



Internet Telephony PBX System

IPX-2000

User's Manual

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

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Revision

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1 Introduction

1.1 Overview

PLANET IPX-2000 series IP PBX system are designed and optimized for the enterprise, and SMB daily communications. The **IPX-2000** is the next generation voice communication platform for the small to medium enterprise. Designed as an open, scalable, and highly reliable telephony solution, the IPX-2000 series are able to accept 200 extension registrations, and effectively meeting scales from various enterprises. Designed to run on a variety of VoIP applications, the IPX-1800 and IPX-2000 provides centralized call control, auto-attendant, voice conferencing, PSTN, and IP-based communications.

The IPX-2000 integrates up to 8 FXO to become a feature-rich PBX system that supports seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints

The IPX-2000 system integrates telephony call processing, call control, voice mail, and a widely PBX application programming interface into a highly scalable architecture designed to support both traditional circuit-based and the Internet telephony service within a distributed enterprise communications network.

With IPX-2000 system, standard SIP phones can be easily integrated in your office; plus the auto-config feature, you may integrate our IP Phone series - VIP-254T/VIP-254T, and the ATA (analog telephone adapter) series - VIP-156/VIP-157 to build up the VoIP network deployment in minutes.

Allowing distributed IP technology to meet traditional voice services, with proactive management interface, the IPX-2000 system in the daily business processes, enterprises can make people more productive, more intelligent tasks, and more customer satisfaction.

1.2 Physical Interfaces

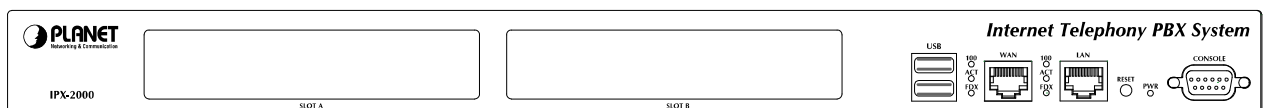


Figure 7. Front Panel of IPX-2000



Figure 8. Rear Panel of IPX-2000

Power cord	100-240 Volt, 50-60 Hz	
Telephony interface ports	IPX-2000	2 x slots integrates up 8 FXO
	FXO module	4 ports per daughter card, numbered from right to the left. The rightmost port at slot 0 is port 1 and the leftmost port at slot 1 (if installed) is port 8. FXO ports are to be connected to FXS jacks on wall or analog PBX using RJ-11 cables.
USB ports	1 external port with compliance to USB 1.1/2.0. Plug in a USB hard drive for voicemail backup from the internal one	
WAN	Connect to a broadband modem or a WAN router	
LAN	Connect to a LAN switch	

Table 1. Physical interfaces description of IP PBX


2 Managing with Web Interface

The factory default of LAN IP address is 192.168.1.1. Connect to LAN port and the configuration Web interface is at <https://192.168.1.1/>. Once connected, the browser will ask for accepting a certificate. Click **Yes** to see the home page. Type in the default administrator ID and password (both are *admin*) to log in for administration. The administrator password can be changed in the **User Management -> User**.



1. Click **admin** in the **Login ID**.
2. Change the password in **Password**.
3. Click **Apply** 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 Route & Trunk**, **Mass Extensions Adding** and **Customize Setup**. The first three icons can lead you step by step to configure some basic settings of IP PBX. Click the **Customize Setup** icon to see all the PBX configurations into detail.

Administrator can click  on the top-left side of the webpage to go back to the home page of IP PBX Web Interface.

3 Wizard Configuration

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

3.1 Add Account Wizard

In **Add User & Device** page, the administrator can setup usergroups, users and devices. You can follow the following steps to finish configuration. After finishing configuration, click  at the bottom of the homepage to take the configuration effect.

Step 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**.
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 a user.

Table 3-1 Add Usergroup Settings

Field	Description
Description	Arbitrary description information. Click SET to add/update

	the information.
Associated Trunks ¹	<p>Select routegroups and outbound trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time.</p> <p>Routegroup: display available routegroups.</p> <p>Trunk: Display available trunks.</p> <p>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.</p> <p>Weight: the weight of a trunk to be selected in a trunk balance group for an outgoing call.</p> <p>Click <input type="button" value="+"/> or <input type="button" value="-"/> to add or delete the associate trunks. After add all trunks, click APPLY.</p> <p> If there is not any appropriate SIP trunk and PSTN trunks to select, you may assign trunks at Setp 5: Assign Trunk in Add Route & Trunk wizard configuration after trunks are created in the previous step.</p>
Reachable User Groups	<p>Select a usergroup and click ADD that is reachable from this usergroup. By default, only users in the same usergroup can reach one another.</p> <p> If there is not any appropriate usergroup to select, come back later to revise this selection, once more usergroups have been created.</p>
Associated PBX Features ²	Select PBX features enabled to this usergroup.
Member List	Show the users 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.

¹ Please refer to 6.6, 6.7, and 6.8 for details.

² Please refer to 7 for details.


Setp 2: Add User

1. Enter settings shown in **Table 3-2**.
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 3-2 Add User Settings

Field	Description
Login ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 20 characters maximum. This is the ID for personal configuration through IP PBX Web management.
Name	Name of the user, either a real or a virtual one, e.g. Alice Lee or Conference Room.
Password	Password for the user to access IP PBX Web management.
Description	Arbitrary description information.
E-mail Address	E-mail address of the user for voicemail notification.
Attach Voicemail in E-mail Notification	Select to enclose the message received in the notification e-mail as an attachment.
Usergroup	Select the usergroup this user belongs to.  If there is not any appropriate usergroup to select, come back later to revise this selection if no appropriate usergroup could be chosen for now.

Step 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.

Step 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.

Enter settings shown in **Table 3-3** ACC (Automatic Client Configuration) Settings and click **ENABLE**.


Note: Consult with your vendor to make sure your SIP phone support Auto Provision function.

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

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

Click **Next** to set a device.

Table 3-3 ACC (Automatic Client Configuration) Settings

Field	Description
Vendor Prefix	Ask your IP Phone vendor for the Prefix. e.g. eip7012.
MAC Address	MAC address of the device.
Supplementary Configuration	Supplementary configuration files for IP Phone. The file name must start with "psc-".
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.

Setp 5: Add Extension


3.1.5.1 Add Extension of IP Phone

1. Enter settings shown in **Table 3-4**.
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 3-4 Add Extension of IP Phone Settings

Field	Description
Extension Number	A unique line number composed of digits only, e.g. 101; 20 digits maximum. This is the login ID on the device configuration side.
Associated Device	Select the Device this extension associates with.
User ³	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.
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. Select Include Reachables check box to be able to pickup calls that belong to other usergroups which is configured in Reachable User Groups of the selected usergroup.
Language	Preferred language for system instructions heard from the extension.
Voicemail	Select enable to allocate voicemail account for the extension.
Voicemail PIN	PIN to access voicemails. This is mandatory if above voicemail option is enabled.
Max Voicemail Space	Enter maximum space in KBytes for voicemail. Enter 0 or leave it as blank for not limiting the voicemail space. The

³ Please refer to 6.1 for details.

	voicemail will be recorded until the storage is full.
Disable Fast Bridging	<p>Select to disable express media forwarding.</p> <p> With Fast Bridging feature enabled, if the two parties involved in a call (for example, one IP extension and one SIP trunk) use different DTMF modes (RFC2833/SIP INFO/Inband), inline transfer (*#) or 2nd-dialing might fail. To avoid such problem, it is recommended to set the same DTMF mode for all IP extensions and SIP trunks in the IPBX, as well as for all IP phones registered to the IPBX. If it is not feasible to set the same DTMF mode for some IP extensions or SIP trunks, and inline transfer or 2nd-dialing is necessary for those IP extensions or SIP trunks, the Fast Bridging feature can be disabled on a per IP extension and per SIP trunk basis. Note that Fast Bridging is enabled by default.</p>
Try Peer-to-peer RTP	<p>If click YES, IP PBX will attempt to notify the two peers in a conversation to try peer-to-peer RTP transmission. This is suggested as long as phones support INVITE or UPDATE method during a connected call to save the resource of IP PBX. However, only SIP INFO DTMF mode phones should enable this since other DTMF modes require IP PBX being RTP relay server to support in-line transfer.</p>
DTMF Mode	<p>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.</p> <p> 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.</p>


3.1.5.2 Add Analog Phone

1. Enter settings shown in **Table 3-5**.
2. Click **ADD** to see the newly added analog phone in the table of the webpage.

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

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

Table 3-5 Add Analog Phone Settings

Field	Description
POTS Port	FXS port index.
Pickup Group	The pickup group that the extension belongs to. Select Include Reachables check box to be able to pickup calls that belong to other usergroups which is configured in Reachable User Groups of the selected usergroup.
Extension Number	A unique line number composed of digits only, e.g. 101; 20 digits maximum.
Unavailable Timeout	Timeout for ringing before a call is answered.
User ⁴	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.
Language	Preferred language for system instructions heard from the extension.
Voicemail	Select Enable to allocate voicemail account for the extension.
Voicemail PIN	PIN to access voicemails. This is mandatory if above voicemail option is enabled.
Max Voicemail Space	Enter maximum space in KBytes for voicemail. Enter 0 or leave it as blank for not limiting the voicemail space. The

⁴ Please refer to 6.1 for details.

	voicemail will be recorded until the storage is full.
--	---

3.2 Add Route & Trunk

In **Add Route & Trunk** page, 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 **Reload** at the bottom of the homepage to take the configuration effect.

Step 1: Add Route

1. Enter settings shown in **Table 3-6**.
2. Click **ADD** to see the newly added route in the table in the webpage.

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

Click **Next** to set a routegroup.

Table 3-6 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 ⁵	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

⁵ For more information about the available digit set and wildcard characters, please refer to **Table 6-7**.

	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.

Setp 2: Add Route 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 3-7**.
5. Click **Back** to return to the **ADD ROUTE GROUP** page.

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

Click **Next** to choose a trunk.

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

Table 3-7 Add Routegroup Settings

Field	Description
Description	Arbitrary description information. Click SET to add/update the information.
Associated Routes ⁶	<p>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 would be matched first for an outgoing call.</p> <p> If there is no appropriate routes to select initially, one can come back later to revise it, once the expected routes are added.</p>

⁶ Please refer to 6.4 for details.

Setp 3: Choose Trunk

In the **Choose Trunk** page, click **SIP TRUNK**, **ANALOG PSTN TRUNK** or **ISDN PSTN TRUNK** to see one of the following pages to add various types of trunks.

Setp 4: Add Trunk

3.2.4.1 Add SIP Trunks


1. Enter settings shown in **Table 3-8**.
2. Click **ADD** to see the newly added SIP trunk in the table in the webpage.



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

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

Table 3-8 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 IP PBX at a remote site with dynamic IP address. Once the remote IP PBX registers, calls from local to remote can be made reversely over the trunk.
SIP Proxy IP	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP proxy, which usually refer to the SIP server on the ITSP side.
SIP Proxy Port	
Registration Required	Select if registration to a registrar is required to activate the trunk. This is true for a remote IP PBX or an ITSP account,

	however, may be not required in case of a SIP gateway.
SIP Registrar IP	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP registrar, which usually refer to the SIP server on the ITSP side (same as proxy).
SIP Registrar Port	
Language	Preferred language for system instructions heard from the trunk.
DID Type	DID means direct inward dialing (also called DDI in Europe). Select a preferred type, None DID , Extension DID , DID by Number DID by Privilege or Centrex DID from the list and then enter configuration in DID Prefix and DID Stripping to have the incoming calls directed to the corresponding trunk.
None DID	When selected None DID , all incoming calls will enter IVR system instead of directly dial to a specified extension. Select a preferred IVR from IVR list for this trunk.
Extension DID	When selected Extension DID , select an extension in the list to be an unconditional destination for incoming calls to this trunk. If prefix or stripping has been given, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2 nd dialing.  If you set a DID extension in a trunk, then only that extension can use this trunk to call out, and all incoming calls to this trunk will connect to that extension directly.
DID By Number	When selected DID By number , 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.
DID By Privilege	When selected DID By Privilege , select a usergroup in the list as the privilege of inbound calls from this trunk. Enter configuration in DID Prefix and DID Stripping to have the incoming calls redirected to dial out.
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. Click All to strip all digits of the original called number.								
Centrex DID	<p>Select this function to make the numbers of incoming/outgoing calls more flexible by prefixing/stripping digits.</p> <table border="1"> <tr> <td>All-number Digitmap</td> <td>Select a digitmap ID for calls via the trunk changing numbers.</td> </tr> <tr> <td>Default Number</td> <td>Enter digits for calls from this trunk displaying this caller ID.</td> </tr> <tr> <td>Inbound Manipulation Digitmap</td> <td>Select a digitmap ID to have the incoming calls direct to the corresponding extensions.</td> </tr> <tr> <td>Outbound Manipulation Digitmap</td> <td>Select a digitmap ID to have the outgoing calls display the corresponding numbers.</td> </tr> </table>	All-number Digitmap	Select a digitmap ID for calls via the trunk changing numbers.	Default Number	Enter digits for calls from this trunk displaying this caller ID.	Inbound Manipulation Digitmap	Select a digitmap ID to have the incoming calls direct to the corresponding extensions.	Outbound Manipulation Digitmap	Select a digitmap ID to have the outgoing calls display the corresponding numbers.
All-number Digitmap	Select a digitmap ID for calls via the trunk changing numbers.								
Default Number	Enter digits for calls from this trunk displaying this caller ID.								
Inbound Manipulation Digitmap	Select a digitmap ID to have the incoming calls direct to the corresponding extensions.								
Outbound Manipulation Digitmap	Select a digitmap ID to have the outgoing calls display the corresponding numbers.								
IVR List ⁷	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 ⁸ of Privilege	<p>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.</p> <p> There may not be appropriate usergroups to select initially. One can come back later once the expected usergroup has been added.</p>								
Disable Fast Bridging	<p>Select to disable express media forwarding.</p> <p> With Fast Bridging feature enabled, if the two parties involved in a call (for example, one IP extension and one SIP trunk) use different DTMF modes (RFC2833/SIP INFO/Inband), inline transfer (*#) or</p>								

⁷ Please refer to 7.12 for details.

