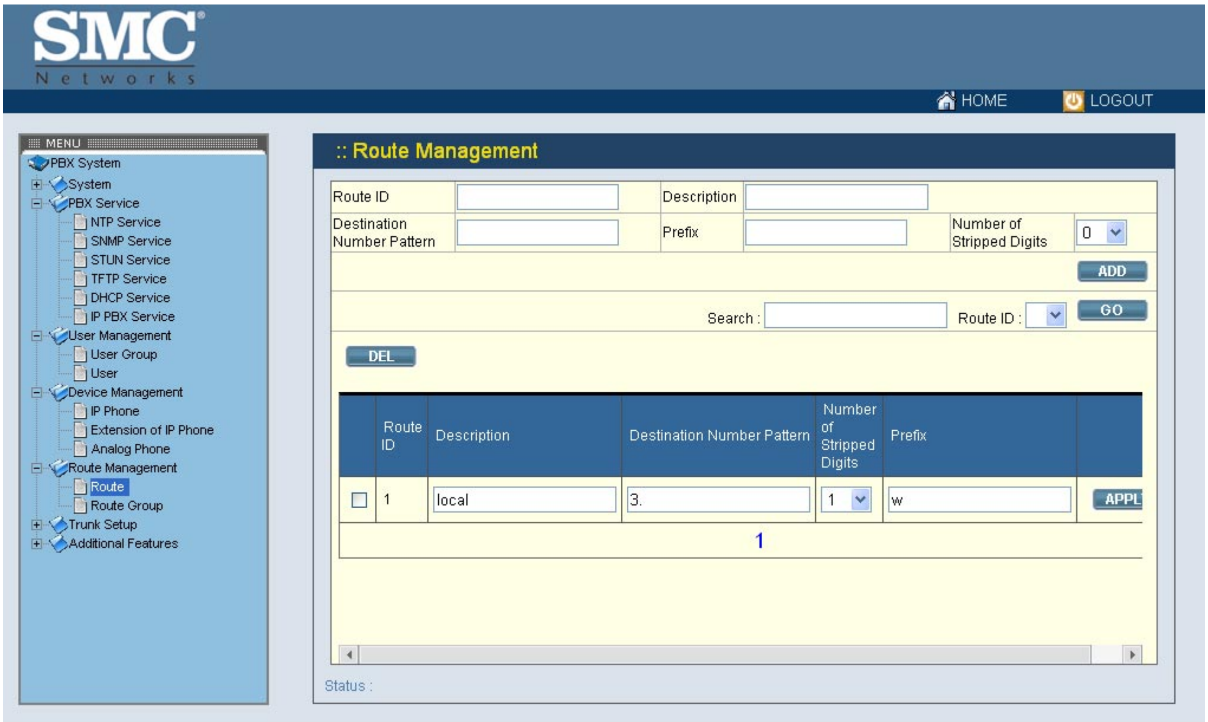


Table 6.4.1 Digit Set and Wildcard Characters for Route Patterns

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, delete and search routes in the Route Management page. The administrator can click **Route ID**, **Description**, **Destination Number Pattern**, **Number of Stripped Digits** and **Prefix** arrows to sort the order of data. Go to **Service → IPPBX Service**, and click **RELOAD** to activate changes.

6.4.1 Add a Route

1. Enter settings shown in **Table 6.4.1**.
2. Click **ADD** to see the newly added route in the **Route ID**.

6.4.2 Edit a Route

1. Edit settings shown in **Table 6.4.1** in a row.
2. Click **APPLY** in the row to update the settings.

6.4.3 Delete a Route

1. Select a **Route ID**.
2. Click **DEL** to remove the route from the **Route ID**.

6.4.4 Search a Route

1. Type a route ID in the **Search** box, or select a route ID in **Route ID** list.
2. Click **GO** to see the data.

Table 6.4.1 Route Configuration Settings

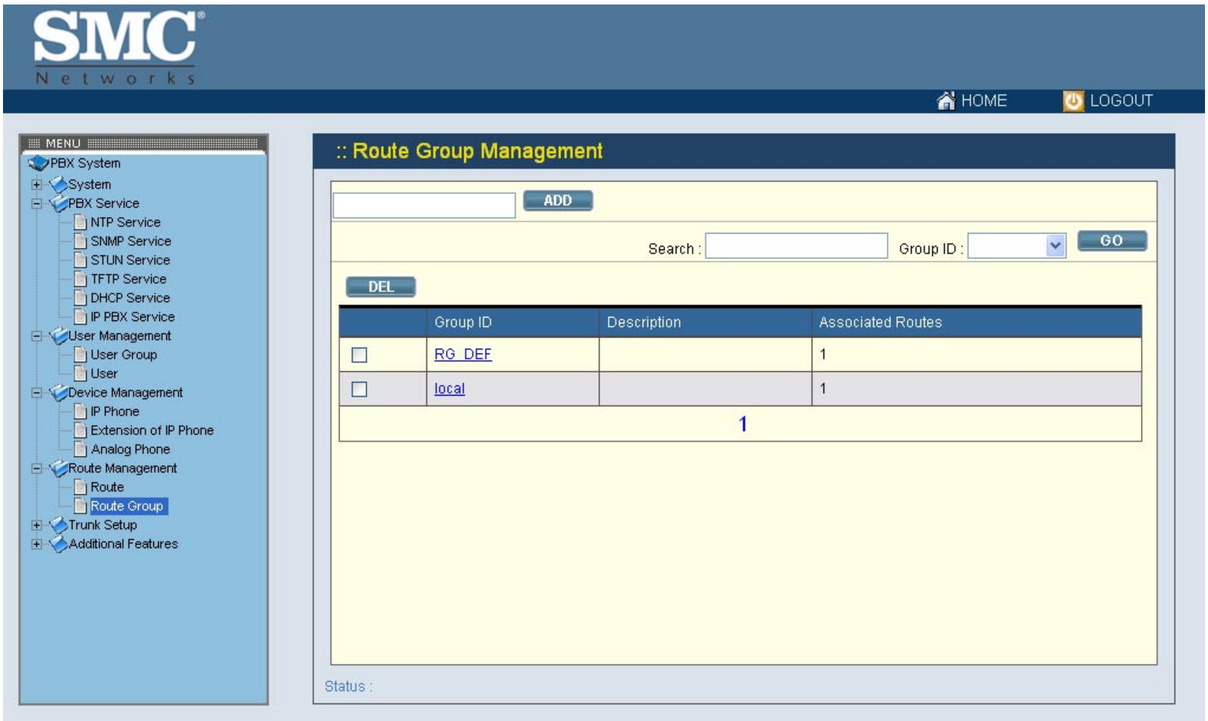
Field	Description
Route ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 16 characters maximum.
Description	Arbitrary description information.
Destination Number Pattern	A destination number pattern consisting of digits, digit set, and wildcard characters, e.g. 9NXXXXXX matches any 7-digit called number starting from a digit larger or equal to 2 and with an extra prefix digit 9.
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.
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.

6.5 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, delete and search routegroups in the ROUTE GROUP MANAGEMENT page. Go to **Service → IPPBX Service**, and click **RELOAD** to activate changes.



6.5.1 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 6.5**, and click **BACK**.
4. The newly added route group should be displayed in the **Group ID**.

6.5.2 Edit a Route Group

1. Click a route group name in **Group ID**.

2. Edit settings shown in **Table 6.5**.

3. Click **BACK** to see the updated information.

6.5.3 Delete a Route Group

1. Select a **Group ID**.

2. Click **DEL** to remove the route group from the **Group ID**.

6.5.4 Search a Route Group

1. Type a group ID in the **Search** box, or click a group ID in **Group ID** list.

2. Click **GO** to see the Update page.

Table 6.5 Routegroup Configuration Settings

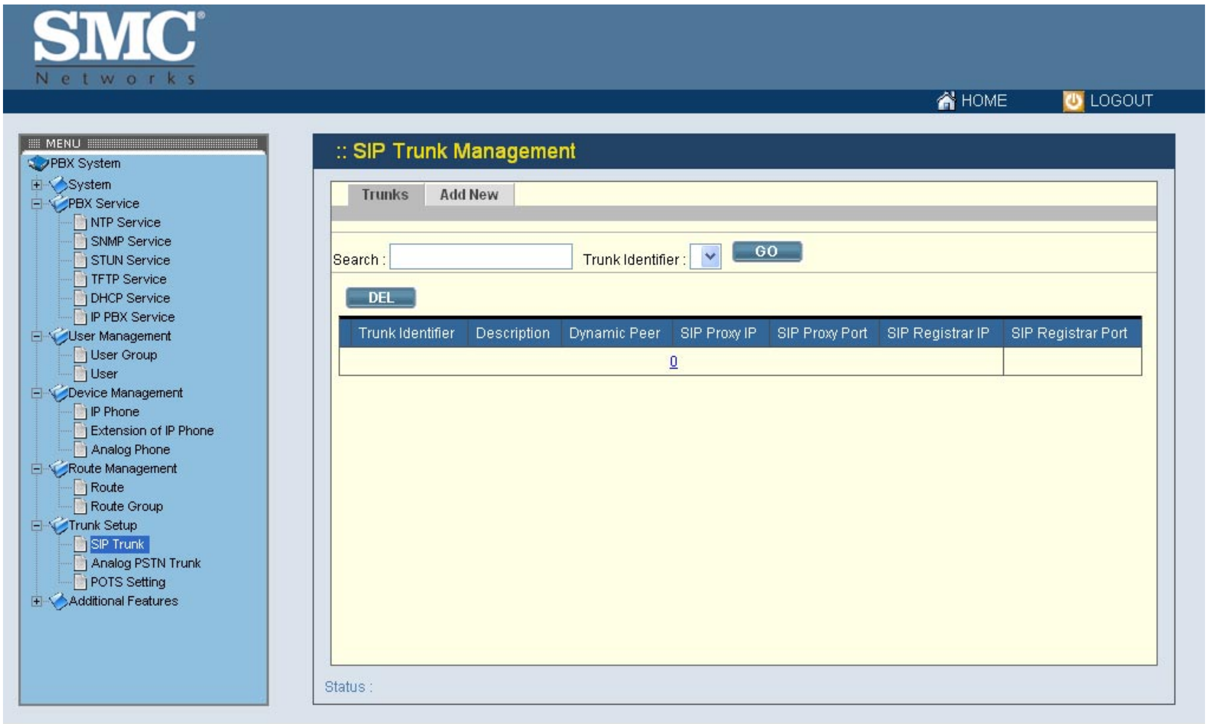
Field	Description
Group ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description information. Click **SET** to add/update the information.
Associated Routes[18]	Select routes belonged to this routegroup. Click  or  button to add or remove a route to or from the routegroup. The right box lists current selected routes. Click **SET** to update the information. Note the order of the selected routes is important since it decides which route is 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.

6.6 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 IPPBX to reach its extensions and PSTN ports.

18 Please refer to **6.4** for details.

The SIP TRUNK MANAGEMENT page allows the administrator to configure SIP trunks used by IPPBX. Select **Trunk --> SIP Trunk**, and one can add, edit, delete and search SIP trunks. Go to **Service --> IPPBX Service**, and click **RELOAD** to activate changes.



6.6.1 Add a SIP Trunk

1. Click the **Add New** tab.

2. Enter settings shown in **Table 6.6**.

3. Click **ADD** to see the newly added SIP trunk in the **Trunk Identifier**.

6.6.2 Edit a SIP Trunk

1. Click an identifier in **Trunk Identifier**.

2. Edit settings shown in **Table 6.6**.

3. Click **UPDATE** to change the information.

6.6.3 Delete a SIP Trunk

1. Click the **Trunks** tab, and select a trunk identifier.

2. Click **DEL** to remove the SIP trunk from the **Trunk Identifier**.

6.6.4 Search a SIP Trunk

1. Type a trunk identifier in the **Search** box, or click a trunk identifier in **Group ID** list.

2. Click **GO** to see the Update page.

Table 6.6.1 SIP Trunk Configuration Settings

Field	Description
Trunk Identifier	A unique number consisting of digits only. Usually give the phone number issued by the ITSP for consistency.
Description	Arbitrary description 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 IPPBX at a remote site with dynamic IP address. Once the remote IPPBX registers, calls from local to remote can be made reversely over the trunk.
SIP Proxy IP 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 IPPBX or an ITSP account, however, may be not required in case of a SIP gateway.
SIP Registrar IP 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).
DID by Privilege	Select to configure **DID Prefix** and **DID Stripping** to have the incoming calls directed to the trunk.
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 **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 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. ☞ If selecting **By Number**, the "number" being manipulated for extension DID is the called (destination) number. As a result, one should confirm what prefix, usually the area code, would be given by the service provider side so that a correct stripping could be configured accordingly.
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 **By Number**, 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. Click **All** to strip all digits of the original called number.
Centrex DID	Please refer to Section 6.6.5.
IVR List[19]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID. Leave it blank and the system will automatically create an IVR for the trunk.
Usergroup[20] 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.
Disable Fast Bridging	Select to disable media relay.
Advanced Settings	Select to see more settings shown below.

[19] Please refer to **7.11** for details.
[20] Please refer to **6.2** for details.