⁸ Please refer to 6.1 for details.

	<p>2nd-dialing might fail. To avoid such problem, it is recommended to set the same DTMF mode for all IP extensions and SIP trunks in the IPBX, as well as for all IP phones registered to the IPBX. If it is not feasible to set the same DTMF mode for some IP extensions or SIP trunks, and inline transfer or 2nd-dialing is necessary for those IP extensions or SIP trunks, the Fast Bridging feature can be disabled on a per IP extension and per SIP trunk basis. Note that Fast Bridging is enabled by default.</p>
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3.2.4.2 Add Analog PSTN Trunks


1. Enter settings shown in **Table 3-9**.
2. Click **ADD** to see the newly added analog PSTN trunk in the table in the webpage.

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

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



Table 3-9 Add Analog PSTN Trunk Settings

Field	Description
Trunk Group	ID number of this PSTN trunk group. A valid number ranges from 1 to 31. 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	Select one or more FXO or FXS ports for this Analog PSTN trunk.
Description	Arbitrary description information.
Port Selection	Click to search for an available port in the group. Rotating means to force ports being selected in turns to even cost.
DID Type	DID means direct inward dialing (also called DDI in Europe). Select a preferred type, None DID , Extension DID , and DID by Privilege from the list and then enter configuration in DID Prefix and DID Stripping to have the incoming calls

	directed to the corresponding trunk.
None DID	When selected None DID , all incoming calls will enter IVR system instead of directly dial to a specified extension. Select a preferred IVR from IVR list for this trunk.
Extension DID	When selected Extension DID , select an extension in the list to be an unconditional destination for incoming calls to this trunk. If prefix or stripping has been given, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2 nd dialing.  If you set a DID extension in a trunk, then only that extension can use this trunk to call out, and all incoming calls to this trunk will connect to that extension directly.
DID By Privilege	When selected DID By Privilege, select a usergroup in the list as the privilege of inbound calls from this trunk. Enter configuration in DID Prefix and DID Stripping to have the incoming calls redirected to dial out.
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. Click All to strip all digits of the original called number.
Language	Preferred language for system instructions heard from the trunk.
IVR List ⁹	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 ¹⁰ of Privilege	When disabled DID, click a usergroup in the list whose reachability to other usergroups and trunks will be used as

⁹ Please refer to 7.12 for details.

¹⁰ Please refer to 6.1 for details.

	<p>the privilege of inbound calls from this trunk.</p> <p> There may not be any appropriate usergroups to select initially. One can come back later to revise it, once the expected usergroups are added.</p>
Caller ID Detection	Select to detect the Caller ID calling from PSTN lines.
Answering by Battery Reversal Detection	<p>If selected, billable time will count from the call is answered.</p> <p> Please enable this function when Central Office (CO) site provides battery reversal.</p>


3.2.4.3 Add ISDN PSTN Trunks


1. Enter settings shown in **Table 3-10**.
2. Click **ADD** to see the newly added ISDN PSTN trunk in the table in the webpage.


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

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

Table 3-10 Add ISDN PSTN Trunk Settings



Field	Description
Trunk Group	ID number of this ISDN trunk group. A valid number ranges from 1 to 31. It should not overlap with existing Analog PSTN trunk groups.
Trunk Channels	<p>The Trunk Channels is the logical range of the sum of B and D channels. Each physical ISDN port occupies three Trunk Ports, two B and one D channels. User only needs to specify the B channel number here, since D channel is reserved in the 3rd trunk port for each physical ISDN port. E.g. Assume there are four ISDN ports in the PBX and no other FXO/FXS modules installed, then one can set each pair of numbers here, like 1,2 but excluding 3,6,9,12.</p> <p> If a four-port FXO/FXS module is also installed, then the Trunk Ports here should be numbered from 5 to 16 instead of 1 to 12. Make sure to specify the indices of ports correctly, or PBX will not start. One can refer to</p>

	the POTS Setting page before configuration.
Description	Arbitrary description information.
Port Selection	Select to search for an available port in the group. Rotating means to force ports being selected in turns to even cost.
DID Type	DID means direct inward dialing (also called DDI in Europe). Select a preferred type, None DID , Extension DID , DID by Number DID by Privilege or Centrex DID from the list and then enter configuration in DID Prefix and DID Stripping to have the incoming calls directed to the corresponding trunk.
None DID	When selected None DID , all incoming calls will enter IVR system instead of directly dial to a specified extension. Select a preferred IVR from IVR list for this trunk.
Extension DID	When selected Extension DID , select an extension in the list to be an unconditional destination for incoming calls to this trunk. If prefix or stripping has been given, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2 nd dialing.  If you set a DID extension in a trunk, then only that extension can use this trunk to call out, and all incoming calls to this trunk will connect to that extension directly.
DID By Number	When selected DID By number , 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.
DID By Privilege	When selected DID By Privilege , select a usergroup in the list as the privilege of inbound calls from this trunk. Enter configuration in DID Prefix and DID Stripping to have the incoming calls redirected to dial out.
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. Click All to strip all digits of the original called number.

Language	Preferred language for system instructions heard from the trunk.
IVR List ¹¹	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID. Leave it blank and the system will automatically create an IVR for the trunk.
Usergroup ¹² of Privilege	When disabled DID, clicks a usergroup in the list whose reachability to other usergroups and trunks will use as the privilege of inbound calls from this trunk.  There may not be any appropriate usergroups to select initially. One can come back later to revise it, once the expected usergroups are added.

Setp 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 3-11**.
3. Click  or  to add or delete the associate trunks. After adding all trunks, click **APPLY**.

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

Table 3-11 Assign Trunk Settings

Field	Description
Routegroup	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.

¹¹ Please refer to 7.12 for details.

¹² Please refer to 6.1 for details.

Weight	The weight of a trunk to be selected in a trunk balance group for an outgoing call.
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3.3 Mass Extension Adding

The **Mass Extension Adding Management** 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.

Step 1: Add User & Extension

1. Enter the digits for the starting extension.
2. Click a number in the **Number of EXT** list for the mass extension adding.
3. Click a usergroup in the **USERGROUP** list.
4. Select enable to allocate voicemail account for the extension.
5. Click an IP phone device in the **DEVICE** list.
6. Select to enable Try-Peer-to-peer RTP. If click **INVITE** or **UPDATE**, IP PBX will attempt to notify the two peers in a conversation to try peer-to-peer RTP transmission.
7. Click a usergroup in the **Pickup Group** list.
8. Select **Include Reachables** check box to enable pickup calls that belong to other usergroup which is configured in Reachable User Groups of the selected usergroup.
9. Choose preferred DTMF mode for extensions. Currently supported types include RFC2833, SIP INFO and Inband.
10. Click **ADD**, and IP PBX will start to add these extensions automatically.
11. Click **Back** to the homepage.

4 System Configuration

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

4.1 PBX System

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

4.2 Time Setup

The **Time Setup** page allows administrator to configure time zone and date for IP PBX. 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 System Time Zone

1. Click a region/country in the **Time Zone** list.
2. Click **Apply** in **System Timezone Setup**.
3. Go to **System -> Shutdown**, select **Rebooting After Shutdown** and Click **Yes** to activate changes.

4.2.2 Real Time Clock (RTC) Setup

1. Click year, month, day, hour, minute, and second in the correspondent list.
2. 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 On-board WAN Setup

The **On-board WAN Setup** page allows administrator to configure WAN network interface for IP PBX. Select **System -> On-board WAN Setup**, and the current setting of WAN network interface is displayed, e.g. type, IP address etc.

Unless the **LAN Only** is selected, you can choose one of the three options, **Static IP**, **DHCP**, and **PPPoE** from the **Type** list for your configuration. Select **LAN Only** check box to disable WAN, and only default router and DNS settings are applicable. Select **MAC Clone** to change the WAN MAC address. Select **Yes** to allow WAN to respond PING request.

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, default gateway and DNS server 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, default gateway and DNS server information.

4.3.4 Allow WAN to Respond PING

Select this option if appropriate and click **Set**. If enabled, the WAN port of IP PBX will respond PING to the originator.

4.3.5 LAN Only

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

4.3.6 MAC Clone

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

4.4 On-board LAN Setup

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

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

Note: By default IP PBX grants IP addresses to LAN devices via DHCP, and translates those addresses into its WAN IP address for access beyond the LAN subnet. Make sure to change DHCP pool and LAN routing (if any) accordingly after changing the system LAN IP subnet. After configuration, go to **Service** -> **IP PBX 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 IP PBX's LAN. It is important to assure that the given gateway IP address sits in the IP PBX's LAN. Each subnet requires an entry even multiple subnets share the same gateway, unless masking does the same.

Examples are adding IP Route IDs *net1* and *net2* with parameters 192.168.128.0/255.255.255.0, 192.168.129.0/255.255.255.0, shared gateway 192.168.1.254 respectively. Or, IP Route ID *net1n2* with 192.168.128.0/255.255.254.0 and gateway 192.168.1.254 would do the same. Added routes enable routing immediately after clicking **Add**. However, the IP PBX Service needs to be restarted to regard calls from designated LAN subnets as LAN traffic. Go to **Service** -> **IP PBX Service**, and click **Restart** to regard calls as LAN traffic.

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 Route 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 **Delete** 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 IP PBX systems. A popular work-around is to adopt domain names provided by Dynamic DNS service providers and run a client on or behind the gateway router (or IP PBX). It is required to apply an account and create a hostname in the account before configuration.

Click **Enable**, give account information and refresh interval to activate a Dynamic DNS client. The client then uses **Username** and **Password** to access its account and update periodically the **Hostname** with the latest WAN IP address at **DynDNS** or **3322.net Service**.

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.

1. Click **Enable**.
2. Click **DynDNS** or **3322.net** in the **Service** list.
3. Enter the **Username**, **Password**, and **Hostname**.
4. 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**, **WAN Downlink Speed**, and **Uplink VoIP Reserved** (bandwidth).
3. Click **Apply**.

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

Codec Information				Bandwidth Calculations			
Codec & Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth Required (Kbps)
G.711 (64 Kbps)	80	10	4.1	160	20	50	87.2
G.729 (8 Kbps)	10	10	3.92	20	20	50	31.2
G.723.1 (6.3 Kbps)	24	30	3.9	24	30	34	21.9
G.723.1 (5.3 Kbps)	20	30	3.8	20	30	34	20.8
G.726 (32 Kbps)	20	5	3.85	80	20	50	55.2
G.726 (24 Kbps)	15	5		60	20	50	47.2

4.7.2 Disable QoS

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

4.8 Virtual Server

You can configure IP PBX as a virtual server for remote users to access services such as the Web or FTP at your local site via Public IP Addresses. With proper settings, IP PBX can automatically redirect inbound traffic from WAN to local servers configured with private IP addresses. In other words,

depending on the requested service (TCP/UDP) port number, the IP PBX redirects the external service request to the appropriate internal server (located at one of your LAN's Private IP Address).

To enable access servers in LAN from a machine beyond WAN, select **System -> Virtual Server** to configure port mappings. **Service ID** names the service. **Protocol** and **Port** specify the TCP/UDP port number on WAN IP to be forwarded to the **Forward to Port** of **Forward to IP** in LAN. Say 192.168.1.5 is a Mail Server to be seen from outside, one should configure TCP port 25 to be forwarded to 192.168.1.5 port 25.