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 IPPBX 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 IPPBX. However, only SIP INFO DTMF mode should enable this since other DTMF modes require IPPBX being RTP relay server to support in-line transfer.
From Caller ID	Send the entered number as Caller ID.
Bandwidth Sensitive	Indicate the trunk is over a bandwidth-sensitive link, e.g. across Internet.
Bandwidth Limitation (kbps)	Leave it blank to disable or, specifies a limit of bandwidth in kbps for call admission.
Call Admission Control	Select to enable the limitation of concurrent calls for a SIP trunk.
Allow <*number*> Concurrent Call(s)	Select **Call Admission Control** and enter a number for allowed concurrent calls.
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	Override default User-Agent header content.
Enable ENUM Resolution	Select to use ENUM resolution, or leave it as blank.
Clear Bindings Prior Registration	Select if failed to the registration, and cannot identify any abnormal settings.
Disable NAT Traversal	IPPBX uses NAT traversal for outgoing traffics by default. Select to disable NAT traversal if there is a machine that could handle NAT issues.
Anti-SIP Blocking	Select to anti the SIP package blocking from ISP.
Gateway Trunk	Select to loose the checking of incoming calls.
Call Keep Alive	Select to check if the call is still ongoing in a certain time.

Registered Keep Alive	Select to check if the register still exists in a certain time.
Delay Before/After Answering	Delay in seconds before and after answering a call from SIP trunk.
RFC2833 Payload Type	The default payload type is 101. Enter a value between 96 and 127 to change the default payload type when selecting DTMF mode as **RFC 2833** or **Not Sure**.

6.6.5 Digitmap Configuration

The digitmap is for **Centrex DID** in SIP trunks. A pattern consists of digits 0-9 (including "-"), "*", "#", digit set, and wildcard characters like ".", "X", "Z", and "N".

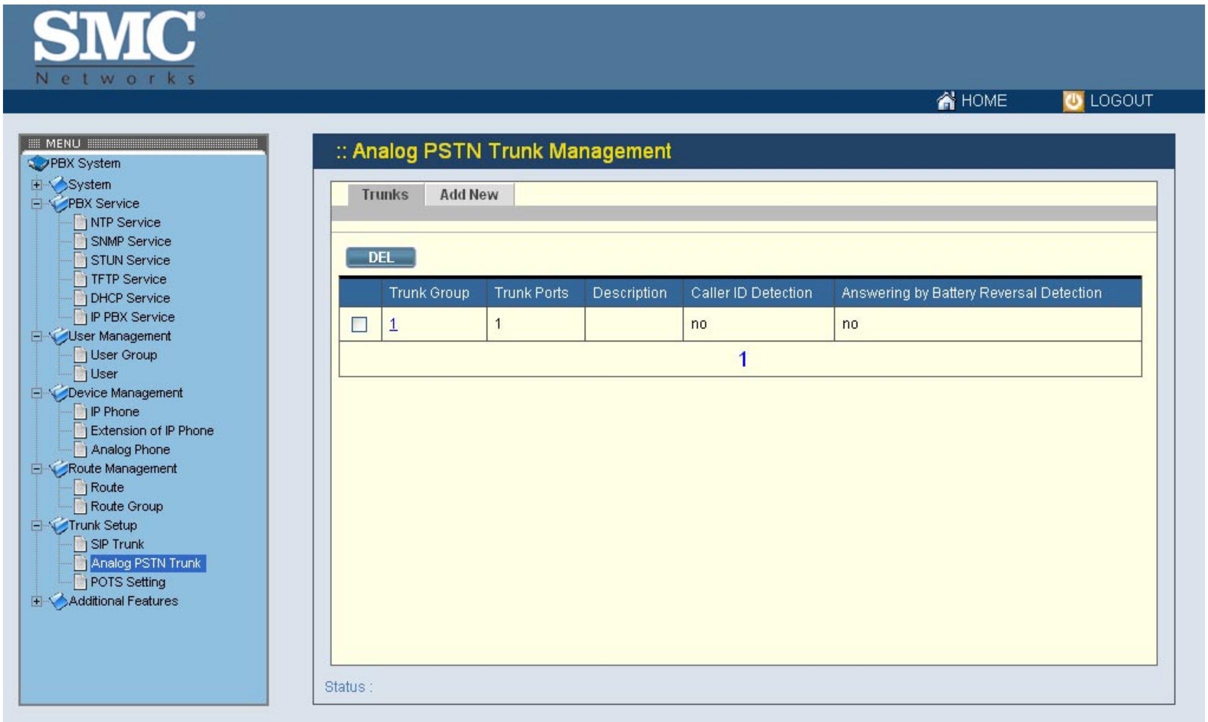
Note: The "#" in the pattern is for some PSTN saver lines that may set "#" as their dial pattern. For most of the IP Phones, press "#" will immediately send out the dialed number.

Field	**Description**
Digitmap ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 17 characters maximum.
Destination Number Pattern	A destination number pattern consisting of digits, digit set, and wildcard characters, e.g. 9NXXXXXX matches any 7-digit number starting from a digit larger or equal to 2 and with an extra prefix digit 9.
Prefix	A sequence of digits to be prefixed to the number after stripping. Using 9NXXXXXX as an example pattern with number of stripped digits equal to 1 and prefix 1408, dialing 95270001 will change to 14085270001. 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.
Number of Stripped Digits	Number of leading digits to be stripped from the original number when matches this digitmap. Using 9NXXXXXX as an example pattern with number of stripped digits equal to 1, dialing 95270001 will be stripped and change to 5270001.

6.7 Analog PSTN Trunk Configuration

An Analog PSTN trunk group is a logical group of one or more FXO or FXS PSTN subscriber lines connecting to FXO or FXS ports on IPPBX.

The Analog PSTN TRUNK MANAGEMENT page allows the administrator to configure PSTN trunks. Select **Trunk → Analog PSTN Trunk**, and one can add, edit and delete PSTN trunks. Go to **Service → IPPBX Service**, and click **RELOAD** to activate changes.



6.7.1 Add an Analog PSTN Phone

1. Click the **Add New** tab.
2. Enter settings shown in **Table 6.7**.
3. Click **ADD** to see the newly added FXO PSTN trunk in **Trunk Group**.

6.7.2 Edit an Analog PSTN Phone

1. Click a trunk group in **Trunk Group**.
2. Enter settings shown in **Table 6.7**.
3. Click **UPDATE** to change the information.

6.7.3 Delete an Analog PSTN Phone

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

Table 6.7 Analog PSTN Trunk Configuration Settings

Field	Description
Trunk Group	ID number of this PSTN trunk group. A valid number

	ranges from 1 to 32. It should not overlap with existing PSTN trunk groups.
Trunk Type	Select the port type, FXO or FXS. If selecting FXS, users can see **By Number** and **By Privilege** in the **DID of Extension** list, and be able to configure **DID Prefix** and **DID Stripping**.
Trunk Ports	FXO and FXS port indices grouped by this PSTN trunk, such as 1 or 1,2 or 1-3, etc. Maximum port index depends on the number of physical ports available.
Description	Arbitrary description information.
Port Selection	Click to search for an available port in the group. **Rotating** means to force ports being selected by turns to even cost.
Caller ID Detection	Select to detect the Caller ID calling from PSTN lines.
Answering by Battery Reversal Detection	If the PSTN service provides battery reversal, select to count billable time starting from the call is answered. ☞ Clear the check box, if you are not sure whether the PSTN service provides the function.
DID of Extension	When enabled DID, clicks an extension in the list to be an unconditional destination for incoming calls to this trunk. The PSTN numbers of the included ports are therefore regarded as the direct line numbers of the extension. If **FXS** is selected in the **Trunk Type** list, you can also click **By Number** or **By Privilege**, and then enter configurations in **DID Prefix** and **DID Stripping** to have the incoming calls directed to the corresponding extension or trunk derived by number manipulation. The PSTN trunk numbers is 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. ☞ If selecting **By Number**, the "number" being manipulated for extension DID is the called (destination) number. As a result, one should confirm what prefix, usually the area code, would

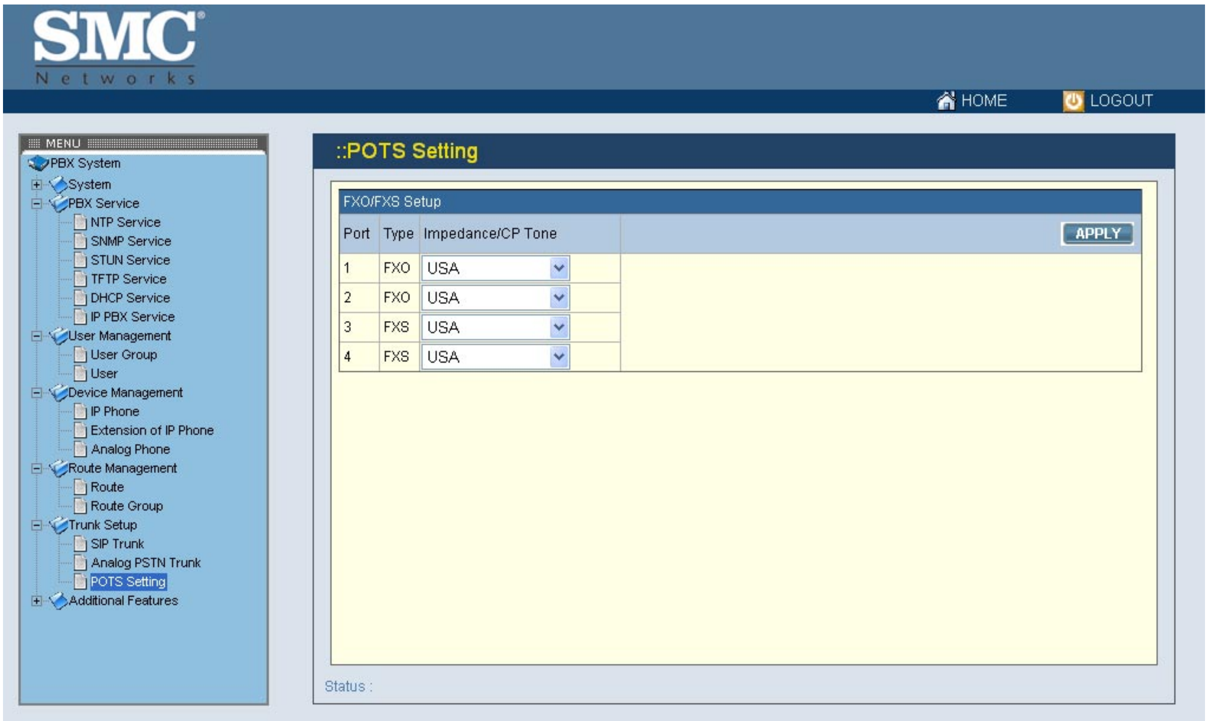
	be given by the service provider side so that a correct stripping could be configured accordingly.
DID Prefix	A digit string to be prefixed to the incoming called number after stripping.
DID Stripping	A number of leading digits to be stripped from the original called number. If prefix or stripping has been given but **DID of Extension** is not **By Number** or **By Privilege**, 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. Click **All** to strip all digits of the original called number.
IVR List[21]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID. Leave it blank and the system will automatically create an IVR for the trunk.
Usergroup[22] 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 any appropriate usergroups to select initially. One can come back later to revise it, once the expected usergroups are added.
Advanced Settings	Select to see more settings shown below.
Input/Output Gain	Voice amplification or attenuation in dB scale to adjust input/output volume of a PSTN line.
Minimum Disconnection Tone	Minimum volume level of the disconnection tone. If a PSTN trunk is found to have disconnection problem and voice sounds low, choose a lower dB.
Delay Before/After Answering	Delay in seconds before and after answering a call from PSTN trunk.
Call Time Restriction	Enter a number (1-1440) in minute to limit the call period. If a call lasts longer than the time, IP PBX will hang up the phone. Enter 0 or leave it blank to disable the function.
1st Frequency Of On-cycle Tone	Enter a number in Hz for call disconnection on-cycle

[21] Please refer to **7.11** for details.
[22] Please refer to **6.2** for details.

2nd Frequency Of On-cycle Tone	tone.
1st Frequency Of Off-cycle Tone	Enter a number in Hz for call disconnection off-cycle tone.
2nd Frequency Of Off-cycle Tone	
Min Duration Of On-cycle Tone	Enter a time in ms for minimum or maximum duration of on-cycle tone.
Max Duration Of On-cycle Tone	
Min Duration Of Off-cycle Tone	Enter a time in ms for minimum or maximum duration of off-cycle tone.
Max Duration Of Off-cycle Tone	
Forced Disconnection If This Duration (ms) Of Silence Detected	Enter a time in ms for disconnecting the call when there is no sound for the period of time.

6.8 POTS Setting

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



7 Feature Configuration

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

7.1 Call Park

During a call, the callee may want to continue the conversation using another phone. The call park feature enables so by letting the callee transfer the call to the call park pilot number. IPPBX 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 IPPBX to resume conversation with the caller. If the callee does not call the given park line number to retrieve his call before timeout, IPPBX 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 7.1**.
2. Click **APPLY**.

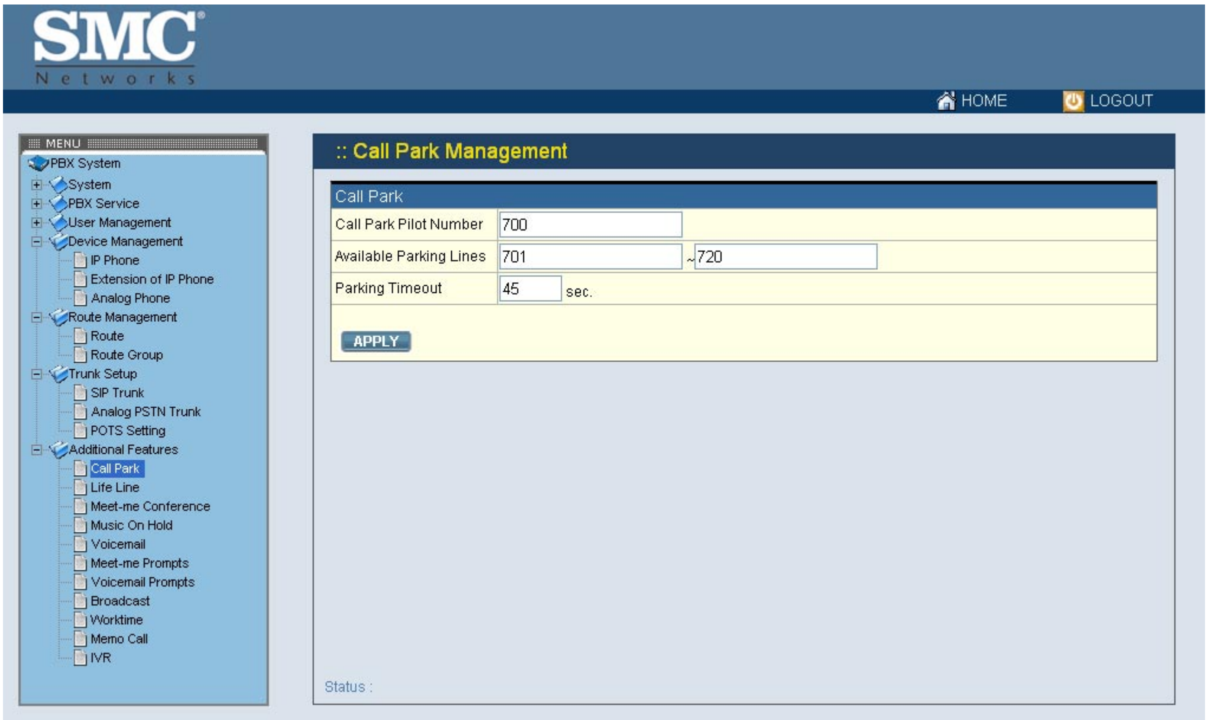


Table 7.1 Call Park Configuration Settings

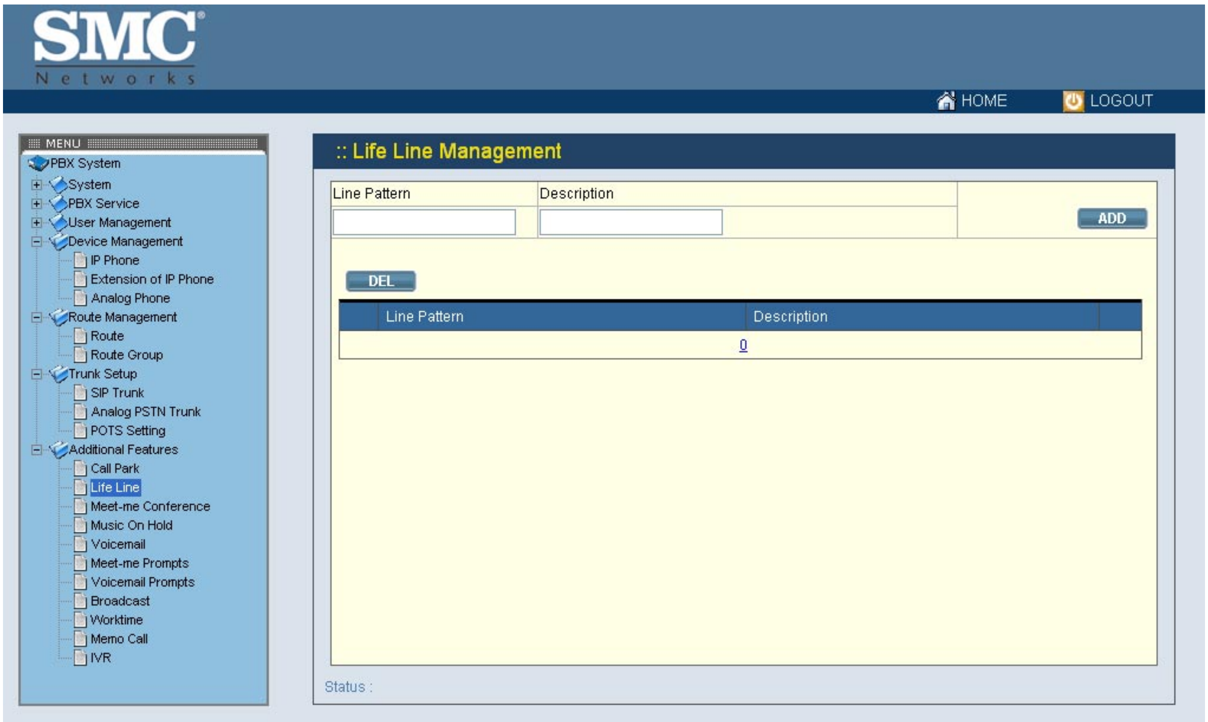
Field	Description
Call Park Pilot Number	A unique extension number for call parking, e.g. 700.
Available Parking Lines	An extension pool for call parking, e.g. 701-720 forms a

	20-line pool available for system to park calls.
Parking Timeout	Timeout waiting for picking up the parked call

7.2 Life Line

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

Select **Feature** → **Life line** to configure life-line feature.



7.2.1 Add a Life Line Pattern

1. Enter settings shown in **Table 7.2**.
2. Click **ADD** to see the newly added pattern in the **Line Pattern**.

7.2.2 Edit a Life Line Pattern

1. Edit settings shown in a row.
2. Click **APPLY** at the end of the row to update the information.

7.2.3 Delete a Life Line Pattern

1. Select a **Line pattern**.
2. Click **DEL** to remove the pattern from the **Line Pattern**.

Table 7.2 Life line Configuration Settings

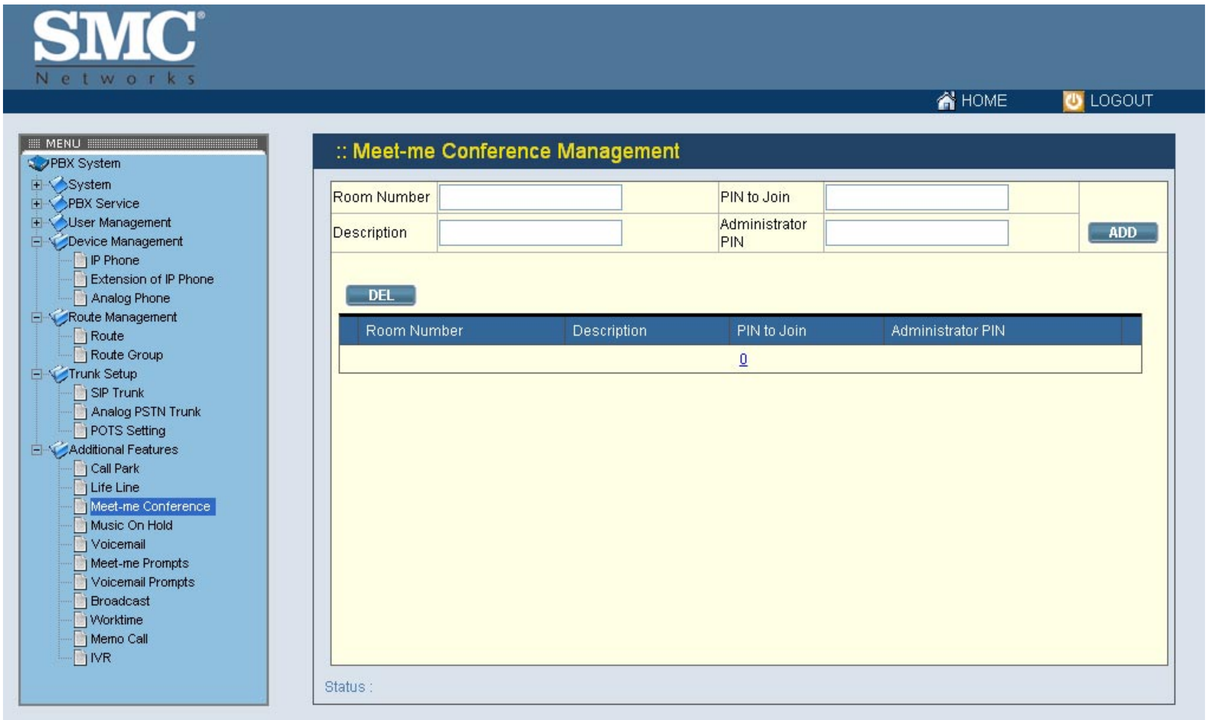
Field	Description
Line Pattern	Pattern for emergency numbers. **Note:** This is the pattern after digit stripping. For example, configure 911 here even if users dial 9911 to reach the 911 service over PSTN when the PSTN trunk has an outbound dialplan of “9.”.
Description	Arbitrary description information.

7.3 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 IPPBX. IPPBX 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.



7.3.1 Add a Meet-me Conference

1. Enter settings shown in **Table 7.3**.

2. Click **ADD** to add a new conference room.

The newly added room should display in the **Room Number**.

7.3.2 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.

7.3.3 Delete a Meet-me Conference

1. Select a room number.
2. Click **DEL** to remove the conference room from the **Room Number**.

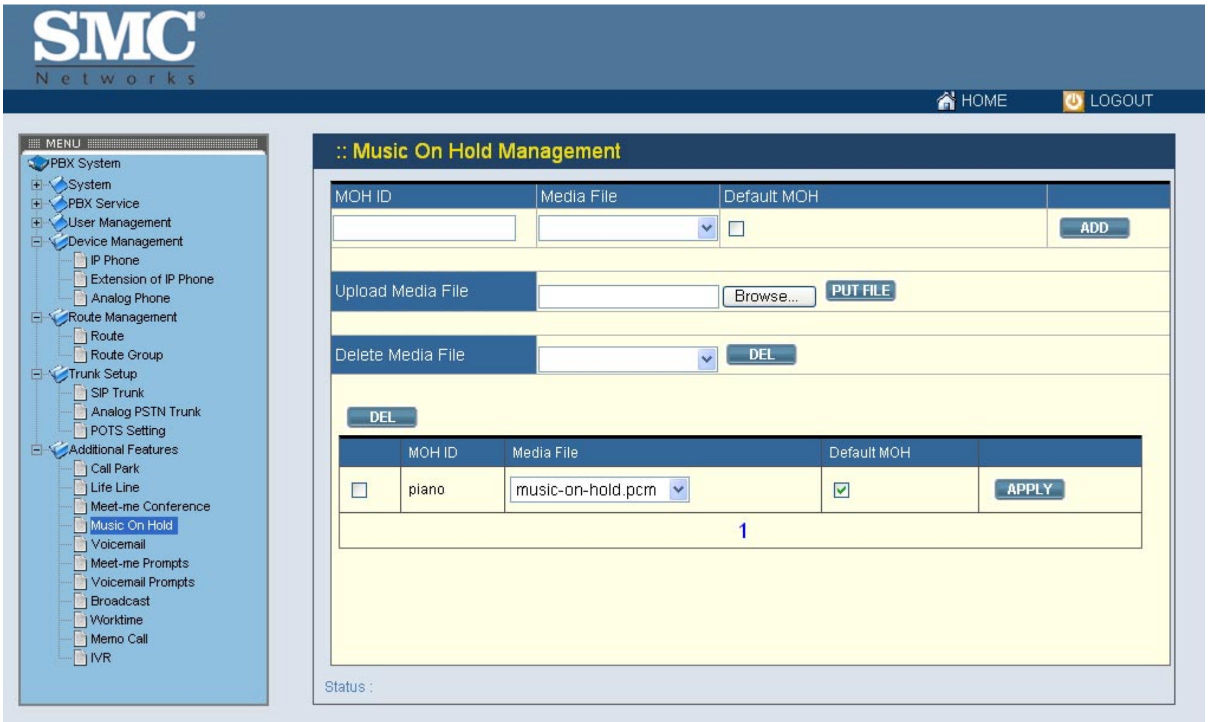
Table 7.3 Meet-me Conference Configuration Settings

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.

7.4 Music on Hold

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

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



7.4.1 Add a MOH File

1. Enter settings shown in **Table 7.4**.
2. Click **ADD** to see the newly added file in the **MOH ID**.

7.4.2 Edit a MOH File

1. Edit settings shown as a table at the bottom of the page.
2. Click **APPLY** in the row.

7.4.3 Delete a MOH File

1. Select a MOH ID.
2. Click **DEL** at the top-left of the table to remove the MOH file from the **MOH ID**.

Table 7.4 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.

7.5 Voicemail

IPPBX 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, IPPBX 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 7.5**.
2. Click **APPLY**.