Note: Following ports are default server ports in use: TCP port 9, 7788, 80, 53, 23, 443 and 13422; UDP port 1024, 9, 53, 67, 5060 and 69.

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.**

Note: the usage of port range: using the syntax **from-port : to-port.** For example, enter the following for the TCP ports 6881 to 6999. It should be 6881:6999.

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 **Delete** to remove the service from the **Service ID.**

4.9 Maintenance

This page includes maintenance functions of IP PBX, 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 (yyyy-mm-dd-hhmmss, eg. 2007-06-26-151100) will be created in ippbx-backup folder on the external drive. After the backup, remove the USB connector of the external drive.

4.9.1.2 Back up to NFS Server

1. Click **NFS Server**.
2. Enter a URL path in **NFS Directory Path**.
3. 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/remote IP PBX, and the **IP Address/Port** from where they register. SIP trunk registrations, if any, also show at the end of the list. The **Dynamic** column shows the listed IP address is dynamic or static. **Reg. Progress** is the response code and message if registration has been attempted but not successful so far. **Slave Registrar** column is used only under the stackable mode. It indicates with which slave box a SIP client is registered. Blank means a client is registered to the master box locally. IP PBX supports share-line function and all information of share-line extension/trunk is recorded on SIP UA page, To remove the specific registered extension/trunk ID, please click **Delete** button in that row.

Table 4-1 The Registration Status of SIP UA

IP PBX	The Description of Reg. Progress	
UAS	Empty	If there is no registration activities existing.
	401 Unauthorized	Authentication failed.
	407 Authentication Required	Authentication required.
UAC	Empty	If there is no registration activities existing.
	Status code and message from UAS	The response information is depended on UAS.

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, epochs when the call was **made**, **answered** and **ended**, and which yield the **Call Duration** and **Billable Time**. The **Result** denotes the disposition of a call like answered or not. The administrator can also click **Get File** to export the record as a *.csv file, click **Delete CDR** to remove all records and click **Show CDR** to display all records on the screen.

4.9.4 System Events

The log records an audit entry in the audit trail whenever certain events occur, such as service reload, firmware upgrade, administrator logging on, logging off, and accessing resources from NTP, DNS, DHCP, and PPPoE service.

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 4-2 The State of Active Call

Field	Description	
Caller	Show the caller's extension number, port number, or SIP trunk ID.	
Caller state	Dial	The client is caller.
	IVR	Calls from FXO are picked up by Auto-Attendant.
	Pause	The client is being paused for some time.
	Prompt	The client is being prompted for a pre-recorded audio.
	Voicemail	The client enters voicemail service.
	Meet-me	The client enters meet-me service.
	Busy Call-back	The client enters busy call-back service.
	ACD	The client enters ACD service.
Trunk	Shows trunk ID that is using in this call.	
Callee state	Answer	The client is callee.
Callee	Show the callee's extension number, port number, or SIP trunk ID.	
Answer Time	Show the answer time of active calls.	

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: Consult with your IP PBX service provider for upgrade procedure.

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

4.11 Stackable Management

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

4.11.1 Operation Modes

IP PBX provides three modes, standalone mode, master mode and slave mode. The default mode is standalone mode while there is no cluster and IP PBX box works independently. Once a cluster is formed, only one IP PBX box runs in master mode, and rest of boxes run in slave mode in the cluster. For instance, with full-capacity case, there would be one master box and 3 slave boxes in a cluster. Select **System** -> **Stackable** to see the current operation mode of each IP PBX box.

4.11.2 Consolidated Management

Stackable Management page of the master box not only indicates IP addresses of current slave boxes in a cluster but also allows you to add new slave boxes to the cluster or remove slave boxes from the cluster. You can simply manage a cluster in Stackable Management page of the master box.

Within a cluster, any extension/user update of configuration information to the master box would affect all slave boxes automatically. As a result, you shall only configure the master box to take control over the whole cluster in most cases. Meanwhile, all boxes within the same cluster share the same registration information and work as one single box.

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

Note: For Meet-me conferences in stack, do not enter the same room number in slave and master boxes, though the function need to be configured respectively.

4.11.3 Configuration Procedure

The following examples illustrate how to configure a stack.

4.11.3.1 Case I: Stack multiple IP PBX boxes from scratch

1. Ensure all boxes have factory default settings before being stacked.
2. Ensure all boxes have TCP routing to each other via LAN interfaces if they are in different local networks.
3. Select one of the boxes to be the master box. Others are slave boxes consequently.
4. For each slave box, go to **System -> Stackable** page to specify the master's IP address and click **Apply**. Make sure to configure slave boxes before assigning to master box.
5. For the master box, go to **System -> Stackable** page. Select **Enable Master Mode** check box.
6. Enter IP address and port of a slave box, and then click **Add**. If clients support SIP 305 or SIP 302 response, select the **By 305 Use Proxy** or **By 302 Moved Temporarily** check box in **Registration Load Balance**.
Do this step again if more than one slave box is stacked.
7. After adding all slave boxes, click **Apply**.
8. Go to **Service -> IP PBX Service** and click **Restart** to reflect the changes.

All boxes are stacked and a cluster is formed. You can start to configure extensions etc. on master box to make internal calls possible. You can also configure SIP and PSTN trunks on each box to enable inbound and outbound calls. Any configuration change in stack requires **Reload** in **Service -> IP PBX Service** of master box.

In order to share SIP and FXO PSTN trunks with other boxes, you need to do the following settings. For instance, a FXO PSTN trunk (with a route pattern) is configured on box A and assigned to some usergroups. You would like to share this FXO PSTN trunk with box B.

1. On box B, configure a route with the pattern equals to the one associated with this FXO PSTN trunk, and then assign this route to the routegroup associated with the intra trunk from box B toward box A.
2. Click **Reload** in **Service -> IP PBX Service** of master box.

To explain step one more clearly, for example, box A is 192.168.1.100 and box B is 192.168.1.101. There is a FXO PSTN trunk "pstn1" with a route pattern "9./1"¹³ configured on box A. In order to share this FXO PSTN trunk among box A and B, on box B we can configure a route "BoxA-pstn1" with pattern

¹³ 9./1 means pattern is 9. and number of stripped digits is 1.

“9./0”¹⁴ and assign it to the routegroup “_192168001101_192168001100” which associates with the intra trunk “_192168001101_192168001100”¹⁵.

4.11.3.2 Case II: Stack a new IP PBX box with an existing running box

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

1. Ensure the new box has factory default settings before being stacked.
2. Ensure the new box has TCP routing to the existing box via LAN interface if they are in different local networks.
3. Now, backup the IP PBX configuration on the existing box. Then restore this configuration to the new one.
4. On new box, remove all trunk configurations including routes and routegroups, and all Meet-me conference configurations first.
5. Go to **System** -> **Stackable** of the new box, specify master's IP address and click **Apply**.
6. Go to **System** -> **Stackable** of the existing box, select **Enable Master Mode** check box, enter IP address and SIP port of the new box, and then click **Add**.
7. Click **Apply**.
8. Go to **Service** -> **IP PBX Service** of the existing box, and click **Restart** to reflect the changes.

The existing box turns to be a master box and the new box is a slave box in the cluster.

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

Similar to Case II, you can also add more slave boxes to an existing cluster.

4.11.3.4 Remove slave boxes from stack

1. Go to **System** -> **Stackable** of the master box.
2. Select IP address of a slave box you would like to remove in **Slave Peers**, and then click **Delete**.
3. Click **Apply**.
4. Go to **Service** -> **IP PBX Service** of the master box, and click **Restart** to reflect the changes.

The cluster no longer owns the removed box. The removed box gets off the cluster and would turn back to be standalone. Moreover, the master box would also turn back to be standalone if all slave boxes are removed and the cluster does not exist any more.

¹⁴ Here, please note that we always use 0 for number of stripped digits.


¹⁵ You probably notice IP addresses of two boxes meshed by an intra trunk determine name of the intra trunk. In this case, the name of intra trunk from box B (192.168.1.101) toward box A (192.168.1.100) is _192168001101_192168001100. Meanwhile, the routegroup associated with this intra trunk has the same name.

4.12 Shutdown

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

Note: If the hardware **RESET** button is hold for 5 seconds, the system will revert to factory default.

4.13 Logout

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

Note: The GUI will automatically log out after 15 minutes, if you do not do any action.

5 Service Configuration

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

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

IP PBX has a built-in STUN client to solve NAT problems. Select **Service** -> **STUN Service** to specify a Simple Traversal of UDP through NATs (STUN) server for NAT traversal. You can enable or disable STUN Service at any time.

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** -> **IP PBX Service**, and click **Restart** to reflect the changes.

5.3.2 Disable STUN Service

1. Click **Disable**.
2. Enter the fully qualified domain name or the static IP address of the external WAN interface.
Usually this address refers to the static WAN IP address if there is a NAT device between the IP PBX and the Internet. If the WAN port of IP PBX directly connects to Internet or it is unused, leave the address blank.
3. Click **Apply**.
4. Go to **Service** -> **IP PBX 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 IP PBX. 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 **Delete 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.

Note: The maximum file size for uploading a file to PC#1 is 12M, and to PC#2 is 6M.

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 IP PBX was shut down abnormally, Select **Service** -> **DHCP Service** and click **Apply**, or Go to **Service** -> **IP PBX Service**, and click **Restart** to activate the DHCP service.

5.5.1 DHCP Service

5.5.1.1 Enable DHCP Service



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

5.5.1.2 Disable DHCP Service

Click **Disable** and click **Apply**.

5.5.2 Add New

5.5.2.1 Add DHCP Range

1. Click **Add New** tab.
2. Enter a pool name (must have an alphabet initial) in **Pool Name**.
3. Select **Range** or **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 options¹⁶ could be configured by entering an option code and value in **Code,Value** and click . The new DHCP option will show in the **Options** list. To delete an option, choose it from the **Options** list and click  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.

5.5.2.2 Edit DHCP Range

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

¹⁶ Refer to RFC 2132 for the details of available DHCP options.

2. Edit the settings.
3. Click **Update** to change the settings.

5.5.2.3 Delete DHCP Range

1. Click any pool name in the **DHCP POOL** list.
2. Click **Delete** to remove the pool name from the **DHCP POOL** list.

5.5.3 Show Leased Clients

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

5.6 IP PBX Service

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

5.6.1 Operations

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

5.6.1.1 Reload IP PBX Configuration

Click **Reload**, and IP PBX will reload the configuration once there is no active call. If there is any active call, it will retain up to 3 minutes, and then IP PBX will reload. This is the most frequently used function in this page since any IP PBX configuration change has to be reloaded to take effect.

5.6.1.2 Backup IP PBX Configuration

Click **Backup**, and IP PBX archives and encrypts current configuration into a time-stamped backup file under tftpboot root directory. To secure configuration files, download them to a local host through the **Get File** function in **Service -> TFTP Service** once a while. Select the **PBX Settings Only** check box for only back up Configurations in PBX, which are parts mainly from User Management to Feature in the tree view; clear the check box to back up both PBX and system (interfaces and services) settings.

Note: Do not change the configuration file name, or the **Restore** function will reject the configuration file.

5.6.1.3 Restore IP PBX Configuration

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

5.6.1.4 Restart IP PBX Configuration

Click **Restart**, and the IP PBX Service will restart completely. Currently active calls will be disconnected immediately. This function is rarely required unless the network setting has been changed, or the service operates abnormally without problematic configuration could be identified.

5.6.1.5 Revert IP PBX Configuration


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

To revert the whole system back to the factory default, hold the hardware **RESET** button more than 5 seconds. All of the configuration settings will be re-set exactly the way they are when users receive the IP PBX from providers. Users have to re-configure the IP PBX 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 Settings

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

Table 5-1 IP PBX Configuration Settings

Field	Description
PBX SIP Port	Specify the UDP port where the SIP service listens on.
RTP Port Range	Limit the UDP ports used by the IP PBX for media transport.  The port range needs to have at least equals to the (number of extensions (also count shared-lines) + number of SIP trunks) * 2. If selecting Enable Video Codec , the total amount needs to multiply by 2 to have the least requirements for RTP port range. Insufficient range of ports may cause call blocking due to port depletion.
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 IP PBX.
Call Keep Alive	Enter a time in second to periodically confirm the call is still connected. The time is between 10 and 180 seconds.
Registered Keep Alive	Enter a time in second to confirm the registration status periodically. The time is between 10 and 180 seconds.
NAT Traversal Keep Alive	Enter a time in seconds to retain NAT hole punched periodically. 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.
Disable Line-in-use Call Back	PBX supports call back function when an extension without multi-line function is in use. Select to disable Line-in-use Call Back function.
Disable Exponential Backoff in SIP Retransmission	Select to disable exponential SIP retransmission.

6 IP PBX Configuration

This section introduces steps to provision the IP telephony part of the IP PBX. Note that reloading configuration is required in order to make new configuration effective¹⁷.

6.1 User Group Configuration

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

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

Note: The maximum usergroup is 64 for PC#1 and 32 is for PC#2.

6.1.1 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 6-1**.
5. Click **Back** to return to the **Usergroup Management** page.

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

6.1.2 Edit a User Group

1. Click a usergroup name in the **Group ID**.
2. Edit settings shown in **Table 6-1**.
3. Click **Back** to see the updated information.

6.1.3 Delete a User Group




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

¹⁷ Please refer to 5.6.1.1 for details.




6.1.4 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 6-1 Usergroup Configuration Settings

Field	Description
Group ID	A unique group name containing alphabets, numbers, and underscore only without spaces; 20 characters maximum.
Description	Arbitrary description information. Click Set to add/update the information.
Associated Trunks ¹⁸	<p>Select routegroups and outbound trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time. Click Routegroup: display available routegroups.</p> <p>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. Normally, block is at the very first in the list, while loopback is at the last in the list.</p> <p>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.</p> <p>Weight: the weight of a trunk to be selected in a trunk balance group for an outgoing call.</p> <p>Click  or  to add or delete the associate trunks. After add all trunks, click Apply.</p> <p> If there is not any appropriate SIP trunk and PSTN trunks to select, come back later to revise selection once trunks have been created.</p>

¹⁸ Please refer to 6.6, 6.7, and 6.8 for details.

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 ¹⁹	Select PBX features enabled to this usergroup.
Member List	Show the users 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.
Group Call	A group call number is a virtual number that presents all extensions belong to this group. Enter a group call number (not an existed extension/service number). Select extensions and click  to add into the group call. After adding all extensions, click Apply .

6.2 User Configuration

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

Select **User Management** -> **User**, and one can add, edit, delete or search users. Go to **Service** -> **IP PBX Service**, and click **Reload** to activate changes.

6.2.1 Add a User

1. Click the **Add New** tab.
2. Enter settings shown in **Table 6-2**.
3. Click **Add**.

¹⁹ Please refer to 7 for details.

4. Click **Back** to see the newly added user in the **Login ID**.

6.2.2 Edit a User

1. Click a user in the **Login ID**.
2. Edit settings shown in **Table 6-2**.
3. Click **Apply**.

6.2.3 Clone a User

1. Click a user in the **Login ID**.
2. Click **Clone**.
3. Edit settings shown in **Table 6-2**.
4. Click **Add**.
5. Click **Back** to see the newly added user in the **Login ID**.

6.2.4 Delete a User


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

6.2.5 Search a User

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

Table 6-2 User Configuration Settings

Field	Description
Login ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 20 characters maximum. This is the ID for personal configuration through IP PBX Web management.
Name	Name of the user, either a real or a virtual one, e.g. Alice Lee or Conference Room.
Password	Password for the user to access IP PBX Web management.
Description	Arbitrary description information.
E-mail Address	E-mail address of the user for voicemail notification.
Attach Voicemail in E-mail Notification	Select to enclose the message received in the notification

	e-mail as an attachment.
Usergroup	Select the usergroup this user belongs to.  If there is not any appropriate usergroup to select, come back later to revise this selection if no appropriate usergroup could be chosen for now.
Extensions	Show the extensions associated with this user.

6.3 Device Configuration

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

6.3.1 IP Phone

The DEVICE PHONE MANAGEMENT page lets the administrator to create IP Phone devices. Before connecting a device to the IP PBX, you need to configure the same account information 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** -> **IP PBX 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 for fast connection to device configuration page or description for the device 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** is a reference for detailed ACC settings which is used for auto-configuring IP phones. One can specify the MAC address and audio preferences of the phone. Note that for phones using HTTP for auto-configuring, DHCP setting needs a new option 151 with a value of *http://<IP PBX LAN IP>/tftpboot/* in the **Code,Value** box in **Service** -> **DHCP Service**. If the phone uses TFTP for auto-configuration, the option is 150 by default.
3. Click **Enable** to see **Enable** shows 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 **Delete** 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 ACC (Automatic Client Configuration) Settings

Field	Description	
Device	A unique ID containing alphabets, numbers, and underscore only without spaces; 20 characters maximum.	
Vendor Prefix	Ask your IP Phone vendor for the Prefix. e.g. eip7012.	
MAC Address	MAC address of the device.	
Supplementary Configuration	Supplementary configuration files for IP Phone. The file name must start with "psc-".	
Codec Preference	Preference order of supported codec and packet times of the phone.	
Redundant Servers	Enter the information as follows.	
	Proxy IP/FQDN	Specify IP address (or fully qualified domain name) and UDP port of the remote IP PBX.
	Proxy Port	
	Registrar IP/FQDN	Specify IP address (or fully qualified domain name) and UDP port of the remote IP PBX
	Registrar Port	
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.	
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.	
Daylight Saving	IP PBX supports auto adjustment for daylight saving time. To enable this function, the administrator must configure the Daylight Format , Start Time and End Time . The following example	

	configures daylight saving time for the U,S, adding one hour starting at midnight on the first Sunday in April and ending at midnight on the last Sunday of October. Select By weekday of a specific month on the Daylight Format, set First Sunday of April at 00 O'clock on the Start Time, and set Last Sunday of October at 00 O'clock on the End Time.
--	--

6.3.2 Extension of IP Phone

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

IP PBX also offers its own extensions Line-in-use Call Back and Transfer Call Back function. For example, extension A and B are within one IP PBX 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, IP PBX will ring extension B within a minute. When extension B is picked up, IP PBX 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.

For transfer-call-back function, extension A calls to extension B, extension B (transferer) tries to transfer the call to extension C (transferee). After the transferee's unavailable timeout reached, extension B gets re-ringback tone from IP PBX.

Note: To trigger the transfer-call-back-to function, all of the following conditions must be true:

1. It is phone-based ring-transfer.
2. The transferee is a local extension.
3. The transferee does not answer until the unavailable timeout is reached.
4. No forwarding settings to voicemail/extension for the transferee.

6.3.2.1 Add an Extension

1. Click the **Add New** tab to set an extension.

2. Enter settings shown in **Table 6-4**.
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 6-4**.
3. Click **Apply** to see the updated information.

6.3.2.3 Clone an Extension

1. Click an extension in the **Extension Number**.
2. Click **Clone**.
3. Edit settings shown in **Table 6-4**.
4. Click **Add** to see the updated information.


6.3.2.4 Delete an Extension

1. Select an extension numbers.
2. Click **Delete** to remove the extension from the **Extension Number**.


6.3.2.5 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 6-4 Device Extension Configuration Settings

Field	Description
Extension Number	A unique line number composed of digits only, e.g. 101; 20 digits maximum. This is the login ID for device registration.
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 ²⁰	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

²⁰ Please refer to 6.1 for details.

	<p>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.</p> <p>Select Include Reachables check box to be able to pickup calls that belong to other usergroups which is configured in Reachable User Groups of the selected usergroup.</p>
Line Type	Specify the type of connection, wired or wireless, of the client with the extension.
Language	Preferred language for system instructions heard from the extension.
Voicemail	Select enable to allocate voicemail account for the extension.
Voicemail PIN	PIN to access voicemails. This is mandatory if above voicemail option is enabled.
Max Voicemail Space	Enter maximum space in KBytes for voicemail. Enter 0 or leave it as blank for not limiting the voicemail space. The voicemail will be recorded until the storage is full.
Unavailable Timeout	Timeout for ringing before a call is answered.
Allow LAN Use Only	Check to reject registration and calls from WAN in a SIP ID same as the extension number. I.e., this extension must be on LAN.
Disable NAT Traversal	IP PBX uses NAT traversal for outgoing traffics by default. Select to disable NAT traversal if there is a machine that could handle NAT issues.
Call Keep Alive	Select to check if the call is still active. Frequency check is specified in 5.6.2.
Registered Keep Alive	Select to check if the register still exists in a certain time.
Disable Fast Bridging	<p>Select to disable express media forwarding.</p> <p> With Fast Bridging feature enabled, if the two parties involved in a call (for example, one IP extension and one SIP trunk) use different DTMF modes (RFC2833/SIP INFO/Inband), inline transfer (*#) or 2nd-dialing might fail. To avoid such problem, it is recommended to set the same DTMF mode for all IP</p>

	<p>extensions and SIP trunks in the IPBX, as well as for all IP phones registered to the IPBX. If it is not feasible to set the same DTMF mode for some IP extensions or SIP trunks, and inline transfer or 2nd-dialing is necessary for those IP extensions or SIP trunks, the Fast Bridging feature can be disabled on a per IP extension and per SIP trunk basis. Note that Fast Bridging is enabled by default.</p>
Try Peer-to-peer RTP	<p>If click YES, IP PBX will attempt to notify the two peers in a conversation to try peer-to-peer RTP transmission. This is suggested as long as phones support INVITE or UPDATE method during a connected call to save the resource of IP PBX. However, only SIP INFO DTMF mode phones should enable this since other DTMF modes require IP PBX being RTP relay server to support in-line transfer.</p>
DTMF Mode	<p>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.</p> <p> 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.</p>
Advanced Settings	<p>Select to see more optional settings shown below.</p>
Selective Call Blocking	<p>Select Block Anonymous Calls to block all calls without a Caller ID.</p> <p>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 .</p>
Block SIP redirection from the extension	<p>Select to ignore the forward settings of IP phone and honor the forward settings of IP PBX.</p>

Forward Options	<p>Select Unconditional Call Forward and clicks a default destination in the list, e.g. Voicemail or Phone Number.</p> <p> 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.</p>
Unavailable Call Forward	<p>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.</p>
Timeout To Next Forward	<p>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 .</p> <p> The time must be shorter than Unavailable Timeout, or the function will not work normally.</p>
Play Unavailable/Line-in-use Forward Prompt	<p>Notify the caller that callee is not available and the call is being forwarded to another extension.</p>
Line-in-use Forward	<p>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.</p> <p> If the function is enabled, the Line-in-use Call Back function will be disabled.</p>
Selective Call Forward	<p>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.</p>

6.3.3 Analog Phone

The **Analog Phone Management** page lets the administrator to create analog phones. Select **Device**

-> **Analog Phone**, and one can add, edit, and delete analog phones. Go to **Service** -> **IP PBX Service**, and click **Reload** to activate changes. Connect an analog phone to a FXS port and configure the properties of the port as detailed in **Table 6-5**.

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 6-5**.
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 6-5**.
3. Click **Apply** to see the edited information.




6.3.3.3 Delete an Analog Phone






1. Select a **POTS Port**.
2. Click **Delete** to remove the extension from the **POTS Port**.

Table 6-5 FXS Extension Configuration Settings

Field	Description
POTS Port	FXS port index.
Extension Number	A unique line number composed of digits only, e.g. 101; 20 digits maximum.
Pickup Group	The pickup group that the extension belongs to. 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. Select Include Reachables check box to be able to pickup calls that belong to other usergroups which is configured in Reachable User Groups of the selected usergroup.
Unavailable Timeout	Timeout for ringing before a call is answered.
User ²¹	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

²¹ Please refer to 6.1 for details.

	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. Enter 0 or leave it as blank for not limiting the voicemail space. The voicemail will be recorded until the storage is full.
Language	Preferred language for system instructions heard from the extension.
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.
Default Codec	Select g711ulaw or g711alaw for connecting to other devices.
Selective Call Blocking	Select Block Anonymous Calls to block all calls without a Caller ID 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	Select Unconditional Call Forward and click a default destination in the list, e.g. Voicemail or Phone Number.  If selecting Phone Number, enter a number to which incoming calls are forwarded unconditionally. The number could be an extension or a PSTN number with appropriate outbound prefix.
Unavailable Call Forward	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	Enter a period of time in seconds for ringing the extension in

	<p>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 .</p> <p> The time must be shorter than Unavailable Timeout, or the function will not work normally.</p>
Play Unavailable/Line-in-use Forward Prompt	Notify the caller that callee is not available and the call is being forwarded to another extension.
Line-in-use Forward	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	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.
Hot/Warmline	<p>A hot line means that a specified number will be called immediately when the phone goes off-hook. A warm line functions similarly except it does so only after a specified delay expired to allow dialing a different number with the line.</p> <p>Enter a number as the direct hot/warm line number, and optionally, a timeout period in seconds if the warm line behavior is expected.</p>

6.3.4 IP Phone Firmware

The **IP Phone Firmware Management** page is for uploading firmware of IP phone. When connecting an IP phone to PBX, the IP phone can automatically upgrade to the latest firmware once there is a version available for download on the IP PBX. Select **Device** -> **IP Phone Firmware** to add, edit, or delete IP phone firmware configurations.