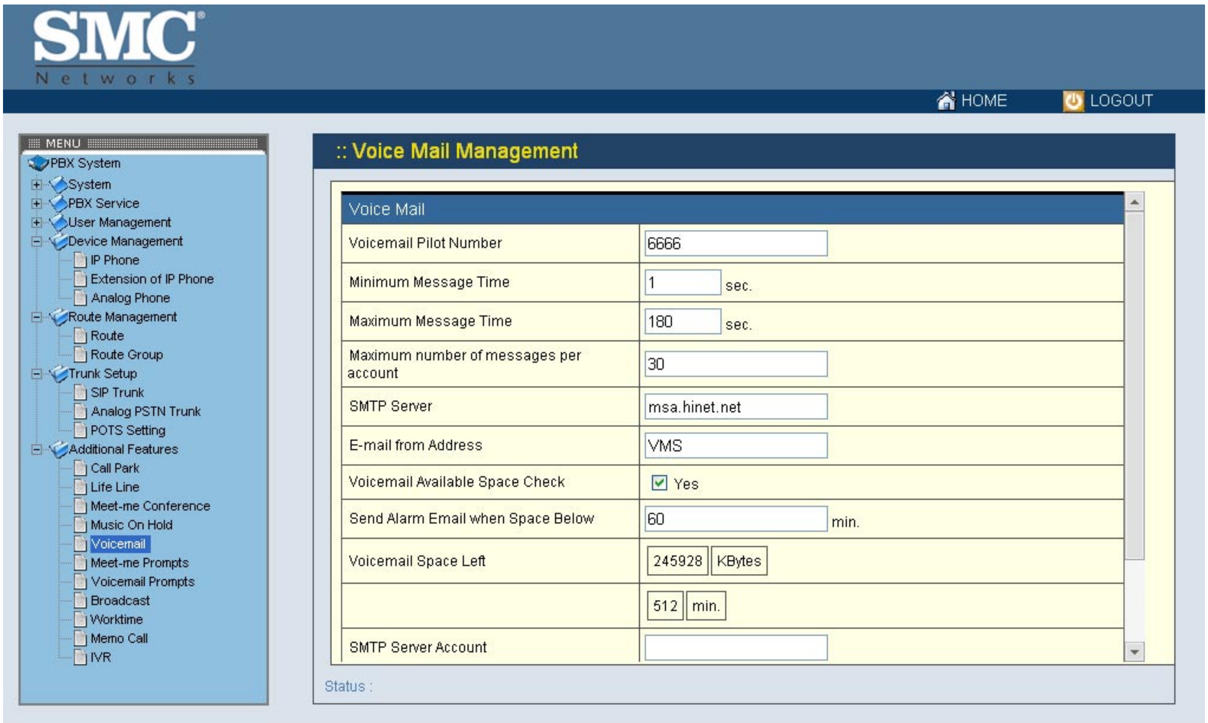


Table 7.5 Voice Mail Configuration Settings

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 IPPBX 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.

7.6 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. Click **Browse** to find a corresponding recording in the local storage.

3. Click **PUT FILE** to complete the replacement.

4. To reset a prompt back to default, click a language and a prompt in the corresponding lists, and then click **USE DEFAULT**.

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

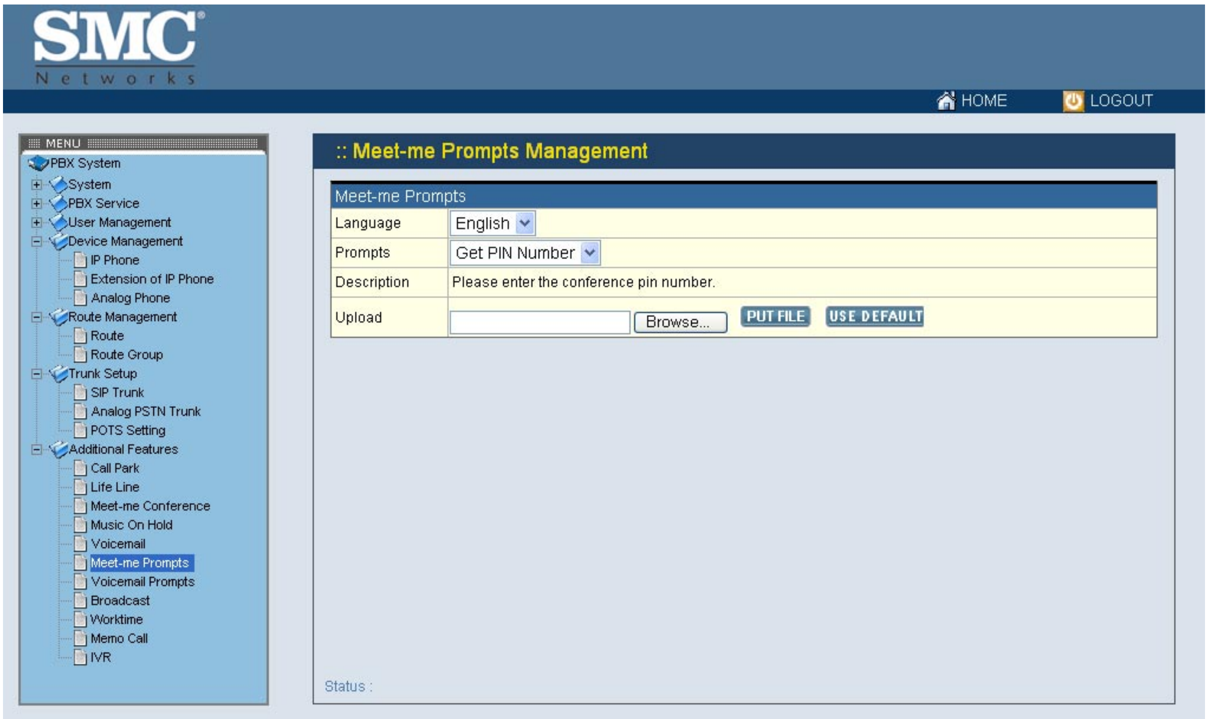


Table 7.6 Replaceable Meet-me Prompts

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

7.7 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, click a language and a prompt in the corresponding lists, and then click **USE DEFAULT**.

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

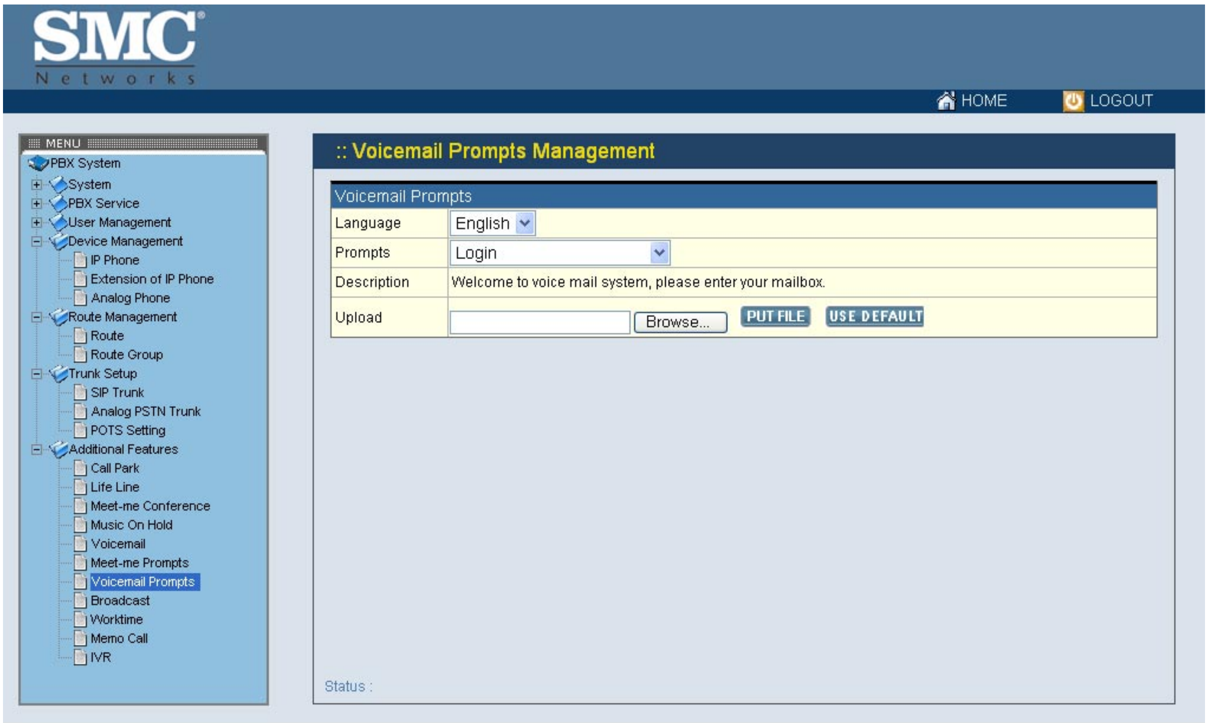


Table 7.7 Replaceable Voicemail System Prompts

Prompt	Description
Login	Welcome to voice mail system, please enter your mailbox.
Password	Password.
Incorrect Mailbox	Login incorrect, mailbox?
Good-bye	Good-bye.
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.
Extension	Extension.
Unavailable	It's not available.
Busy	It's on the phone.

7.8 Broadcast

A User can arrange an event at the exact time in IPPBX. IPPBX will inform all users that set in the **Callee Extensions** list by ringing their extensions. For example, one arranges a meeting and wants to remind all attendants, he/she may enter settings. When the time set in **Date/Time** is up, IPPBX will call to the extensions, and then executes the **Action** to each of the calls. Select **Feature** --> **Broadcast** to configure Broadcast feature.

7.8.1 Add a Broadcast

1. Click the **Add New** tab.
2. Enter settings shown in **Table 7.8**.
3. Click **ADD** at the bottom of the page to see the newly added broadcast event in the **Events**.

7.8.2 Edit a Broadcast

1. Click the **Management** tab.
2. Click an **Event**.
3. Edit settings shown in **Table 7.8**.
4. Click **UPDATE** to change the information.

7.8.3 Delete a Broadcast

1. Click the **Management** tab.
2. Select an **Event**.
3. Click **DEL** to remove the broadcast event from the **Events**.

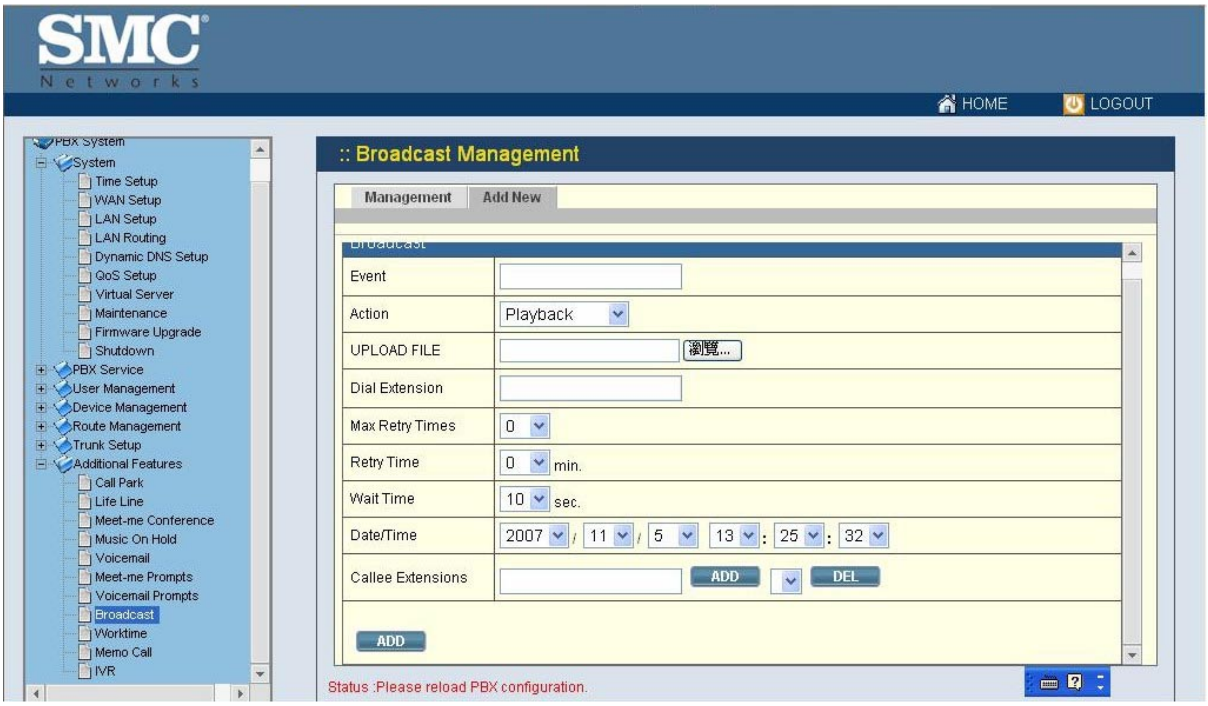


Table 7.8 Broadcast Configuration Settings

Field	**Description**
Event	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Action	Select one of the three actions to execute when the Date/Time is up. **Playback**: Play the uploaded WAV file in the **Upload File** box to the callee extensions. **Dial**: IPPBX will ring the party set in the **Callee Extensions** box first, and then call back to the one in the **Dial Extension** field to establish a conversation. It is suggested to set one number only in the Callee Extensions unless the extension number in the Dial Extension field is multi-line. **MusicOnHold**: Play default music to extensions in the **Callee Extensions** box.
Upload File	Upload a *.wav file, if **Playback** is selected in the **Action** list. ☞ The recording format must be 8000 Hz, 16 bit, Windows PCM WAV file.
Dial Extension	Select an extension to call back if **Dial** is selected in the **Action**.
Max Retry Times	Maximum redial times if callees did not answer.

Retry Time	A period of time in minutes between two retrying.
Wait Time	Enter timeout in seconds when ringing a callee.
Date/Time	Select a Date/Time to trigger this broadcast event.
Callee Extensions	Intended extensions to be called at the Date/Time.

7.9 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.

7.9.1 Add a Worktime

1. Click the **Add New** tab.
2. Enter settings shown in **Error! Reference source not found.**.
3. Click **ADD** at the bottom of the page.

The newly added worktime should display in the **Group ID**.

7.9.2 Edit a Worktime

1. Click the **Management** tab.
2. Click a **Group ID**.
3. Edit settings shown in **Error! Reference source not found.**
4. Click **UPDATE** to change the information.

7.9.3 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**.

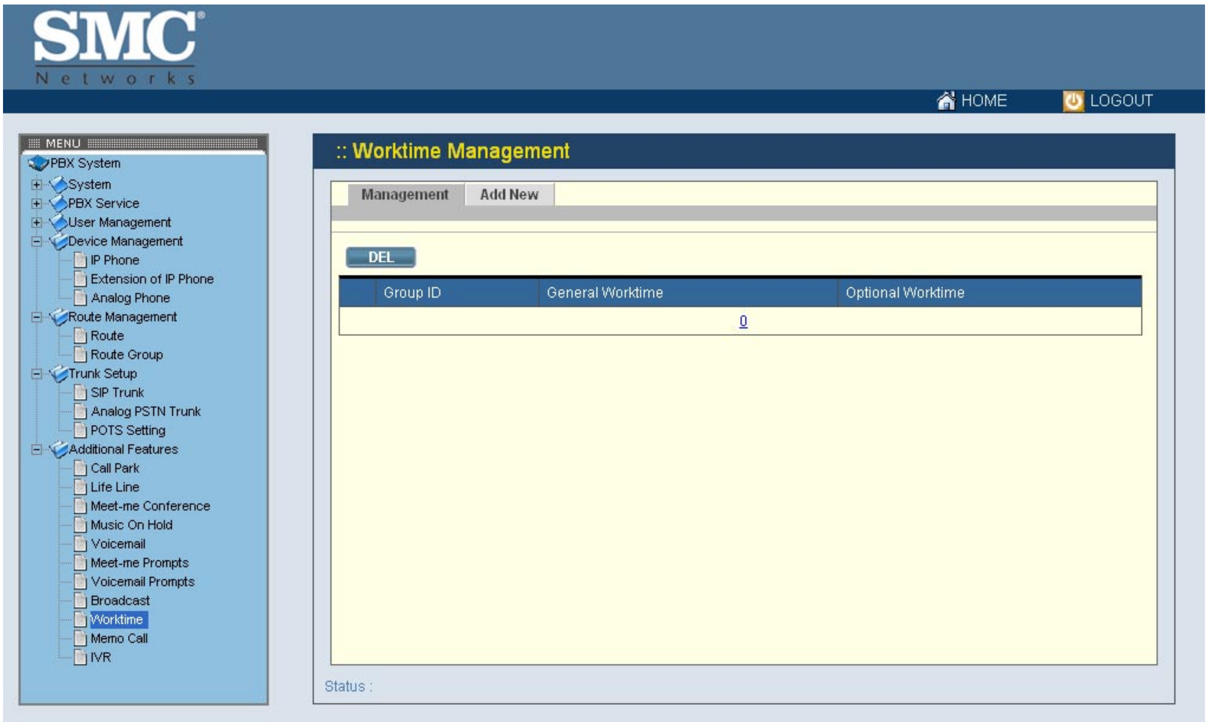


Table 7.9 Worktime Configuration Settings

Field	Description
Group ID	A unique ID containing numbers only.
General Worktime	The work time from Monday to Friday.
Optional Worktime	Special holidays or work day. User can set date and its work time, or set it to a whole-day holiday.

7.10 Memo Call

A user can set a memo at a specific time, e.g. a morning call, in IPPBX to inform the user or another user set in the **Extension** list by call at the time. When the user picks up the phone, IPPBX plays “voice file” or “music”. The user can also set this memo as a daily routine so that IPPBX informs the user at the exact time every day. Select **Feature** --> **Memo Call** to configure memo calls.

7.10.1 Add a Memo Call

1. Click the **Add New** tab.
2. Enter settings shown in **Table 7.10**.
3. Click **ADD** at the bottom of the page to see the newly added memo call in the **Memo**.

7.10.2 Edit a Memo Call

1. Click the **Management** tab.
2. Click a **Memo**.
3. Edit settings shown in **Table 7.10**.
4. Click **UPDATE** to update the information.

7.10.3 Delete a Memo Call

1. Click the **Management** tab.
2. Select a **Memo**.
3. Click **DEL** to remove the memo call from the **Memo**.

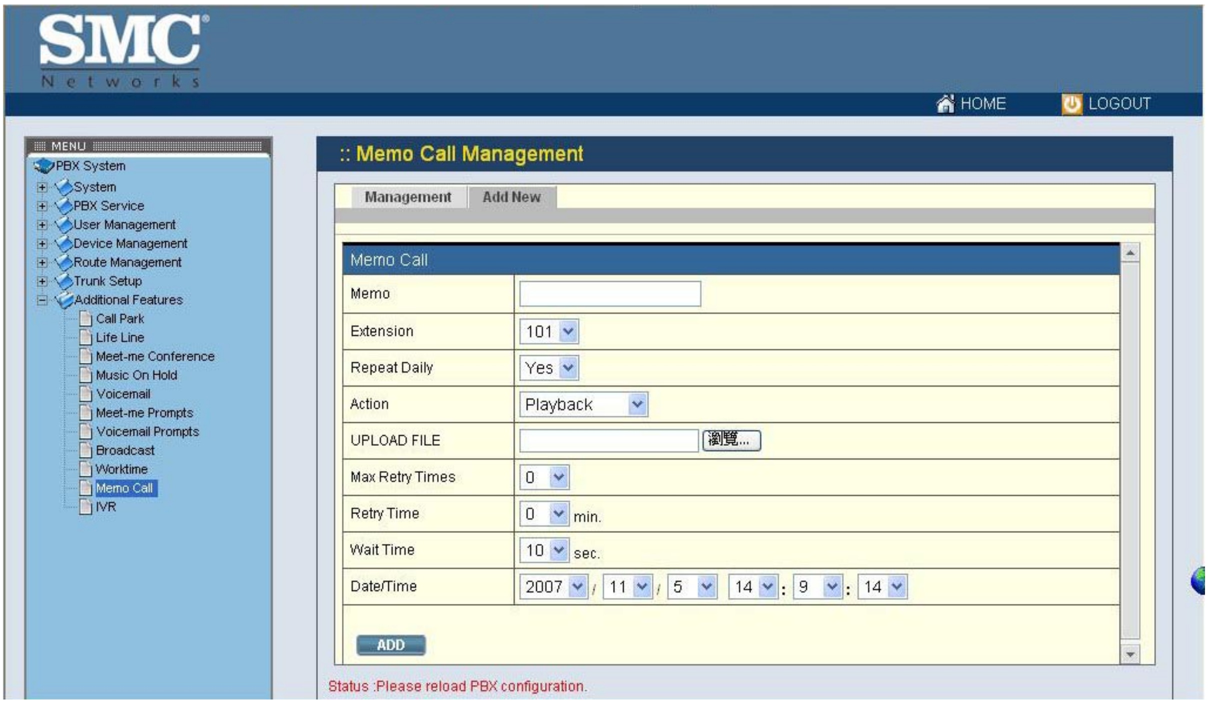


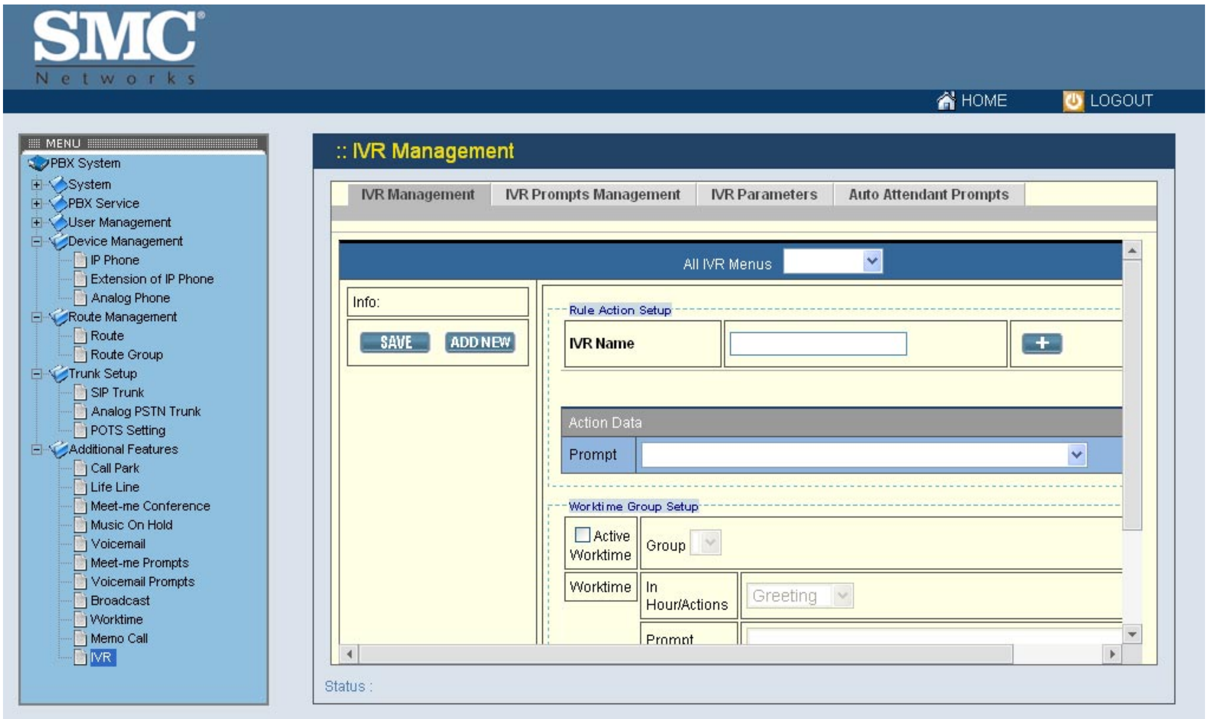
Table 7.10 Memo Configuration Settings

Field	Description
Memo	A unique ID containing alphabets, numbers, and underscore only without spaces.
Extension	Click an extension to call.
Repeat Daily	Select **Yes** to enable looping playback every day at the exact time.
Action	Select one out of the two available actions for the memo. **Playback**: Play the uploaded WAV file in the **Upload File** box to the extension(s). **MusicOnHold**: Play default music to the extension(s).

Upload File	Upload a *.wav file, if **Playback** is selected in the **Action** list. ☞ The recording format must be 8000 Hz, 16 bit, Windows PCM WAV file.
Max Retry Times	Maximum redial times if callees did not answer.
Retry Time	A period of time in minutes between two retries.
Wait Time	Enter timeout in seconds when ringing a callee.
Date/Time	Select a Date/Time to trigger this memo call.

7.11 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. IPPBX 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.



7.11.1 Add a new IVR Menu

1. Click **ADD NEW**, enter a name of an IVR menu in **IVR Name**, and click a file in the **Prompt** list.

2. Click  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. If Worktime setting is not required, click **Cancel** in the pop-up window.

3. Enter settings shown in **Table 7.11**.

4. Click **SAVE** to add the new IVR menu.

5. Click the IVR in the **All IVR Menus** list to see it as a tree view in **Info**.

6. 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 *Thank you* from **Prompt** list in **Action Data** block.
- Click the  next to the **IVR Name** box.
- Click **Cancel** in the pop-up window to confirm the Worktime setting is not required.
- Click **SAVE**.
- Now, *Basic_AA* should be available in the **IVR List** of Trunk pages.

7.11.2 Edit an IVR Menu

1. Click an IVR name in the **All IVR Menus** list.

2. Click **MODIFY** if you want to change the prompt and worktime settings for the first level of IVR.

3. Select a prompt and modify the worktime settings.

4. Click **UPDATE** to change the promprt and worktime settings.

5. For other settings, edit settings shown in **Table 7.11**.

6. Click **SAVE** to update the changes.

7.11.3 Delete an IVR Menu

1. Click an IVR name in the **All IVR Menus** list.

2. Click **DEL** to delete the IVR menu.

Table 7.11.1 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, and displays information for items in the tree view.
IVR Name	Specify the name of the IVR.
Rule	Click a number(0-9), *, or # in the **Key** list and one of the following actions in the **Action** list to associate an action with a key.