6.3.4.1 Add IP Phone Firmware

1. Enter settings shown in **Table 6-6**.

2. Click **Put File** to see the newly added firmware in the table below.

Note: The uploaded file and auto generated file will be displayed at the TFTP Service page under root directory.

6.3.4.2 Delete IP Phone Firmware

1. Select a vendor prefix of IP phone.
2. Click **Delete** at the top-left side of the table to remove the firmware.

Note: The deleted file and originally auto generated file will be also deleted from the TFTP Service page under root directory.

Table 6-6 IP Phone Firmware Configuration Settings


Field	Description
Vendor Prefix	Ask your IP phone vendor for the prefix. e.g. eip7012.
Version	Enter the firmware version that is going to be upgraded for the IP phone, e.g. 1.0.0123
Firmware File to Upload	Click Browse to find a firmware file from the local host.

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 6-7** 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 6-7 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. Go to **Service** -> **IP PBX Service**, and click **Reload** to activate changes.

6.4.1 Add a Route

1. Enter settings shown in **Table 6-8**.
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 6-8** 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 **Delete** 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 6-8 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, 32 characters maximum, 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

	<p>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.</p> <p>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.</p>
Number of Stripped Digits	<p>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.</p>

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. The administrator can click **Group ID**, **Description** and **Associated Routes** arrows to sort the order of data.

Select **Route Management-> Route Group**, and the administrator can add, edit, delete and search routegroups in the **Route Group Management** page. Go to **Service -> IP PBX 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 6-9**, and click **Back**.

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 6-9**.
3. Click **Back** to see the updated information.




6.5.3 Delete a Route Group

1. Select a **Group ID**.
2. Click **Delete** 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 6-9 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 ²²	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 would be matched first for an outgoing call.  If there is no appropriate routes to select initially, one can come back later to revise it, once the expected routes are added.

6.6 SIP Trunk Configuration

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

²² Please refer to 6.4 for details.

The **SIP Trunk Management** page allows the administrator to configure SIP trunks used by IP PBX. Select **Trunk -> SIP Trunk**, and one can add, edit, delete and search SIP trunks. Go to **Service -> IP PBX 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 6-10**.
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 6-10**.
3. Click **Apply** to change the information.

6.6.3 Clone a SIP Trunk

1. Click an identifier in **Trunk Identifier**.
2. Click **Clone**.
3. Edit settings shown in **Table 6-10**.
4. Click **Add** to see the updated information.

6.6.4 Delete a SIP Trunk

1. Click the **Trunks** tab, and select a trunk identifier.
2. Click **Delete** to remove the SIP trunk from the **Trunk Identifier**.

6.6.5 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 6-10 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.


Auth. Name	Specify the name for authentication if different to the Trunk Identifier . ☞ IP PBX do not accept the following characters: ` ; ~ ! # \$ % ^ * () + - = \ { } [] " ' , / < > .
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 IP PBX at a remote site with dynamic IP address. Once the remote IP PBX registers, calls from local to remote can be made reversely over the trunk.
SIP Proxy IP	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP proxy, which usually refer to the SIP server on the ITSP side.
SIP Proxy Port	
SIP Domain	Specify the SIP domain used by the proxy and registrar. If not specified, IP address will be used as the domain by default.
Registration Required	Select if registration to a registrar is required to activate the trunk. This is true for a remote IP PBX or an ITSP account, however, may be not required in case of a SIP gateway.
SIP Registrar IP	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP registrar, which usually refer to the SIP server on the ITSP side (same as proxy).
SIP Registrar Port	
Language	Preferred language for system instructions heard from the trunk.
DID Type	DID means direct inward dialing (also called DDI in Europe). Select a preferred type, None DID , Extension DID , DID by Number DID by Privilege or Centrex DID from the list and then enter configuration in DID Prefix and DID Stripping to have the incoming calls directed to the corresponding trunk.
None DID	When selected None DID , all incoming calls will enter IVR system instead of directly dial to a specified extension. Select a preferred IVR from IVR list for this trunk.
Extension DID	When selected Extension DID , select an extension in the list to be an unconditional destination for incoming calls to this trunk. If prefix or stripping has been given, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2 nd dialing.

	<p>☞ 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.</p>	
DID By Number	When selected DID By number , 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.	
DID By Privilege	When selected DID By Privilege , select a usergroup in the list as the privilege of inbound calls from this trunk. Enter configuration in DID Prefix and DID Stripping to have the incoming calls redirected to dial out.	
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. Click All to strip all digits of the original called number.	
Centrex DID	Select this function to have more flexible calling numbers.	
	All-number Digitmap	Select a digitmap ID for calls via the trunk changing numbers.
	Default Number	Enter digits for calls from this trunk displaying this caller ID.
	Inbound Manipulation Digitmap	Select a digitmap ID to have the incoming calls direct to the corresponding extensions.
	Outbound Manipulation Digitmap	Select a digitmap ID to have the outgoing calls display the corresponding numbers.
IVR List ²³	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 ²⁴ 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.	

²³ Please refer to 7.12 for details.

²⁴ Please refer to 6.1 for details.

	<p>☞ There may not be appropriate usergroups to select initially. One can come back later once the expected usergroup has been added.</p>
Disable Fast Bridging	<p>Select to disable express media forwarding.</p> <p>☞ With Fast Bridging feature enabled, if the two parties involved in a call (for example, one IP extension and one SIP trunk) use different DTMF modes (RFC2833/SIP INFO/Inband), inline transfer (*#) or 2nd-dialing might fail. To avoid such problem, it is recommended to set the same DTMF mode for all IP extensions and SIP trunks in the IPBX, as well as for all IP phones registered to the IPBX. If it is not feasible to set the same DTMF mode for some IP extensions or SIP trunks, and inline transfer or 2nd-dialing is necessary for those IP extensions or SIP trunks, the Fast Bridging feature can be disabled on a per IP extension and per SIP trunk basis. Note that Fast Bridging is enabled by default.</p>
Advanced Settings	Select to see more settings shown below.
DTMF Mode	Select a preferred DTMF mode, Inband , 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 Auto from the list, and IP PBX will automatically detect the DTMF mode on this SIP trunk is Inband or RFC2833.
Try Peer-to-peer RTP	Click NO to disable or IP PBX will attempt to notify the two peers in a conversation to try peer-to-peer RTP transmission. This is suggested as long as phone and ITSP side support re-INVITE or UPDATE method during a connected call to save the resource of IP PBX. However, only SIP INFO DTMF mode should enable this since other DTMF modes require IP PBX being RTP relay server to support in-line transfer.
User Agent	Override default User-Agent header content.
From Caller ID	Send the entered number as Caller ID.
Bandwidth Sensitive	Click Bandwidth Sensitive and specify a limit of bandwidth in kbps for call admission in Bandwidth Limitation .
Bandwidth Limitation (kbps)	
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.
Enable ENUM Resolution	Select to use ENUM resolution, or leave it as blank.
Clear Bindings Prior Registration	If failed to the registration and cannot identify any abnormal settings, select to un-register before registering again.
Disable NAT Traversal	IP PBX uses NAT traversal for outgoing traffics by default. Select to disable NAT traversal if media ports in SDP are trusted.
Anti-SIP Blocking	Select to ignore illegitimately inserted CANCEL messages from ISP.
Gateway Trunk	Select to (1) trust the peer if IP matches and (2) retain caller ID.
Call Keep Alive	Select to check if the call is still ongoing in a certain time.
Registered Keep Alive	Select to check the registration status 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 Auto .
WAN LBR Preference	Select a low bit rate codec to be used as the first priority for outbound calls. e.g. select Auto to use caller's low bit rate codec, if any, or G.729 as the first priority.  To use WAN LBR Preference on a SIP trunk, the WAN Bandwidth Saver function must be enabled.

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 IP PBX.

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** -> **IP PBX Service**, and click **Reload** to activate changes.

6.7.1 Add an Analog PSTN Trunk

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

2. Enter settings shown in **Table 6-11**.
3. Click **Add** to see the newly added Analog PSTN trunk in **Trunk Group**.

6.7.2 Edit an Analog PSTN Trunk



1. Click a trunk group in **Trunk Group**.
2. Enter settings shown in **Table 6-11**.
3. Click **Apply** to change the information.

6.7.3 Delete an Analog PSTN Trunk

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

Table 6-11 Analog PSTN Trunk Configuration Settings

Field	Description
Trunk Group	ID number of this PSTN trunk group. A valid number ranges from 1 to 31. It should not overlap with existing ISDN PSTN trunk groups.
Trunk Type	Select the port type, FXO or FXS .
Trunk Ports	Select one or more FXO or FXS ports for this Analog PSTN trunk.
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 Type	If selecting FXO , user can see None DID , Extension DID and DID By Privilege in the DID Type list. If selecting FXS , users can see None DID , Extension DID and DID By Privilege in the DID Type list.
None DID	When selected None DID , all incoming calls will enter IVR system instead of directly dial to a specified extension. Select a preferred IVR from IVR list for this trunk.
Extension DID	When selected Extension DID , select an extension in the list to be an unconditional destination for incoming calls to this trunk. If prefix or stripping has been given, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic

	<p>2nd dialing.</p> <p> 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.</p>
DID By Number	When selected DID By number , enter configurations in DID Prefix and DID Stripping to have the incoming calls directed to the corresponding extension derived by number manipulation. The trunk numbers is therefore regarded as the direct line of the extension.
DID By Privilege	When selected DID By Privilege , select a usergroup in the list as the privilege of inbound calls from this trunk. Enter configuration in DID Prefix and DID Stripping to have the incoming calls redirected to dial out.
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. Click All to strip all digits of the original called number.
Language	Preferred language for system instructions heard from the trunk.
IVR List ²⁵	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 ²⁶ 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. <p> There may not be any appropriate usergroups to select initially. One can come back later to revise it, once the</p>

²⁵ Please refer to 7.12 for details.

²⁶ Please refer to 6.1 for details.

	expected usergroups are added.
Delay Before/After Answering	Delay in seconds before and after answering a call from PSTN trunk.
Forced Disconnection of 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 ISDN PSTN Trunk Configuration

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

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

6.8.1 Add an ISDN PSTN Trunk

1. Click the **Add New** tab.
2. Enter settings shown in

Table 6-12.

3. Click **Add** to see the newly added ISDN PSTN trunk in the **Trunk Group**.

The newly added ISDN Trunk shall display in the **Trunk Group**.

6.8.2 Edit an ISDN PSTN Trunk

1. Click a trunk group in **Trunk Group**.
2. Enter settings shown in

Table 6-12.


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

6.8.3 Delete an ISDN PSTN Trunk

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

Table 6-12 ISDN Trunk Configuration Settings

Field	Description
Trunk Group	ID number of this ISDN trunk group. A valid number ranges from 1 to 31. It should not overlap with existing Analog PSTN trunk groups.
Trunk Channels	Each physical ISDN port occupies three channels, two B and one D channels. The Trunk Channels displays B channels only. User only needs to select one or more B channels here. E.g. 1,2 is for the first physical port, 3,4 is for the second physical port and so on.
Description	Arbitrary description information.
Port Selection	Select to search for an available port in the group. Rotating means to force ports being selected by turns to even cost.
Switch Type	Supports European switch type by default.
DID Type	DID means direct inward dialing (also called DDI in Europe). Select a preferred type, None DID , Extension DID , DID by Number and DID by Privilege from the list and then enter configuration in DID Prefix and DID Stripping to have the incoming calls directed to the corresponding trunk.
None DID	When selected None DID , all incoming calls will enter IVR system instead of directly dial to a specified extension. Select a preferred IVR from IVR list for this trunk.
Extension DID	When selected Extension DID , select an extension in the list to be an unconditional destination for incoming calls to this trunk. If prefix or stripping has been given, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2 nd dialing. ☞ If you set a DID extension in a trunk, then only that extension can use this trunk to call out, and all incoming calls to this trunk will connect to that extension directly.
DID By Number	When selected DID By number , enter configurations in DID Prefix and DID Stripping to have the incoming calls directed to the corresponding extension derived by number manipulation. The ISDN trunk numbers is therefore regarded as the direct line of the extension.

DID By Privilege	When selected DID By Privilege , select a usergroup in the list as the privilege of inbound calls from this trunk. Enter configuration in DID Prefix and DID Stripping to have the incoming calls redirected to dial out.
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. Click All to strip all digits of the original called number.
Language	Preferred language for system instructions heard from the trunk.
IVR List ²⁷	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 ²⁸ of Privilege	When disabled DID, clicks a usergroup in the list whose reachability to other usergroups and trunks will use as the privilege of inbound calls from this trunk.  There may not be any appropriate usergroups to select initially. One can come back later to revise it, once the expected usergroups are added.
Caller ID	Leaves it blank to have the default caller ID, or enters a caller ID that is provided by your ISDN service provider.

6.9 POTS Setting



The POTS Setting page allows users to set the parameters of FXO/FXS/ISDN ports based on the requirement from the Central Office (CO) site. Users can select impedance from drop-down list and fine-tune the parameters. Click **Apply** to save modifications. Go to **Service -> IP PBX Service**, and click **Restart** to activate new settings.

²⁷ Please refer to 7.12 for details.

²⁸ Please refer to 6.1 for details.

6.9.1 FXO Port Configuration Settings


Table 6-13 FXO Port Configuration Settings

Field	Description
Type	The type of this port.
Port	The ID number of physical port.
Impedance/CP Tone	The Impedance/CP Tone setting determines the dial tone, busy tone, and ringback tone to the originating party. Select the appropriate impedance/call-progress tone for the local region.
Default Codec	Select g711ulaw or g711alaw for connecting to other devices.
Input/Output Gain	Voice amplification or attenuation in dB scale to adjust input/output volume of a PSTN line.
Caller ID Detection	Select to detect the Caller ID calling from PSTN lines.  When DID to an extension number, it is suggest to enable this function, or the forwarding settings at phone side may not work properly.
Answering by Battery Reversal Detection	If selected, billable time will count from when the call is answered.  Please enable this function when Central Office (CO) site provides battery reversal.
Minimum Disconnection Tone	Minimum volume level of the disconnection tone. If a PSTN trunk is found to have disconnection problem and low-voice , choose a lower dB.
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 force to terminate the call. Enter 0 or leave it blank to disable the function.
1 st / 2 nd Frequency of On-cycle Tone	Enter a number in Hz for call disconnection on-cycle tone.
1 st / 2 nd Frequency of Off-cycle Tone	Enter a number in Hz for call disconnection off-cycle tone.
Min/Max Duration of On-cycle Tone	Enter a time in ms for minimum or maximum duration of on-cycle tone.

Min/Max Duration of Off-cycle Tone	Enter a time in ms for minimum or maximum duration of off-cycle tone.
Forced Disconnection of this Duration (ms) of Silence Detected	Enter a time in ms for disconnecting the call when there is no sound for the specific period of time.

6.9.2 FXS Port Configuration Settings

Table 6-14 FXS Port Configuration Settings

Field	Description
Type	The type of this port.
Port	The ID number of physical port.
Impedance/CP Tone	The Impedance/CP Tone setting determines the dial tone, busy tone, and ringback tone to the originating party. Select the appropriate impedance/call-progress tone for the local region.
Default Codec	Select g711ulaw or g711alaw for connecting to other devices.
Input/Output Gain	Voice amplification or attenuation in dB scale to adjust input/output volume of a PSTN line.
Caller ID Detection	Select to detect the Caller ID calling from PSTN lines.
Answering by Battery Reversal Detection	<p>If selected, billable time will count from when the call is answered.</p> <p> Please enable this function when Central Office (CO) site provides battery reversal.</p>
Minimum Disconnection Tone	Minimum volume level of the disconnection tone. If a PSTN trunk is found to have disconnection problem and low-voice, choose a lower dB.
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 force to terminate the call. Enter 0 or leave it blank to disable the function.

6.9.3 ISDN Port Configuration Settings

Table 6-15 ISDN Port Configuration Settings


Field	Description
Type	The type of this port.
Port	The ID number of physical port.
Impedance/CP Tone	The Impedance/CP Tone setting determines the dial tone, busy tone, and ringback tone to the originating party. Select the appropriate impedance/call-progress tone for the local region.
Default Codec	Select g711ulaw or g711alaw for connecting to other devices.
Compand Type	Compand standards vary between countries. Select MULAW or ALAW for the local region.
Signaling	Select Point to point or Point to multipoint depends on the link type between ISDN service provider and your device.
Caller ID	Leave it blank to have the default caller ID, or enter a caller ID that is provided by your ISDN service provider.

6.10 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”. in Route Configuration explains digit set and wildcard characters.

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.

6.10.1 Add a Digitmap

1. Click the **Add New** tab.
2. Enter settings show in **Table 6-16**.
3. Click  to see the pattern in the list.
4. Click **Add** to have newly added digitmap.

6.10.2 Edit a Digitmap

1. Select a digitmap ID.
2. Edit settings show in **Table 6-16**.
3. Click  or  to add or delete the pattern.
4. Click  or  to adjust the order of sequence.
5. Click **Apply**.

6.10.3 Delete a Digitmap

1. Select a digitmap ID.
2. Click **Delete** to remove the ID.

Table 6-16 Digitmap Configuration Settings

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.
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.
Prefix	<p>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.</p> <p>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.</p>

7 Feature Configuration

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

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 (*# + Pilot Number). IP PBX will respond an available park line from the pool of call park numbers to the callee. After that the callee may hang up current phone, move to another phone, and dial the park line number told by IP PBX to resume conversation with the caller. If the callee does not call the given park line number to retrieve his call before timeout, IP PBX will ring the original extension where the callee answered the call. To configure Call Park feature, select **Feature** -> **Call Park**.

1. Enter settings shown in Table 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 **Delete** 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 devices. A calling party could reach this virtual conference room by dialing in a conference. Moreover, you can enter the meet-me conference as administrator or normal user level. Administrator has more privilege during the conference. Please refer to Table 7-3.

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

Note: The administrator 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 **Delete** 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	<p>PIN for normal users to join the conference.</p> <p>During a conference, a normal user has following options:</p> <ul style="list-style-type: none">- # to quit conference- *1 to mute/unmute- *9 to log in as the administrator if there is no administrator dialed in yet.
Administrator PIN	<p>PIN for the administrator of the conference.</p> <p>During a conference, the administrator has following options:</p> <ul style="list-style-type: none">- # to quit conference- *1 to mute/unmute- *2 to lock/unlock the conference- *3 to invite a user into the conference- *4 to drop a party from the conference- *5 to drop all parties in the conference

-
- | | |
|--|--|
| | <ul style="list-style-type: none">- *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 Music on Hold Management

7.4.1.1 Add a MOH File

Enter settings shown in

1. Table 7-4.
2. Click **Add** to see the newly added file in the **MOH ID**.

7.4.2 Media File Management

7.4.2.1 Edit a MOH File

1. **Upload Media File:** Click **Browse** to select a file and Click **Put File** to upload.
2. **Delete Media File:** Select a file from the drop-down list and Click **Delete** to remove it.
3. **Default MOH:** Click a MOH ID from the drop-down list and click **Set** in the row.

7.4.3 MOH ID List

7.4.3.1 Edit a MOH ID

1. Click a file from the **Media File** and click **Apply** in the row.

7.4.3.2 Delete a MOH File

1. Select a MOH ID.
2. Click **Delete** to remove the MOH ID from the **MOH ID List**.

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 File list, and click Delete to remove it.
Default MOH	Select to use this music file for system default MOH globally.


7.5 Voicemail

IP PBX has a built-in voice mail subsystem with a sophisticated IVR menu. A call to an extension in use or no answer could be configured to enter voice mail recording procedure. After leaving a message, a notification e-mail will be sent to the user owns the extension with or without the message in the form of an attached WAV file. The Message Waiting Indicator (MWI) on IP phones (if any) will be lit. For analog phones, the user will hear six short beeps before the normal dial tone when picking up the analog phone. The user could then dial the voicemail pilot number to enter voice mail system to manage messages such as playback, delete, or move them from inbox to different folders. In addition to indicating current voice mail capacity on the management page, IP PBX can send an alarm email to the administrator when the available voice mail space reaches the threshold. To configure Voicemail feature, select **Feature -> Voicemail**.

1. Enter settings shown in Table 7-5.
2. Click **Apply**.

Table 7-5 Voice Mail Configuration Settings

Field	Description
Voicemail Pilot Number	Number to access voice mail system IVR, e.g. 6666.
Minimum Message Time	Messages less than this duration will not be saved as a message. 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 <i>from</i> 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	<p>Show the available space in Kbytes and minutes.</p> <p> The storage inside IP PBX saves not only voice mails but also some other stuff, such as CDR and logs. The remained disk space is all for voice mails, and it is the “maximum” available voice mail space.</p>
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 and the prompt is marked “*” in the dropdown list if it is not a default prompt. 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 and the prompt is marked “*” in the dropdown list if it is not a default prompt. 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, please enter your mailbox again.
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.

Lineisdown	Device Unreachable.
Isnotreg	Device Unregistered.

7.8 Broadcast

A user can arrange an event at the exact time in IP PBX. IP PBX 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, IP PBX 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 **Apply** to change the information.

7.8.3 Delete a Broadcast

1. Click the **Management** tab.
2. Select an **Event**.
3. Click **Delete** 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.

	<p>Playback: Play the uploaded WAV file in the Customized Media File box to the callee extensions.</p> <p>Dial: IP PBX 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.</p> <p>It is suggested to set one number only in the Callee Extensions unless the extension number in the Dial Extension field is multi-line.</p> <p>Play default music: Play default music configured in the Music On Hold page to extensions in the Callee Extensions box.</p>
Customized media File	<p>Upload a *.wav file, if Playback is selected in the Action list.</p> <p> The recording format must be 8000 Hz, 16 bit, mono Windows PCM WAV file.</p>
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. Click  /  to select/remove the selected callee extension(s) from this list.

7.9 Worktime

Worktime page 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 Table 7-9.
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 Table 7-9.
4. Click **Apply** to change the information.

7.9.3 Delete a Worktime

1. Click the **Management** tab.
2. Select a **Group ID**.
3. Click **Delete**.

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.
Mode	Select one of the four modes: 1: No work on weekends. 2: Work off and on by turns on Saturdays. 3: Work half-day on Saturdays. 4: Work on all 7 days
General Worktime	The work time from Monday to Friday.
Saturday Worktime	The work time for Saturdays, this field only active when mode is set to 2 or 3.
Optional Worktime	Special holidays or work day. User can set date and its work time, or set it to a whole-day holiday. Click  /  to select/remove the selected item(s) from this list.

7.10 Memo Call

A user can set a memo at a specific time, e.g. a morning call, in IP PBX to inform the user or another user set in the **Extension** list by call at the time. When the user picks up the phone, IP PBX plays “voice file” or “music”. The user can also set this memo as a daily routine so that IP PBX 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 **Apply** to update the information.

7.10.3 Delete Memo Call

1. Click the **Management** tab.
2. Select a **Memo**.
3. Click **Delete** 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 Customized Media File box to the extension(s). Play default music: Play default music configured in the Music On Hold page to the extension(s).
Customized Media File	Upload a *.wav file, if Playback is selected in the Action list.  The recording format must be 8000 Hz, 16 bit, mono 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 Automatic Call Distribution

IP PBX implements Automatic Call Distribution (ACD) by queues and agents. A pool of agents is created first and then one or more queues could be added with distinct service numbers and a list of agent extensions. Each agent should be bound with an existed extension to take calls and each queue could have a specific distribution policy, such as round robin, ring all, least recent, fewest calls, and random. With such settings one can enable agents located at any location to serve a unified queue of calls. This comes with a benefit that administrative changes can be made in a single place, and these changes immediately become effective for all calls and all agents. This not only enables distribution centers to be operated cost-effectively, but also reduces the cost and complexity of implementing home-based agents, who now only need an IP connection to serve calls in queues. Refer to for creating an agent pool, and for setting queues. Select **Feature** -> **ACD** to configure agents and queues.

Note that the agents are limited to local extensions; login through trunk is not allowed.

7.11.1 Set Agent Login and Logout

Before adding agents or queues of the ACD, the administrator has to give universal Agent Login and Logout numbers for all agents.

1. Click the **Agent Login / Logout** tab.
2. Enter numbers in **Agent Login** and **Agent Logout**.
3. Click **Apply** to apply login and logout numbers to all agents.

Note that the agent login and logout numbers can not be set the same.

7.11.2 Add an Agent

1. Click the **Agent Management** tab.
2. Enter settings shown in **Table 7-11**.
3. Click **Add** to see the information in the table below.

7.11.3 Edit an Agent

1. Click the **Agent Management** tab.

2. Edit settings shown in **Table 7-11** in a row of the table below.
3. Click **Apply** in the row to update the information.

7.11.4 Delete an Agent

1. Click the **Agent Management** tab.
2. Select an account in the table.
3. Click **Delete** to remove the agent from the table.