	Click **ext** and select a usergroup in the **Group** list in **Action Data** to allow dialing any extension in this usergroup. Click **timeout** and select actions except **Next Layer** and **Select Language** in the **Action** list to assign an action after the user response time is up. Click to add the rule in the **Node** list. ☞ If select **ext**, and the caller dialed a wrong extension, the system prompts an error message followed by a beep to request reentering a correct one. Any entered digit before the beep will not take effect.	
	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.
	Return	To go back to the previous layer.
Node	Information of the configured keys and actions. Click a node and to delete the node and its underlying structure.	
Child Rule	If a **Next Layer** is selected, Child Rule sets the key-action associations with the next-layer menu.	
Action Data	Specify applicable parameter(s) for an action.	
	Prompt	Select a *.wav recording file that you add from the **IVR Prompts Management** tab, or select one of the default voice file.
	Group	Select a usergroup.
	Language	Select a language of the IVR.
	Extension	Enter an extension number to be transferred to.
Active Worktime	Select to set work time for the IVR.	
Group	Select a work time group set in **Feature** --> **Worktime**.	
In-Hour Actions	Select one action during business hours.	
	Greeting	To play the selected prompt. The phone directly hang up after playing the prompt.
	Announce	To play the selected prompt. The caller can press keypad to enter numbers during the

		prompt is played.
	Call To	To transfer to an extension
	No Action	Play the prompt in **Action Data**.
Prompt	Select a prompt if **Greeting** or **Announce** is selected in the **In-Hour Actions** list.	
Extension	Enter an extension number if **Call To** is selected in the **In-Hour Actions** list.	
Off-Hour Actions	Select one action during the off hours.	
	Greeting	To play the selected prompt. The phone directly hangs up after playing the prompt.
	Announce	To play the selected prompt. The caller can press keypad to enter numbers during the prompt is played.
	Call To	To transfer to an extension
	No Action	Play the prompt in **Action Data**.
Prompt	Select a prompt if **Greeting** or **Announce** is selected in the **Off-Hour Actions** list.	
Extension	Enter an extension number if **Call To** is selected in the **Off-Hour Actions** list.	

7.11.4 IVR Prompts Management

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

7.11.4.1 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.

7.11.4.2 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.

7.11.5 IVR Parameters

IVR Parameters page offers the administrator to set to options, **Digit Input Timeout** and **User Response Timeout**. Select **Feature** --> **IVR**, and then click **IVR Parameters** tab to see the configurations.

Table 7.11.5 IVR Parameters Configuration

Field	Description
Digit input timeout	Enter timeout for digit collection, e.g. 5 sec.
User response time out	Enter timeout for caller response, e.g. 15 sec.

7.11.6 Auto Attendant Prompts

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

1. Click a language and a prompt in the corresponding lists.

2. Find the new 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. The recording format must be 8000 Hz, 16 bit, Windows PCM WAV file.

Table 7.111.6 Replaceable Auto Attendant Prompts

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

8 Example Provisioning

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

8.1 Internal Extension Configuration

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

- Configure a usergroup named **ALL** (refer to **Error! Reference source not found.**).
- Configure users and assign every user to usergroup **ALL** (refer to **Error! Reference source not found.**).
- Configure devices and an extension for each device (refer to **Error! Reference source not found.**).
- Assign each extension to a corresponding user (refer to **Error! Reference source not found.**).
- Configure each client phone with respect to the extension number and password in its IPPBX extension configuration accordingly.
- Reload the IPPBX service (refer to **Error! Reference source not found.**).

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

8.2 Case Ⅰ: Single Site Configuration

This case describes the typical settings of a single-site configuration; say Company A. Assume Company A has a DSL connection for Internet access and 2 PSTN subscriber lines as shown in **Figure 8-1**. The provisioning tasks include:

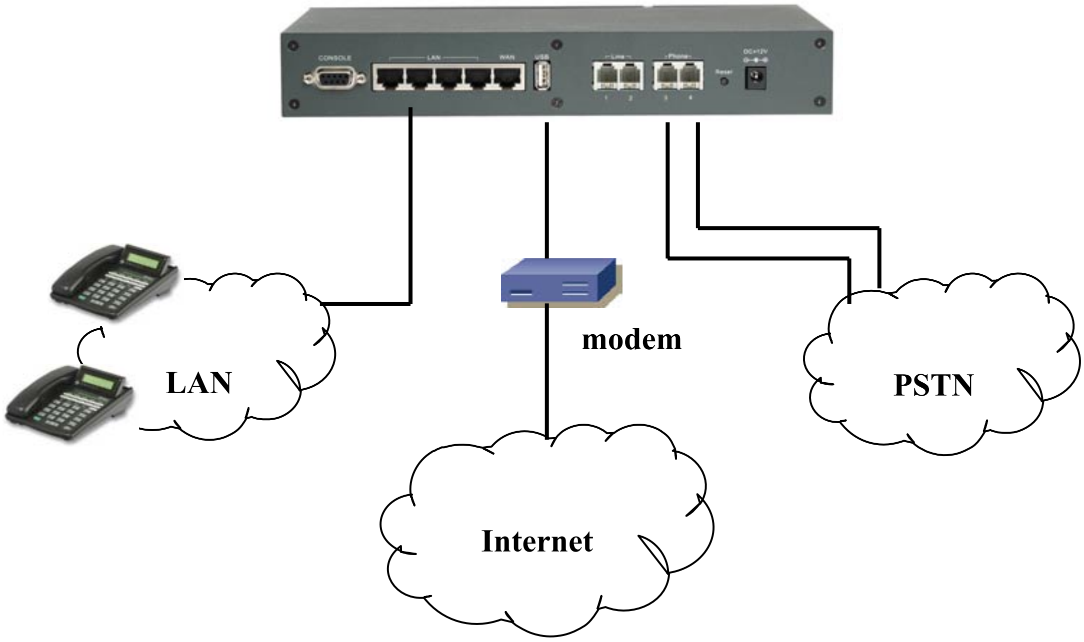


Figure 8-1

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

Configuration steps:

1. Create usergroups named **staff**, **utility**, and **ext-all**.

2. Add **staff** and **utility** in the Reachable User Groups of **ext-all**.

3. Create a user account for each staff and assign it to usergroup **staff**.

4. Create an additional user account named **public** and assign it to usergroup **utility**.

5. Create a device for each physical phone and designate an extension.

6. Assign extensions of staff phones to corresponding users.

7. Assign all extensions of utility phones to share the same user **public**.

8. Create a route, named **pstn** with pattern "**9.**" number of digits stripped "**1**", and no prefix.

9. Create a routegroup named **pstn-out**, and add route **pstn** only.

10. Create a PSTN trunk with ID “1”, port “1-2”, choose pstn-out as outbound routegroup, do not select any DID of extension, and select **ext-all** as the usergroup of privilege.

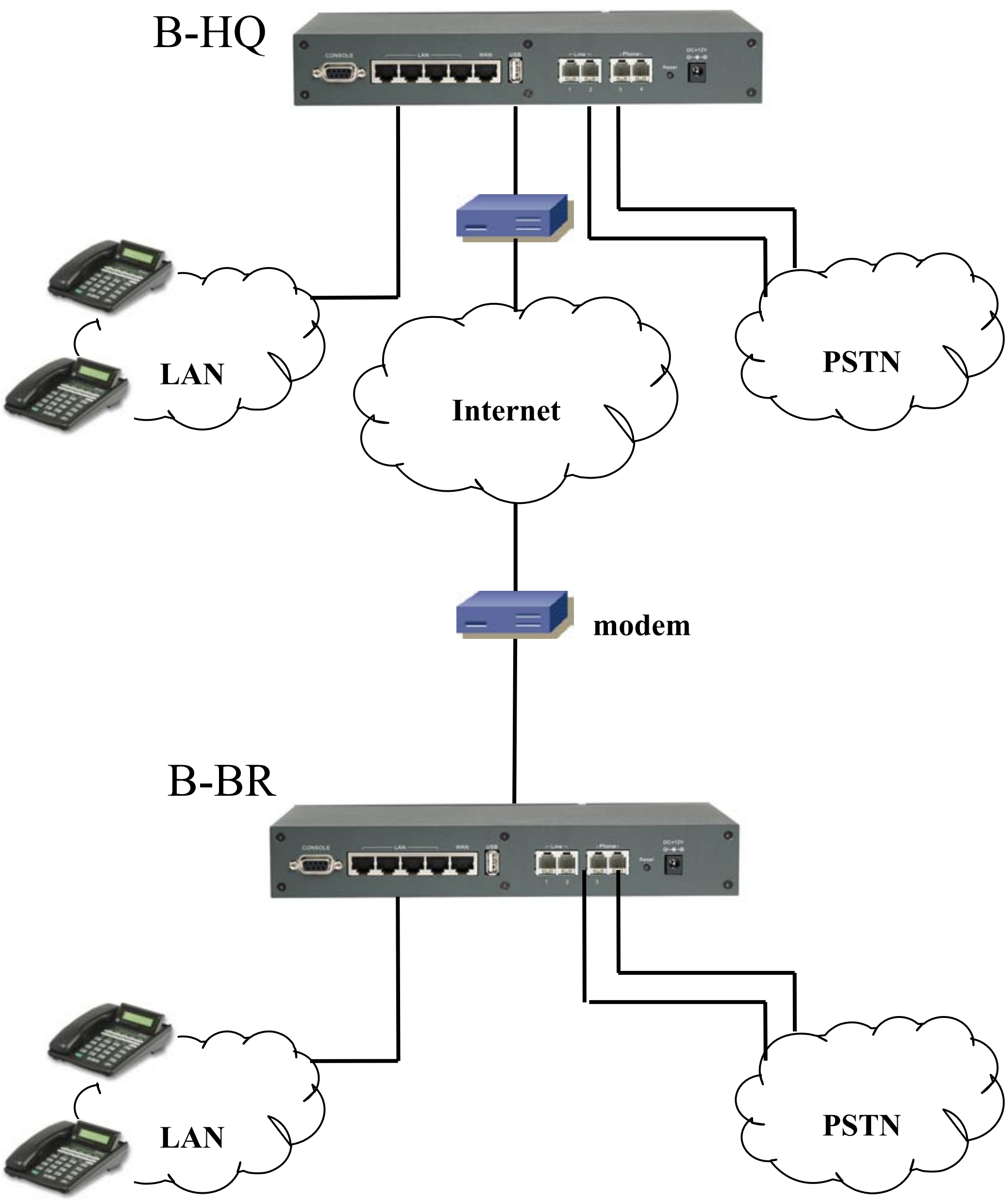
11. Return to usergroup configuration. For usergroup **staff**, click **pstn1** in Associated Trunks, and select **utility** in Reachable User Group; while for usergroup **utility**, only select **staff** in Reachable User Group.

12. Reload the IPPBX Service.

8.3 Case Ⅱ: Two sites Configuration

This case describes the typical settings of a two-site configuration; say Company B headquarters B-HQ and its branch B-BR located in another country. Assume each site has a DSL connection for Internet access. B-HQ has 4 PSTN subscriber lines and B-BR has 2 lines as shown in **Figure 8-2**. The provisioning tasks include:

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



◆ Configuration steps in B-HQ:

1. Create usergroups named **staff**, **utility**, and **ext-all**.

2. Add **staff** and **utility** in the Reachable User Groups of **ext-all**.

3. Create a user account for each staff and assign it to usergroup **staff**.

4. Create an additional user account named **public** and assign it to usergroup **utility**.

5. Create a device for each physical phone and designate an extension.

6. Assign extensions of staff phones to corresponding users.

7. Assign all extensions of utility phones to share the same user, **public**.

8. Create a route, **pstn**, with pattern “**9Z.**” with number of digits stripped “**1**”, no prefix.

9. Create a route, **pstn-br**, with pattern “**90118621.**” with number of digits stripped “**8**”, prefix “**9**”.

10. Create a route, **ext-br**, with pattern “**2XX**” and number of digits stripped “**0**”, no prefix.

11. Create a routegroup, **pstn-out**, containing route **pstn** only.

12. Create a routegroup, **to-br**, containing routes **pstn-br** and **ext-br**.

13. Create a PSTN trunk with ID “**1**”, port “**1-4**”, choose **pstn-out** as outbound routegroup, do not select any DID of extension, and select **ext-all** as the usergroup of privilege.

14. Create a dynamic peer SIP trunk with ID **100**; password **hq-secret**; choose **to-br** as outbound routegroup, do not select DID, and select **staff** as the usergroup of privilege.

15. Return to usergroup configuration. For usergroup **staff**, select **100**, **pstnl** in Associated Trunks, and select **utility** in Reachable User Group; while for usergroup **utility**, only select **staff** in Reachable User Group.

16. Reload the IPPBX Service.

◆ Configuration steps in B-BR:

1. Create usergroups named **staff**, **utility**, and **ext-all**.

2. Add **staff** and **utility** in the Reachable User Groups of **ext-all**.

3. Create a user account for each staff and assign it to usergroup **staff**.

4. Create an additional user account named **public** and assign it to usergroup **utility**.

5. Create a device for each physical phone and designate an extension.

6. Assign extensions of staff phones to corresponding users.

7. Assign all extensions of utility phones to share the same user, **public**.

8. Create a route, **pstn**, with pattern “**9Z.**” with number of digits stripped “**1**”, no prefix.

9. Create a route, **pstn-hq**, with pattern “**9001408.**” with number of digits stripped “**7**”, prefix “**9**”.

10. Create a route, **ext-hq**, with pattern “**1XX**” and number of digits stripped “**0**”, no prefix.

11. Create a routegroup, **pstn-out**, containing route **pstn** only.

12. Create a routegroup, **to-hq**, containing routes **pstn-hq** and **ext-hq**.

13. Create a PSTN trunk with ID **1**, port **1-2**, choose **pstn-out** as outbound routegroup, do not select any DID of extension, and select **ext-all** as the usergroup of privilege.

14. Create a SIP trunk with ID **100** pointing to **64.1.0.1** port **5060**; password **hq-secret**; choose **to-hq** as outbound routegroup, do not select DID, and select **staff** as the usergroup of

privilege.

15. Return to usergroup configuration. For usergroup **staff**, select **100**, **pstnl** in Associated Trunks and select **utility** in Reachable User Group; while for usergroup **utility**, only select **staff** in Reachable User Group.