7.11.5 Add a Queue

1. Click the **Add New Queue** tab.
2. Enter settings shown in **Table 7-12**.
3. Click **Add**.

The newly added queue displays in the **Management** tab.

7.11.6 Edit a Queue

1. Click the **Manage** tab and a queue.
2. Enter settings shown in **Table 7-12**.
3. Click **Apply**.

The updated queue displays in the **Management** tab.




7.11.7 Delete a Queue

1. Click the **Management** tab.
2. Select a queue.
3. Click **Delete** to remove the queue from the Management page.

Table 7-11 Agent Configuration Settings

Field	Description
Account	Enter an account number for the agent.
PIN	Enter a login password for the agent.
Agent Name	The name of agent. A unique ID containing alphabets, numbers, and underscore only without spaces; 20 characters maximum.

Table 7-12 Queue Configuration Settings



Field	Description
Queue Name	Name a queue. A unique ID containing alphabets, numbers, and underscore only without spaces; 20 characters maximum.
Waiting Music	The default media file in Music On Hold is the waiting music.
Distribution Policy	Ways of distributing calls to the agents in a queue: Round Robin: take turns to ring each available agent. Ring All: ring all available agents until one answers (default). Least Recent: ring the agent that was least recently called in this queue. Fewest Calls: ring the agent with fewest completed calls in this queue. Random: ring the agents randomly.
User Group	Select a usergroup for this queue. Only extensions that belong to the usergroup can login the queue.
Timeout	Set a timeout period in seconds when no agent answers the call.  The time must be shorter than Timeout To Next Forward of the extension, or the call will be forwarded to an extension that is set in Unavailable Call Forward in Extension of IP Phone .
Retry Time	Set a waiting time in seconds to ring agents in the queue again if the call is not answered.
Service Extension	Set the representative number of this queue, e.g. a queue named Support, and its service extension is 500. There are three agents with extensions of 155, 157, and 166 in Support. The Distribution Policy is set to Ring All . When a customer calls the service extension 500 for Support, extensions 155, 157, and 166 will all ring simultaneously.
Max Len	Select number of calls for waiting when all agents are occupied.
Agent List	Click  /  to select/remove the selected agent from this queue. The left box lists the available agents, and the right box shows the selected agents.

7.12 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. IP PBX enables multiple configurable IVR menus in a single system, and each of them could have a hierarchy up to three layers. Select **Feature** -> **IVR** to add, edit and delete the IVR menus. You can

also manage IVR prompts, used by IVR menus, in this page.

7.12.1 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 7-13**.
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.

7.12.2 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 prompt and worktime settings.
5. For other settings, edit settings shown in **Table 7-13**.
6. Click **Save** to update the changes.



7.12.3 Clone an IVR Menu


1. Select an IVR name from the **All IVR Menus** list.
2. Click **Clone**.
3. Enter a new IVR name.
4. Edit settings shown in **Table 7-13**.
5. Click **Save** to have the IVR in the **All IVR Menus** list.

7.12.4 Delete an IVR Menu

1. Click an IVR name in the **All IVR Menus** list.
2. Click **Delete** to delete the IVR menu.

Table 7-13 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	<p>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.</p> <p>Click ext and select a usergroup in the Group list in Action Data to allow dialing any extension in this usergroup.</p> <p>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.</p> <p>Click  to add the rule in the Node list.</p> <p> 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.</p> <table border="1"> <tr> <td>Hang Up</td> <td>To cut off the call immediately.</td> </tr> <tr> <td>Greeting</td> <td>To play the selected prompt. The caller can press keypad to enter numbers when the prompt plays.</td> </tr> <tr> <td>Announce</td> <td>To play the selected prompt. The caller can not enter any digits when the prompt plays .</td> </tr> <tr> <td>Call To</td> <td>To call an extension.</td> </tr> <tr> <td>Go to Top</td> <td>To go back to the root menu of the IVR.</td> </tr> <tr> <td>Next Layer</td> <td>To go to the next layer of the IVR menu.</td> </tr> <tr> <td>Set Language</td> <td>To choose a language.</td> </tr> <tr> <td>Next Layer & Set Language</td> <td>To go to the next layer of the IVR menu and select a language for the layer.</td> </tr> <tr> <td>Auth. Dial</td> <td>To allow caller entering the password for Auth.</td> </tr> </table>	Hang Up	To cut off the call immediately.	Greeting	To play the selected prompt. The caller can press keypad to enter numbers when the prompt plays.	Announce	To play the selected prompt. The caller can not enter any digits when the prompt plays .	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.	Set Language	To choose a language.	Next Layer & Set Language	To go to the next layer of the IVR menu and select a language for the layer.	Auth. Dial	To allow caller entering the password for Auth.
Hang Up	To cut off the call immediately.																		
Greeting	To play the selected prompt. The caller can press keypad to enter numbers when the prompt plays.																		
Announce	To play the selected prompt. The caller can not enter any digits when the prompt plays .																		
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.																		
Set Language	To choose a language.																		
Next Layer & Set Language	To go to the next layer of the IVR menu and select a language for the layer.																		
Auth. Dial	To allow caller entering the password for Auth.																		

		Dial ²⁹ function
	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. If Next Layer action is selected, you can edit its prompt and worktime settings by selecting the node and clicking Modify next to the box.	
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	<p>Select a recording file that you added from the IVR Prompts Management tab, or select one of the following default voice files. The following default files include all languages that IP PBX has for you to select. Please click a language in the Languages list.</p> <ul style="list-style-type: none"> - Please enter a new extension followed by the # key. - Thank you. - I'm sorry, that is not a valid extension, please try again. - Please hold while I try out that extension. - System is busy at this moment. Please try again later. - Goodbye. - I'm sorry, I do not understand your response. - Beep.
	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 .	

²⁹ Please refer to 6.1 for details.

In-Hour Actions	Select one action during business hours.	
	Greeting	To play the selected prompt. The caller can press keypad to enter numbers when the prompt plays.
	Announce	To play the selected prompt. The caller can not enter any digits when the prompt plays.
	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 caller can press keypad to enter numbers when the prompt plays.
	Announce	To play the selected prompt. The caller can not enter any digits when the prompt plays.
	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.	

7.12.5 IVR Prompts Management

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

7.12.5.1 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.

Note that the uploading file format must be 8000 Hz, 16 bit, mono Windows PCM WAV file.

7.12.5.2 Delete an IVR Prompt

1. Select a *.wav file from the **All Files** list.

2. Click **Delete**.

The deleted file shall disappear from the **All Files** list.

7.12.6 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 7-14 IVR Parameters Configuration Settings

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.

7.12.7 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. If you want to change back to the original default prompt, click a language and a prompt in the **Language** and **Prompts** list, 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. The recording format must be 8000 Hz, 16 bit, mono Windows PCM WAV file.

Table 7-15 Replaceable Auto Attendant Prompts Configuration Settings

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.

8 Example Provisioning

This chapter introduces several practical configuration examples of IP PBX deployment. The configuration of IP PBX is very flexible and the expressiveness of usergroups, routegroups, and trunks are scalable enough to support various network architectures. Users could 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 6.1).
- Configure users and assign every user to usergroup **ALL** (refer to 6.2).
- Configure devices and an extension for each device (refer to 6.3).
- Assign each extension to a corresponding user (refer to 6.3).
- Configure each client phone with respect to the extension number and password in its IP PBX extension configuration accordingly.
- Reload the IP PBX service (refer to 5.6.2).

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

8.2 Case I: Singe-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 6-1**. The provisioning tasks include:

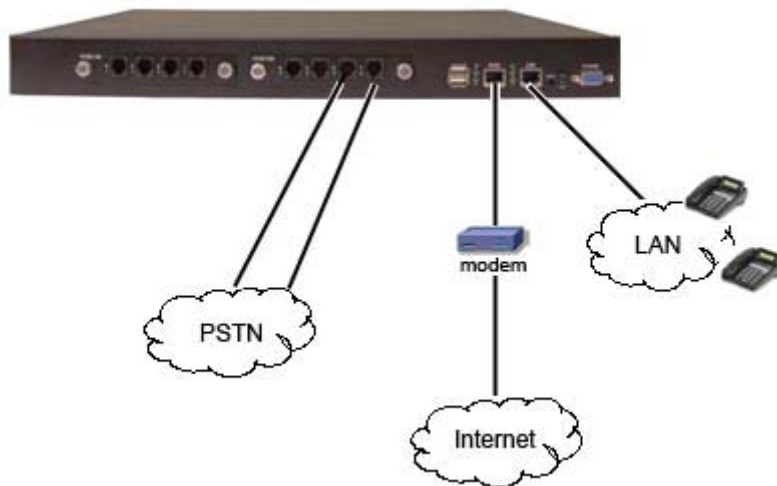


Figure 6-1

- 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", do not select any DID of extension, and select **ext-all** as the usergroup of privilege.
11. Return to usergroup configuration. For usergroup **staff**, choose **pstn-out** as outbound routegroup, 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 IP PBX Service.

8.3 Case II: Two-site configuration

This case describes the typical settings of a two-site configuration; say Company B headquarters B-HQ and its branch B-BR 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 6-2**. The provisioning tasks include:

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

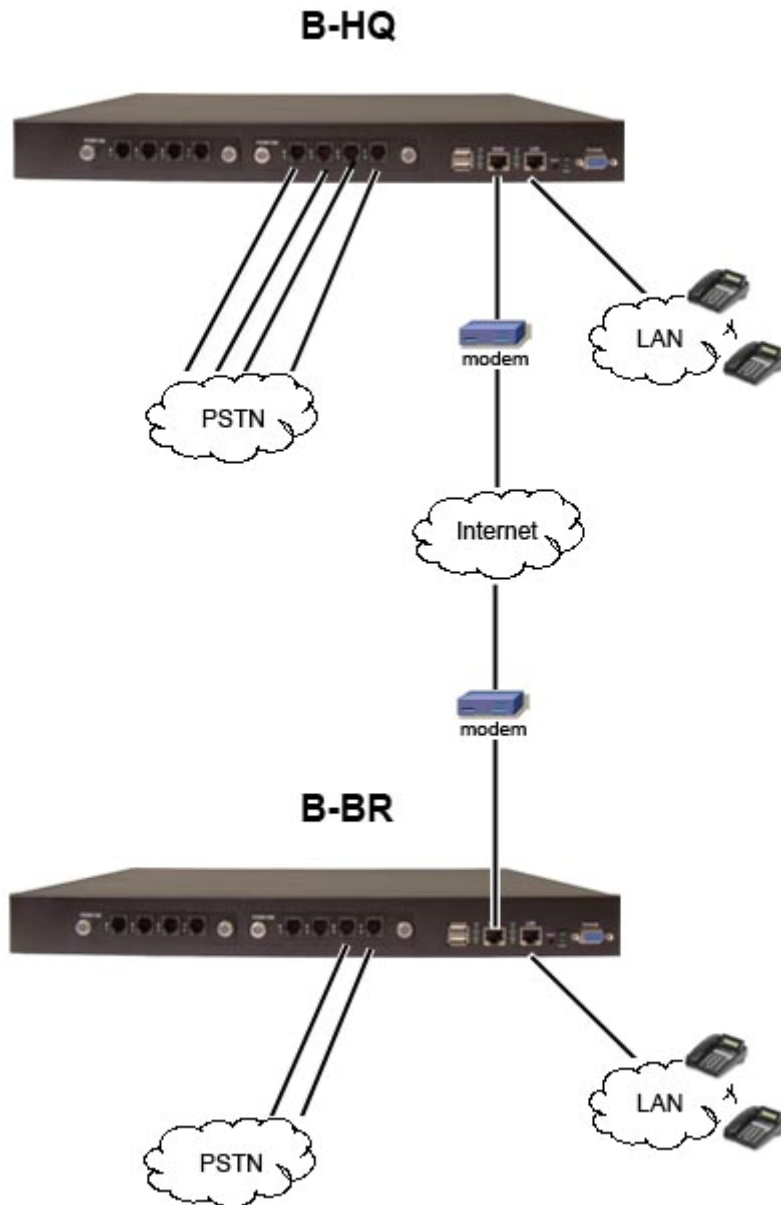


Figure 6-2

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**”, 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**; do not select DID, and select **staff** as the usergroup of privilege.
15. Return to usergroup configuration. For usergroup **staff**, choose **pstn-out** as outbound routegroup for **pstn1**, choose **to-br** as outbound routegroup for **100** in Associated Trunks, and select **utility** in Reachable User Group; while for usergroup **utility**, only select **staff** in Reachable User Group.
16. Reload the IP PBX 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**, 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**; do not select DID, and select **staff** as the usergroup of privilege.
15. Return to usergroup configuration. For usergroup **staff**, choose **pstn-out** as outbound routegroup for **pstn1**, choose **to-hq** as outbound routegroup for **100** in Associated Trunks and select **utility** in Reachable User Group; while for usergroup **utility**, only select **staff** in Reachable User Group.
16. Reload the IP PBX Service.