16. Reload the IPPBX Service.

9 Appendices

9.1 Keypad Default Settings for IPPBX

IPPBX has some default keypad settings for general users to directly access functions via the keys on a phone.

Table 91 Keypad Default Settings for IPPBX

Keypad	Description
*8	If extensions are in the same pickup group[23], any extension in the pickup group can press *8 to pick up the call in ringing state**.**
*# + extension	Press *# plus an extension number to transfer the call to the extension. **Note:** When the caller from trunks press *# to transfer the call, the third party will see the caller ID of caller.
*# + Call Park Pilot Number	Press *# plus Call Park Pilot Number to transfer the call to Call Park[24].

9.2 Manage with CLI Commands

9.2.1 Instruction

The IPPBX Console Interface (CLI) administration guidelines describe how to manage some basic functions of the IPPBX using console interface. The console interface is set for two levels, general user and system administrator. General users can see general information once login to the interface. System administrator requires entering password of the admin to use more functions such as add, edit and delete user, usergroup, and import or export data.

9.2.2 Console Interface

By connecting to console port, general users can see some basic information. System administrator can enter some commands for adding, editing, deleting debugging.

9.2.2.1 Connection

1. Connect the console port with a computer using RS232.
2. Open a terminal emulator such as HyperTerminal on the computer.
3. Configure the serial port setting as follows:
    - Baud Rate: 115200

[23] Refer to **Table 6.3.** for more detail of pickup group.
[24] Refer to **Error! Reference source not found.** for more detail of Call Park.

- Data: 8 bit
- Parity: none
- Stop: 1 bit
- Flow Control: none

4. Click **OK** to connect.

9.2.2.2 Login

Type in the login username and password (both are ***estcli***).

9.2.2.3 Basic Commands

This section describes basic command lines for general users.

Help

Display the help message command available for the level.

CLI Command: help / ?

Quit / Logout

Enter the command to Logout CLI.

CLI Command: quit / logout

Exit

Exit the current level and go back to the previous level. If you are in the basic level, the console interface will logout. If you are in admin level, enter this command will disable and back to basic level.

CLI Command: exit

History

Displays the commands you entered before. You can also press ↑ ↓ on the keypad to show the used commands.

CLI Command: history

Enable

Enter the admin level. The system will request for a password. The password is the same as the administrator's password for web interface. Admin level allows to enter Admin and PBX Commands.

CLI Command: enable

Note: Make sure to log out the administrator in the web interface, or the command will be rejected.

Show

Display the system and network interface information.

System Uptime

Display the running time of the IPPBX system.

CLI Command: show system uptime

System Version

Display the PBX firmware OS AP and CLI version.

CLI Command: show system ver

System Memory Space

Display the PBX system memory space and usage information.

CLI Command: show system free

System Disk Space

Display the PBX system disk space information.

CLI Command: show system df

System Channel

Display the current active calls.

CLI Command: show system channel

WAN Interface

Display the WAN interface information.

CLI Command: show interface wan

LAN Interface

Display the LAN interface information.

CLI Command: show interface lan

All Interface

Displays all interface information

CLI Command: show interface all

Tool

Ping

Send ICMP echo request to determine if a network connection is established.

CLI Command: tool ping <*IP address*>

Trace Route

Display the path between two computers.

CLI Command: tool traceroute <*destination IP address*> <*source IP address*>

9.2.2.4 Admin Commands

This section is for the system administrator and requires password to use these commands. For commands in admin, the administrator can start, stop and restart system services, debug SIP, and shutdown or reboot the PBX system.

1. Enter **enable**, and then admin's password to log in as a system administrator.
2. Enter **admin terminal** to start using the admin commands.

Service

Manage with the PBX services such as Web, SNMP, TFTP and DHCP.

Web Service

Start or stop the web service. After start or stop the service, you need to enable the CLI system with password again to enter the admin level.

CLI Command: service web < *start / stop* >

Note: After starting or stopping the web service, the administrator needs to enable with password to login as administrator again.

Note: After stopping the web service, the web server will automatically restart within one minute.

SNMP Service

Start, stop or restart the SNMP service.

CLI Command: service snmp < *start / stop / restart*>

Note: This command is for system running only. The PBX configuration and the status in web interface will not be changed.

TFTP Service

Start, stop or restart the TFTP service.

CLI Command: service tftp < *start / stop / restart*>

Note: This command is for system running only. The PBX configuration and the status in web interface will not be changed.

DHCP Service

Start, stop or restart the DHCP service.

CLI Command: service dhcp < *start / stop / restart*>

Note: This command is for system running only. The PBX configuration and the status in web interface will not be changed.

PBX Service

Start, stop or restart the PBX service.

CLI Command: service pbx < *start / stop / restart*>

System

Use this command to shutdown or reboot the PBX system.

Shutdown System

Shut down the PBX system.

CLI Command: system shutdown

Reboot System

Reboot the PBX system.

CLI Command: system reboot

Firmware

Upgrade the PBX firmware version via TFTP or SSH server.

CLI Command: firmware upgrade <*tftp / scp*>://<*url*>:<*filename*>

Configuration

Backup, revert or restore the IPPBX configurations.

Export Configuration

Export all configurations to TFTP directory.

CLI Command: config export

Export PBX Configuration

Export only PBX configurations to TFTP directory.

CLI Command: config pbxexport

Revert PBX Configuration

Revert PBX configurations to factory default.

CLI Command: config revert

Restore PBX Configuration

Restore the PBX configurations from the remote TFTP or SSH server.

CLI Command: config import *<tftp / scp>*://*<url>*:*<filename>*

Debug

This section is for debugging SIP log and dump.

Start to Debug

Start to log SIP debug message.

CLI Command: debug star

Stop Debugging

Stop logging SIP debug messages, and save the log file as etxdbg-errlog.tar.gz in the TFTP directory.

CLI Command: debug stop

Note: Make sure to download the file in the web interface without log out the CLI system. Once log out the CLI system, the file will be deleted.

Note: If rename the debug log file, keep .tar.gz at the end.

9.2.2.5 PBX Commands

This section is for the system administrator and requires password to use these commands. For commands in pbx, the administrator can add, edit and delete user, usergroup, and export or import CSV (common split volume) files with the information of user, usergroup, IP phone, extensions, route, routegroup and SIP trunk.

1. Enter **enable**, and then admin's password to log in as a system administrator.
2. Enter **pbx terminal** to start using the admin commands.

User Configuration

The administrator can display, add, edit or delete users.

Show Users

Display one or all PBX users' information.

CLI Command: list user <*Login ID*> / list user

Add a User

Add one PBX user. Restart the PBX server in Admin Commands after adding the user.

CLI Command: add user -loginid <*Login ID*> -name <*Name*> -password <*Password*> -usergroup <*Group ID*> -description <*Description*> -attach <*yes / no*> -email <*e-mail address*>

Modify a User

The administrator can modify a PBX user. Restart the PBX server in Admin Commands after modifying the user.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify user –loginid <Login ID>	-name	"<Name>"
	-password	<Password>
	-usergroup	<Group ID>
	-description	"<Description>"

Delete a User

The administrator can delete a PBX user. Restart the PBX server in Admin Commands after deleting the user.

CLI Command: del user <*Login ID*>

Usergroup Configuration

The administrator can display, add, edit or delete usergroups.

Show Usergroups

Display one or all PBX usergroups' information.

CLI Command: list group <*Group ID*> / list group

Add a Usergroup

Add one PBX usergroup. Restart the PBX server in Admin Commands after adding the usergroup.

CLI Command: add group <*Group ID*>

Modify a Usergroup

The administrator can modify a PBX usergroup. Restart the PBX server in Admin Commands after modifying the usergroup.

CLI Command: (Use one of the following commands)

Command	Parameter 1	Parameter 2	Value
modify group <Group ID>	add	rules_tail	<Routegroup ID, Trunk, ID, Weight>
	add	group	<Group ID for Reachable Usergroup>
	add	feature	<vm / mm / parkedcalls>
	del	rules_tail	<Routegroup ID, Trunk, ID, Weight>
	del	group	<Group ID for Reachable Usergroup>
	del	feature	<vm / mm / parkedcalls>
	descript	----	<Description>

Delete a Usergroup

The administrator can delete a PBX usergroup. Restart the PBX server in Admin Commands after deleting the usergroup.

CLI Command: del group <*Group ID*>

IP Phone Configuration

The administrator can display, add, edit or delete IP phones.

Show IP Phones

Display one or all IP phones' information.

CLI Command: list ipphone <*Device ID*> / list ipphone

Add an IP Phone

Add an IP phone, and restart the PBX server in Admin Commands after adding the phone.

CLI Command: add ipphone <*Device ID*>

Modify an IP Phone

The administrator can modify an IP phone. Restart the PBX server in Admin Commands after modifying the phone.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify ipphone –ipphone *<Device ID>*	-url	<IP address>
	-venderprefix	<Vender Prefix>
	-mac	<MAC Address>
	-1code	<1st Codec>
	-1packettime	<1st Packet Time>
	-2code	<2nd Codec>
	-2packettime	<2nd Packet Time>
	-3code	<3rd Codec>
	-3packettime	<3rd Packet Time>
	-vad	<yes / no>
	-dtmf	<rfc2833 / inband / sipinfo>
	-1proxyip	<Proxy IP/FQDN of 1st Redundant Server>
	-1proxyport	<Proxy Port of 1st Redundant Server >
	-1regip	<Registrar IP/FQDN of 1st Redundant Server >
	-1regport	<Registrar Port of 1st Redundant Server >
	-1sipdom	<SIP Domain of 1st Redundant Server >
	-2proxyip	<Proxy IP/FQDN of 2nd Redundant Server>
	-2proxyport	<Proxy Port of 2nd Redundant Server >
	-2regip	<Registrar IP/FQDN of 2nd Redundant Server >
	-2regport	<Registrar Port of 2nd Redundant Server >
	-2sipdom	<SIP Domain of 2nd Redundant Server >

The administrator can also copy from one existing IP phone configuration to the other one. The command requires having the MAC address of destination IP phone.

CLI Command: modify ipphone –ipphone <Destination Device ID> -mac <Destination MAC Address> -clonefrom <Source Device ID>

Delete an IP Phone

The administrator can delete an IP phone. Restart the PBX server in Admin Commands after deleting the phone.

CLI Command: del ipphone <*Device ID*>

Extension of IP Phone Configuration

The administrator can display, add, edit or delete extensions of IP phones.

Show Extensions of IP Phones

Display one or all extensions' information.

CLI Command: list ipexten <*Extension Number*> / list ipexten

Add an Extension of IP Phone

Add an extension, and restart the PBX server in Admin Commands after adding the extension.

CLI Command: add ipexten –exten <*Extension Number*> –device <*Device ID*> –password <*Password*> –user <*User*>

Modify an Extension of IP Phone

The administrator can modify an extension. Restart the PBX server in Admin Commands after modifying the extension.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify ipexten –exten <*Extension Number*> -device <*Device ID*>	-password	<Password>
	-user	<User's Login ID>
	-group	<Pickup Group>
	-linetype	<wired / wireless>
	-lang	<en / tw / jp> **Note:** The above commands represent the following configurations. **en:** English. **tw:** Mandarin (Taiwan). **jp:** Japanese.
	-voicemail	<yes / no>
	-vmpin	<Voicemail Pin>

	-maxvmspace	<Max Voicemail Space in Kbytes>
	-unavtimeout	<Time in Second>
	-lanonly	<yes / no>
	-natdiable	<yes / no>
	-callalive	<yes / no>
	-regalive	<yes / no>
	-peertp	<INVITE / UPDATE / NO>
	-dtmf	<rfc2833 / inband / sipinfo>
	-callbolcking	<Phone Number> **Note:** If more than one phone number need to be added, use comma (,) to separate. Clear all phone numbers by typing *NULL*.
	-blockanon	<yes / no>
	-sipredirectiondeny	<yes / no>
	- allfwd	<Phone Number / voicemail / NULL>
	-timeoutfwd	<Extension Number>/<Time in Second> **Note:** If more than one entry need to be added, use comma (,) to separate. Clear all phone numbers by typing *NULL*.
	-fwdprompt	<yes / no>
	-busyfwd	<Phone Number>
	-selectcallfwd	<Source Phone Number>:Local/<DestinationPhone Number> **Note:** If more than one entry need to be added, use comma (,) to separate. Clear all phone numbers by typing *NULL*.

The administrator can also copy from one existing extension configuration to the other one.

CLI Command: modify ipexten –exten <Destination Extension Number> -device

<Destination Device ID> -clonefrom <Source Extension Number>, <Source Device ID>

Delete an Extension of IP Phone

The administrator can delete an extension of IP phone. Restart the PBX server in Admin Commands after deleting the phone.

CLI Command: del ipexten *<Extension Number>*

Analog Phone Configuration

The administrator can display, add, edit or delete analog phones.

Show Analog Phones

Display one or all analog phones' information.

CLI Command: list analogphone *<Extension Number>* / list analogphone

Add an Analog Phone

Add the information of an analog phone, and restart the PBX server in Admin Commands after adding the information.

CLI Command: add analogphone –port *<POTS Port>* –exten *<Extension Number>* –user *<User>* –group *<Usergroup>*

Modify an Analog Phone

The administrator can modify the information of an analog phone. Restart the PBX server in Admin Commands after modifying the information.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify analogphone –port <POTS Port>	-group	<Pickup Group>
	-unavtimeout	<Time in second>
	-user	<User>
	-voicemail	<yes / no>
	-vmpin	<Voicemail Pin>
	- maxvmspace	<Max Voicemail Space>
	-lang	<en / tw / jp> **Note:** The above commands represent the following configurations. **en:** English. **tw:** Mandarin (Taiwan).

		jp: Japanese.
	-t38enable	<auto / yes / no>
	-udptl	<UDPTL Redundancy Level>
	-ingain	<Input Gain>
	-outgain	<Output Gain>
	-callbolcking	<Phone Number> **Note:** If more than one phone number need to be added, use comma (,) to separate. Clear all phone numbers by typing NULL.
	-allfwd	<Extension Number / voicemail / NULL>
	-timeoutfwd	<Extension Number>/<Time in Second> **Note:** If more than one entry need to be added, use comma (,) to separate. Clear all phone numbers by typing NULL.
	-fwdprompt	<yes / no>
	-inusefwd	<Phone Number>
	-selectcallfwd	<Source Phone Number>:Local/<Destination Phone Number> **Note:** If more than one entry need to be added, use comma (,) to separate. Clear all phone numbers by typing NULL.
	-blockanon	<yes / no>

The administrator can also copy from one existing extension configuration to the other one.

CLI Command: modify analogphone –port *<POTS Port>* -clonefrom *<Source POTS Port>*

Delete an Analog Phone

The administrator can delete the information of an analog phone. Restart the PBX server in Admin Commands after deleting the information.

CLI Command: del analogphone <*POTS Port*>

Route Configuration

The administrator can display, add, edit or delete routes.

Show Routes

Display one or all PBX routes' information.

CLI Command: list route <*Route ID*> / list route

Add a Route

Add one route, and restart the PBX server in Admin Commands after adding the route.

CLI Command: add route -routeid <Route ID> -pattern <Destination Number Pattern> -stripped <Number of Stripped Digits> -prefix <Prefix> -description <Description>

Modify a Route

The administrator can modify a route. Restart the PBX server in Admin Commands after modifying the route.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify route –routeid <Route ID>	-pattern	<Destination Number Pattern>
	-stripped	<Number of Stripped Digits >
	-prefix	<Prefix>
	-description	"<Description>"

Delete a Route

The administrator can delete a route. Restart the PBX server in Admin Commands after deleting the route.

CLI Command: del route <*Route ID*>

Routegroup Configuration

The administrator can display, add, edit or delete routegroups.

Show Routegroups

Display one or all routegroup information.

CLI Command: list rg <*Group ID*> / list rg

Add a Routegroup

Add one routegroup, and restart the PBX server in Admin Commands after adding the routegroup.

CLI Command: add rg <*Group ID*>

Modify a Routegroup

The administrator can modify a routegroup. Restart the PBX server in Admin Commands after modifying the routegroup.

CLI Command: (Use one of the following commands)

Command	Parameter		Value
modify rg <Group ID>	addroute		<Route ID>
	deleteroute		----
	description	----	"<Description>"

Note: Do not add Route ID after deletroute command, and the CLI system will delete the very last route in Associated Routes.

Delete a Routegroup

The administrator can delete a routegroup. Restart the PBX server in Admin Commands after deleting the routegroup.

CLI Command: del rg <*Group ID*>

SIP Trunk Configuration

The administrator can display, add, edit or delete SIP trunk.

Show SIP Trunks

Display one or all SIP trunks' information.

CLI Command: list siptrunk <*Trunk Identifier*> / list siptrunk

Add a SIP Trunk

Add SIP trunk, and restart the PBX server in Admin Commands after adding the SIP trunk.

CLI Command: add siptrunk <*Trunk Identifier*>

Note: Some items have default values, and the administrator may need to modify those values.

Modify a SIP Trunk

The administrator can modify SIP trunk. Restart the PBX server in Admin Commands after modifying the SIP Trunk.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify siptrunk –trunkid <Trunk Identifier>	-description	<Description>
	-dynapeer	<yes / no>
	-name	<Auth. Name>
	-password	<Auth. Password>
	-proxy	<IP Address>
	-proxyport	<UDP Port>
	-registration	<yes / no>
	-registrar	<IP Address>
	-regport	<UDP Port>
	-did	<Extension Number / bynumber>
	-prefix	<DID Prefix>
	-stripping	<DID Stripping>
	-lang	<en / tw / jp> **Note:** The above commands represent the following configurations. **en:** English. **tw:** Mandarin (Taiwan). **jp:** Japanese.
	-ivr	<IVR List>
	-group	<Usergroup of Privilege>
	-domain	<SIP Domain>
	-useragent	<User-agent Content>
	-sensitive	<yes / no>
	-bwlimit	<yes / no>
	-dtmf	<rfc2833 / sipinfo / notsure>
	-peertp	<INVITE / UPDATE / NO>
	-enumenable	<yes / no>
	-clearbidings	<yes / no>
	-natdisable	<yes / no>
	-beforedelay	<time in second>

	-afterdelay	<time in second>
	-callalive	<yes / no>
	-regalive	<yes / no>
	-antisipblock	<yes / no>
	-gwtrunk	<yes / no>
	-calleridfrom	<From Caller ID>
	-payload	<Substitute Payload Type>
	-trunkrelay	<yes / no>
	-allowconcall	<number of allowed concurrent calls>

The administrator can also copy from one existing SIP trunk configuration to the other one.

CLI Command: modify siptrunk –trunk <Destination Trunk Identifier> -clonefrom <Source Trunk Identifier>

Delete a SIP Trunk

The administrator can delete SIP trunk. Restart the PBX server in Admin Commands after deleting the SIP trunk.

CLI Command: del siptrunk <*Trunk Identifier*>

Analog PSTN Trunk Configuration

The administrator can display, add, edit or delete analog PSTN trunk.

Show Analog PSTN Trunks

Display one or all analog PSTN trunks' information.

CLI Command: list analogtrunk <*Trunk Group*> / list analogtrunk

Add an Analog PSTN Trunk

Add an analog PSTN trunk, and restart the PBX server in Admin Commands after adding the trunk.

CLI Command: add analogtrunk -pstn <*Trunk Group*> -port <*Trunk Ports*> -type <*FXS / FXO*>

Modify an Analog PSTN Trunk

The administrator can modify any analog PSTN trunk. Restart the PBX server in Admin Commands after modifying the trunk.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify analogtrunk –pstn <Trunk Group>	-description	<Description>
	-seleport	<ANR / AR / DNR / DR> **Note:** The above commands represent the following configurations. **ANR:** ascending and not rotating. **AR:** ascending and rotating. **DNR:** descending and not rotating. **DR:** descending and rotating.
	-callerid	<yes / no>
	-billing	<yes / no>
	-did	<Extension Number / bynumber / bygroup / NULL> **Note:** bynumber and bygroup only available when setting the trunk type as FXS. Disable DID function by typing NULL.
	-didprefix	<DID Prefix>
	-didstripping	<DID Stripping>
	-lang	<en / tw / jp> **Note:** The above commands represent the following configurations. **en:** English. **tw:** Mandarin (Taiwan). **jp:** Japanese.
	-ivr	<IVR List>
	-group	<Usergroup of Privilege>
	-ingain	<Input Gain>

	-outgain	<Output Gain>
	-mindiscon	<Minimum Disconnection Tone>
	-beforedelay	<time in second>
	-afterdelay	<time in second>

The administrator can also copy from one existing analog PSTN trunk configuration to the other one.

CLI Command: modify analogtrunk –pstn <*Destination Trunk Group*> -clonefrom <*Source Trunk Group*>

Delete an Analog PSTN Trunk

The administrator can delete any analog PSTN trunk. Restart the PBX server in Admin Commands after deleting the trunk.

CLI Command: del analogtrunk <*Trunk Group*>

POTS Setting Configuration

The administrator can display, or edit POTS setting.

Show POTS Setting

Display all POTS status.

CLI Command: list pots

Modify POTS Setting

The administrator can modify POTS settings. Restart the PBX server in Admin Commands after modifying the POTS setting.

CLI Command: (Use one of the following commands)

Command	Parameter	Value
modify pots –portchannel <Port Number / Trunk Channel>	-mode	*<analog / isdn>*
	-compand	*<mulaw / alaw>*
	-impedance	*<Country>* **Note:** Available countries for this command are argentina, australia, austria, bahrain, belgium, brazil, bulgaria, canada, chile, china, columbia,

		croatia, cyprus, czech, denmark, ecuador, egypt, elsalvador, fcc, finland, france, germany, greece, guam, hongkong, hungary, iceland, india, indonesia, ireland, israel, italy, japan, jordan, kazakhstan, kuwait, latvia, lebanon, luxembourg, macao, malaysia, malta, mexico, morocco, netherlands, newzealand, nigeria, norway, oman, pakistan, peru, philippines, poland, portugal, romania, russia, saudiarabia, singapore, slovakia, slovenia, southafrica, southkorea, spain, sweden, switzerland, syria, taiwan, tbr21, thailand, uae, uk, usa, and yemen.

<u>Other</u>

<u>List</u>

Apart from listing users and usergroups, the administrator can also list other items such as route, routegroup, extensions, IP phone and SIP trunks.

CLI Command: list <*route / rg / ipextension / ipphone / siptrunk*>

9.3 Voicemail Box Menu Tree

This section displays the options for IPPBX voicemail box once users login.

|-1.New/old messages
| |-3.Advanced function
| | |-1.Send your reply
| | |-2.Call the person who send this message
| | | |-1.Call this number

| | | |-2.Enter a different number
| | | |-*.Cancel
| | |-3.Hear the message envelope
| | |-4.Place the outgoing call
| | |-8.Transfer message to another user
| | | |-Please press a number you want to call and then press #
| | | |-*.Cancel
| | |-*.Return to the main menu
| |-4.Play the previous message
| |-5.Repeat the current message
| |-6.Play the next message
| |-7.Delete this message
| |-8.Transfer this message to another user
| | |-1.Prepend message
| | |-2.Divorce the message without prepending
| | |-*.Return to the main menu
| |-9.Save this message
| | |-0.New messages
| | |-1.Old messages
| | |-2.Work messages
| | |-3.Family messages
| | |-4.Friends messages
| | |-#.Cancel
| |-*.Help
| |-#.Exit
|-2.Change folder
| |-0.New messages
| | |-(The same as Friends message menu.)
| |-1.Old messages
| | |-(The same as Friends message menu.)
| |-2.Work messages
| | |-(The same as Friends message menu.)
| |-3.Family messages
| | |-(The same as Friends message menu.)
| |-4.Friends messages
| | |-1.Friend (New/Old/Work/Family) messages
| | | |-3.Advanced function
| | | | |-1.Send your reply

```
|          |         |         |         |-2.Call the person who send this message
|          |         |         |         |         |-1.Call this number
|          |         |         |         |         |-2.Enter a different number
|          |         |         |         |         |-*.Cancel
|          |         |         |         |-3.Hear the message envelope
|          |         |         |         |-4.Place an outgoing call
|          |         |         |         |-*.Return to the main menu
|          |         |         |-4.Play the previous message
|          |         |         |-5.Repeat the current message
|          |         |         |-6.Play the next message
|          |         |         |-7.Delete this message
|          |         |         |-8.Transfer this message to another user
|          |         |         |         |-1.Prepend message
|          |         |         |         |-2.Divorce the message without resending
|          |         |         |         |-*.Return to main menu
|          |         |         |-9.Save this message
|          |         |         |         |-0.New message
|          |         |         |         |-1.Old message
|          |         |         |         |-2.Work message
|          |         |         |         |-3.Family message
|          |         |         |         |-4.Friend message
|          |         |         |         |-#.Cancel
|          |         |         |-*.Help
|          |         |         |-#.Exit
|          |         |-2.Change folder
|          |         |         |-0.New messages
|          |         |         |-1.Old messages
|          |         |         |-2.Work messages
|          |         |         |-3.Family messages
|          |         |         |-4.Friends messages
|          |         |         |-#.Cancel
|          |         |-3.Advanced function
|          |         |         |-4.Place an outgoing call
|          |         |         |-*.Return to main menu
|          |         |-0.Mail box function
|          |         |          |-1.Record your unavailable message
|          |         |          |           |-1.Accept the recording
|          |         |          |           |-2.Listen to it
```

```
|          |         |          |          |-3.Re-record it
|          |         |          |-2.Record your busy message
|          |         |          |          |-1.Accept the recording
|          |         |          |          |-2.Listen to it
|          |         |          |          |-3.Re-record it
|          |         |          |-3.Record your name
|          |         |          |          |-1.Accept the recording
|          |         |          |          |-2.Listen to it
|          |         |          |          |-3.Re-record it
|          |         |          |-4.Change your password
|          |         |          |-*.Return to main menu
|          |         |-*.Help
|          |         |-#.Exit
|          |-#.Cancel
|-3.Advanced function
|          |-4.Place outgoing call
|          |-*.Return the main menu
|-0.Mail box function
|          |-1.Record your unavailable message
|          |        |-1.Accept the recording
|          |        |-2.Listen to it
|          |        |-3.Re-record it
|          |-2.Record your busy message
|          |        |-1.Accept the recording
|          |        |-2.Listen to it
|          |        |-3.Re-record it
|          |-3.Record your name
|          |        |-1.Accept the recording
|          |        |-2.Listen to it
|          |        |-3.Re-record it
|          |-4.Change your password
|          |-*.Return to main menu
|-*.Help
|-#.Exit
```

TECHNICAL SUPPORT

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