9 Appendix

9.1 Keypad Default Settings for IP PBX

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

Keypad	Description
*6	Place a call on hold.
*7	Remove the call from hold.
*8	If extensions are in the same pickup group ³⁰ , any extension in the pickup group can press *8 to pick up the call in ringing state.
8	Pickup a call for a specific extension.
*# + 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 ³¹ .

9.2 Managing with CLI Commands

9.2.1 Introduction

The IP PBX Console Interface (ESTCLI) administration guidelines describe how to manage some basic functions of the IP PBX 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.

³⁰ Refer to Table 3-4 for more detail of pickup group.

³¹ Refer to 7.1 for more detail of Call Park.

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
 - 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 / ?**

Display the available commands after the commands you entered

CLI Command: *<CLI command> ?*

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

Display the commands you entered before. You can also press \uparrow \downarrow 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: The message "Someone login. Press 'y' to login anyway, or any other key to quit" will show up if there is an old web GUI session existing.

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.

Help

Display the help message command available for the level.

CLI Command: **help / ?**

Display the available commands after the commands you entered

CLI Command: *<CLI command> ?*

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 \uparrow \downarrow on the keypad to show the used commands.

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.

List Commands

The administrator can also list items such as user, usergroup, route, routegroup, extensions, IP phone and trunks.

Show Users

Display one or all PBX users' information.

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

Show Usergroups

Display one or all PBX usergroups' information.

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

Show IP Phones

Display one or all IP phones' information.

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

Show Extensions of IP Phones

Display one or all extensions' information.

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

Show Analog Phones

Display one or all analog phones' information.

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

Show Routes

Display one or all PBX routes' information.

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

Show Routegroups

Display one or all routegroup information.

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

Show Digitmap

Display one or all SIP trunks' information.

CLI Command: **list digitmap** <Digitmap ID> / **list digitmap**

Show SIP Trunks

Display one or all SIP trunks' information.

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

Show Analog PSTN Trunks

Display one or all analog PSTN trunks' information.

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

Show ISDN PSTN Trunks

Display one or all ISDN PSTN trunks' information.

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

Show POTS and Channel Setting

Display all Analog including FXO and FXS POTS status.

CLI Command: **list pots**

Display all ISDN channel status.

CLI Command: **list channel**

Import/Export Commands

The administrator can import and export CSV files including items such as user, usergroup, route, routegroup, extensions, IP phone, SIP trunks or all data via TFTP or SSH server. Restart the PBX server in Admin Commands after deleting the usergroup.

Import CLI Command: **import** <user / usergroup / route / ipphone / siptrunk / full data> <tftp / scp>://<IP address / host>:<filename>.csv

Export CLI Command: **export** <user / usergroup / route / ipphone / siptrunk / full data>

Note: It is highly recommended to import full data when there isn't any information in the system.

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

Note: For importing data, the system only adds the new records or updates the existed records. If there is any dependency error, the import action will stop and go back to the original status.

Note: When exporting data, admin in User, UG_DEF in Usergroup and RG_DEF in Routgroup will not be exported.

Style for CSV file

This section displays the CSV file style that can import to CLI system. Each item will have two paragraphs. The first paragraph is field name, and the second one is an example. Note that comma (,) is required at the end of each line, and it is suggested to use UNIX text file format.

Note: As comma (,) is used for separating each data, avoid using comma including comma within quotation (“,”) in each data.

Note: When importing, Route and Routgroup will be imported together. As a consequence, they must be in the same CSV file. The same situation also applies to IP phone and extension of IP phone.

Note: Some items are related to others, such as Reachable Usergroup in Usergroup, Usergroup in User and so on. When importing an item that the related items do not exist in the system, the system will show the error message and the imported item will be incomplete.

User

#Tag,UserID,Name,Password,Usergroup,Attach,Description,Email,
\$User,user_01,AAA,111,UG_01,yes,I am user_01,user_01@a.com,

Usergroup

#Tag,GroupID,Features,RouteGroup;Trunk;GroupID;Weight,ReachableUG,Description,PassEnable,Passcode,
\$UG_Data,UG_01,mm,3301g;7710;1;1|3301g;7710;1;2,UG_10|UG_09,ThisisUG_01,

Route

#Tag,Route_ID,Description,Number_Pattern,Stripped_Digits,Prefix,
\$Route_s,331,331_out,331Z,4,3310,

Routegroup

#Tag,Route_Group_ID,Description,Associated_Routes,
\$Route_Group_s,3301g,3301gr,331;332,

IP Phone

#Tag,Device_ID,Device_URL,Verder_Prefix,Mac,Codec_0,PACKET_TIME_0,Codec_1,PACKET_TIME_1,Codec_2,PACKET_TIME_2,VAD,DTMF,1st_ProxyIP,1st_ProxyPort,1st_RegIP,1st_RegPort,1st_SIPDom,2nd_ProxyIP,2nd_ProxyPort,2nd_RegIP,2nd_RegPort,2nd_SIPDom,

\$IPPhone_s,201,192.168.1.221,sip201,aa:11:22:33:44:51,g711ulaw,10,g711alaw,20,g726r16,30,yes,inband,

Extension of IP Phone

#Tag,Exten_No,Asso_Device,Passwd,User,Pickup_UG,Lang,Voicemail_YN,Vm_pin,DTMF,Forward_Num,Unav_Timeout,LineType,LAN_Only,Disable_NAT,PEERTP,Call_Block,Timeout_Forward,Unav_Fwd_Prompt,Busy_Forward,Sele_Call_Fwd,Block_Anony,Max_VMSpace,Call_Alive,Reg_Alive,sipredirectiondeny,Fastbridgedisable,

\$Extension_IPPhone_1,201,201,pp201,user_01,UG_01,en,yes,201,inband,201201,10,wired,yes,no,INVITE,206;206,101201/10;201202/20,yes,211,111:Local/111;112:Local/112,yes,2110,yes,no,

SIP Trunk

#Tag,SIP_ID,Descript,Dyna_Peer,Auth_Name,Auth_Password,SIP_Proxy,SIP_Proxy_Port,Registration,SIP_Registrar,SIP_Reg_Port,DID,DID_Prefix,DID_Stripping,IVR,Lang,User_Group,SIP_Domain,User_agent,BW_Sensitive,BW_Limit,DTMF,PEERTP,Enable_ENUM,Clear_Bindings,Disable_NAT,Delay_Before,Delay_After,Call_alive,Reg_alive,Anti_SIPBlock,GW_trunk,From_callerID,Sub_Payload,Trunk_relay,Allow_Concurrent_Call,Centrexdid,Allnumberdigitmap,Centrexdefaultnumber,Inboundmanipulation,Outboundmanipulation,Fastbridgedisable,

\$SIP_Trunk_s,888,SIP Trunk

Data,yes,888,88,dynamic,,no,,bygroup,02,5,,en,UG_DEF,,no,,Auto,UPDATE,no,yes,no,1,1,yes,no,yes,,89132070,126,yes,3,no,,,,yes,

Full Data

Apply all the data above into a CSV file to import.

9.3 IP PBX Voicemail System Menu Tree

This section displays the options for IP PBX 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 prepending

*.Return to main menu

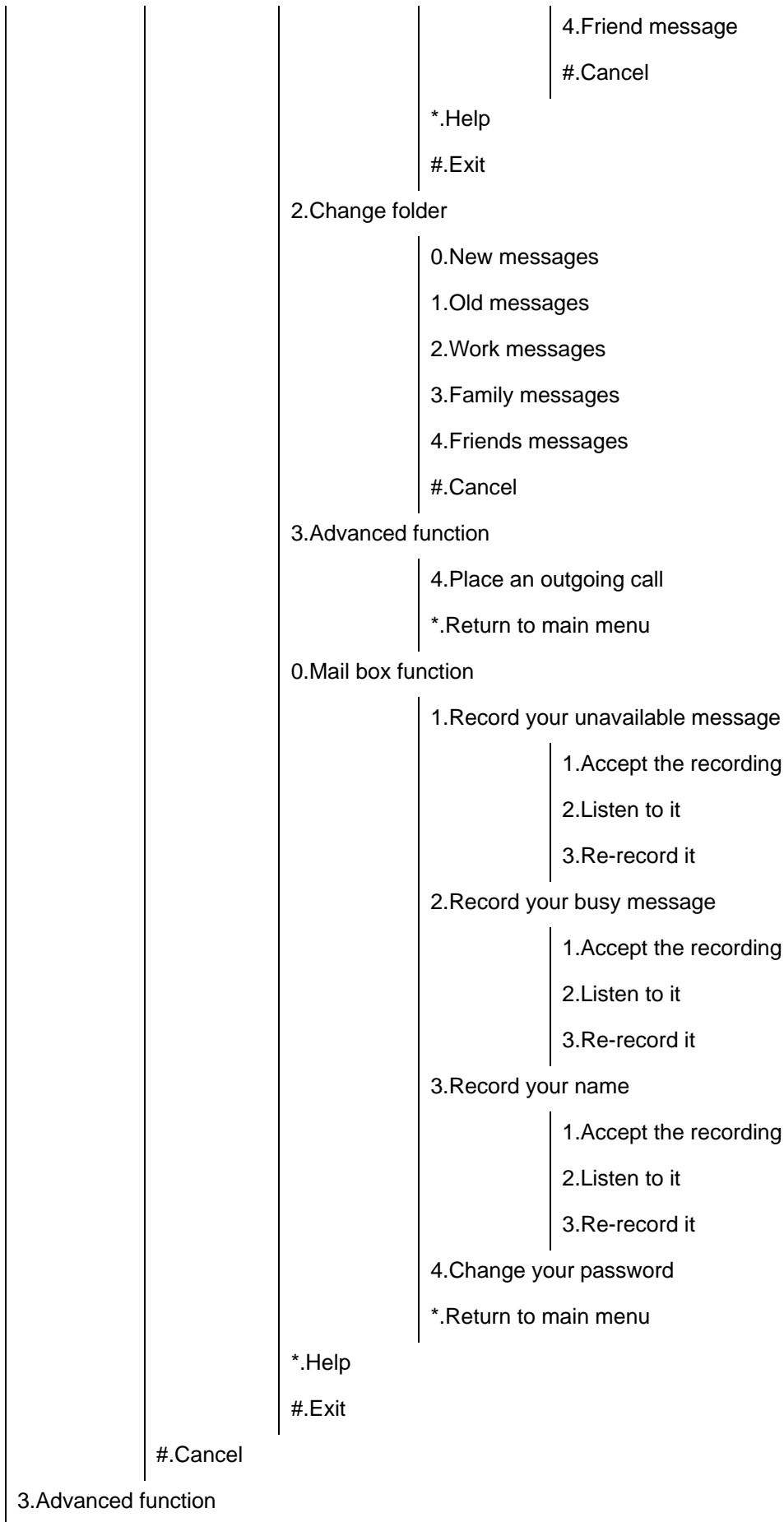
9.Save this message

0.New message

1.Old message

2.Work message

3.Family message



- 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

EC Declaration of Conformity

For the following equipment:

*Type of Product : Internet Telephony PBX system
 *Model Number : IPX-2000

* Produced by:

Manufacturer's Name : **Planet Technology Corp.**
 Manufacturer's Address: 11F, No 96, Min Chuan Road
 Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to 1999/5/EC R&TTE.

For the evaluation regarding the R&TTE, the following standards were applied:

Conducted / Radiated	EN 55022	(1998+A1:2000 +A2:2003, Class B)
Harmonic	EN 61000-3-2	(2000)
Flicker	EN 61000-3-3	(2001)
Immunity	EN 55024	(1998+A1:2001+A2:2003)
ESD	EN 61000-4-2	(1995+A1:1998+A2:2001)
RS	EN 61000-4-3	(2002+A1:2002)
EFT/ Burst	EN 61000-4-4	(1995+A1:2001+A2:2001)
Surge Test	EN 61000-4-5	(2004)
CS	EN 61000-4-6	(1993+A1:2001)
Voltage Disp	EN 61000-4-11	(1994+A1:2001)

Responsible for marking this declaration if the:

Manufacturer Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

Company Address: 11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C

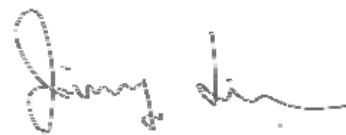
Person responsible for making this declaration

Name, Surname Jimmy Lin

Position / Title : Product Manager

Taiwan
Place

20 Oct., 2005
Date



 Legal Signature

PLANET TECHNOLOGY CORPORATION